



# Product Overview

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## New Information for This Release

The following changes have been made and the following features have been added to release 8.0:

- New configurable parameters have been added. See [Appendix D, “SIP IP Phone Parameters.”](#)
- Caveats can be found on the product release notes page at this URL:  
[http://www.cisco.com/en/US/partner/products/sw/voicesw/ps2156/prod\\_release\\_notes\\_list.html](http://www.cisco.com/en/US/partner/products/sw/voicesw/ps2156/prod_release_notes_list.html)

## Cisco Unified IP Phone 7960G and 7940G Overview

Cisco Unified IP Phone 7960G and 7940G models are full-featured telephones that can be plugged directly into a SIP network and can be used very much like a standard PBX telephone. Phone terminals can attach to the existing data network infrastructure, using 10BASE-T and 100BASE-T 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 or PBX.

The Cisco Unified IP Phone 7960G and 7940G also supports an adjustable ring tone, a hearing-aid compatible handset, and a headset.

The Cisco Unified IP Phone 7960G and 7940G complies with RFC 3261, as described in [Appendix A, “Compliance with RFC 3261.”](#)

See [Figure 1-1](#) and [Figure 1-2](#) to identify the buttons and hardware on your Cisco IP phone.

**Figure 1-1 Cisco IP Phone 7960G**



**Figure 1-2 Cisco IP Phone 7940G**

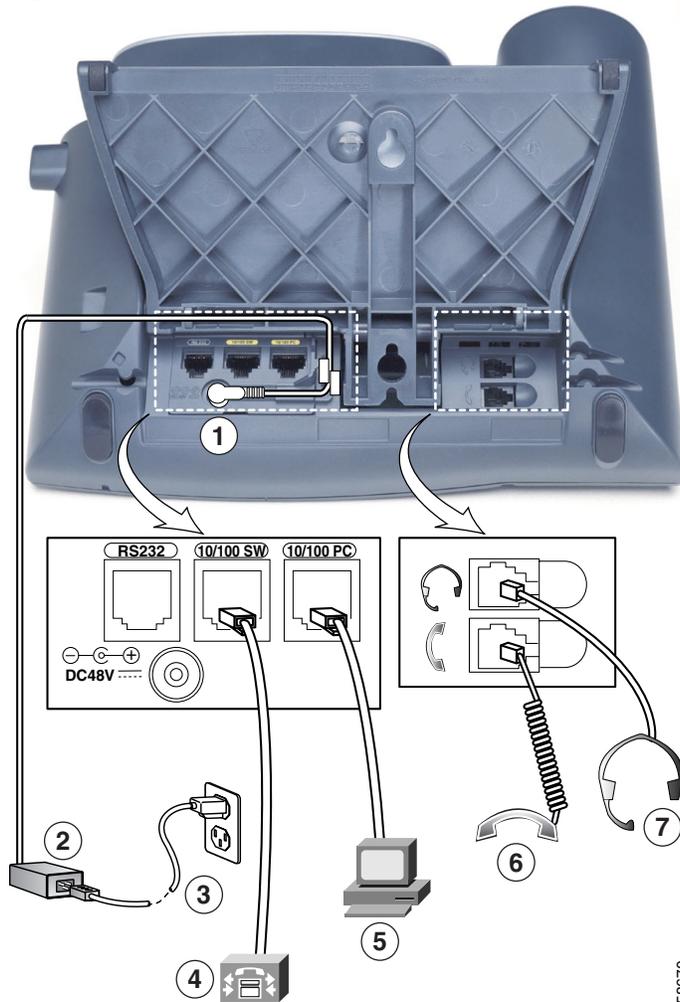


1	Handset with indicator light	The light strip at the top of the handset blinks when the phone rings and can be set to remain lit when there is a voice message.
2	LCD screen	Displays information about the Cisco IP phone, such as the time, date, phone number, caller ID, line and call status, and the softkey tabs. The screen is 4.25 x 3 inches (10.79 x 7.62 cm) and has an adjustable contrast.
3	Cisco IP Phone model type	Indicates the Cisco IP phone model.
4	Line or speed-dial button 	Opens a new line or speed-dials the number on the LCD screen. Phones in the Cisco Unified IP Phone 7960G series have six line or speed-dial buttons, and phones in the Cisco Unified IP Phone 7940G series have two.
5	Footstand adjustment	Allows adjustment of the angle of the phone base.
6	Directories button 	Provides access to call histories and directories.
7	<i>i</i> or ? button 	Provides online help for selected keys or features and network statistics about the active call. Pressing the button and then the up or down scroll key displays a descriptor of the key. For example, pressing the <i>i</i> or ? button and then the up or down scroll key displays a screen that instructs you how to scroll up and down on the LCD.
8	Settings button 	Provides access to phone settings such as contrast and ring sound, network configuration, and status information.
9	Speaker button 	Toggles the speaker on or off.
10	Mute button 	Toggles the mute on or off.
11	Headset button 	Toggles the headset on or off.
12	Volume button 	Increases or decreases the volume for the handset, headset, or speakerphone (depending upon which is currently active). Also controls the ringer volume (if the handset is in its cradle) and the LCD screen contrast.
13	Services button 	Provides access to any available phone services.
14	Messages button 	Provides access to a message system, if available.

15	Navigation button 	Allows scrolling through text and selection of features displayed on the LCD screen.
16	Dial pad	Works exactly like the dial pad on a traditional telephone.
17	Softkeys	Activates any functions displayed on the corresponding LCD screen tabs. Softkeys point to feature options displayed along the bottom of the LCD screen.

Figure 1-3 shows the connections on the back of the Cisco IP phone. Cisco Unified IP Phone 7960G/7940G models have the same hardware configuration.

Figure 1-3 Cisco IP Phone Cable Connections



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- |   |   |
|---|---|
| 1 | AC/DC adapter port (DC48V) for power connector. For redundancy, you can use the AC adapter even if you are using inline power from Cisco Catalyst switches. The Cisco Unified IP Phone 7960G and 7940G can share the power 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.   |
| 2 | Power supply with AC plug.  |
| 3 | Power cable with wall socket plug for connecting to power.  |
| 4 | Network port (10 and 100 SW) RJ-45 to connect the phone to the network supporting 10- or 100-Mbps half- or full-duplex Ethernet connections to external devices. You can use either Category 3 or Category 5 cabling for 10-Mbps connections, but use Category 5 for 100-Mbps connections. To avoid collisions, use full-duplex mode. 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. |

5	Access port (10 and 100 PC) RJ-45 to connect a network device, such as a computer, to the phone supporting from 10- to 100- Mbps half- or full-duplex Ethernet connections to external devices. You can use either Category 3 or Category 5 cabling for 10-Mbps connections, but use Category 5 for 100-Mbps connections. To avoid collisions, use full-duplex mode. You must use a straight-through cable on this port.
6	Handset port for connecting a handset.
7	<p>Headset port for connecting a headset. Enables the headset. The phone supports a four- or six-wire headset jack. The volume and mute controls 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 Unified IP Phone 7960G and 7940G.</p> <p>The phone supports the following Plantronics four- or six-wire headsets: Tristar Monaural, Encore Monaural H91, and Encore Binaural H101.</p> <p>When a headset is used, an amplifier is not required. However, a coil cord is required to connect the headset to the headset port on the back of your Cisco Unified IP Phone 7960G and 7940G. For information on ordering compatible headsets and coil cords for the Cisco Unified IP Phone 7960G and 7940G, go to <a href="http://cisco.plantronics.com">http://cisco.plantronics.com</a> or <a href="http://www.vxicorp.com/cisco">http://www.vxicorp.com/cisco</a>.</p>

## Session Initiation Protocol Overview

Session Initiation Protocol (SIP) is the Internet Engineering Task Force (IETF) standard for multimedia conferencing over IP. SIP is an ASCII-based, application-layer control protocol (defined in RFC 3261) that can be used to establish, maintain, and terminate calls between two or more endpoints.

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

## SIP Capabilities

SIP provides the capabilities to do the following:

- Determine the location of the target endpoint—SIP supports address resolution, name mapping, and call redirection.
- Determine the media capabilities of the target endpoint—Using Session Description Protocol (SDP), SIP determines the “lowest level” of common services between the endpoints. Conferences are established using only the media capabilities that can be supported by all endpoints.
- Determine the availability of the target endpoint—If a call cannot be completed because the target endpoint 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 that indicates why the target endpoint was unavailable.
- Establish a session between the originating and target endpoint—If the call can be completed, SIP establishes a session between the endpoints. SIP also supports midcall changes, such as the addition of another endpoint to the conference or the changing of a media characteristic or codec.
- Handle the transfer and termination of calls—SIP supports the transfer of calls from one endpoint to another. During a call transfer, SIP simply establishes a session between the transferee and a new endpoint (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.

Conferences can consist of two or more parties and can be established using multicast or multiple unicast sessions.

**Note**

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

## SIP Components

SIP is a peer-to-peer protocol. The peers in a session are called user agents (UAs). A user agent can function in one of the following roles:

- User agent client (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.

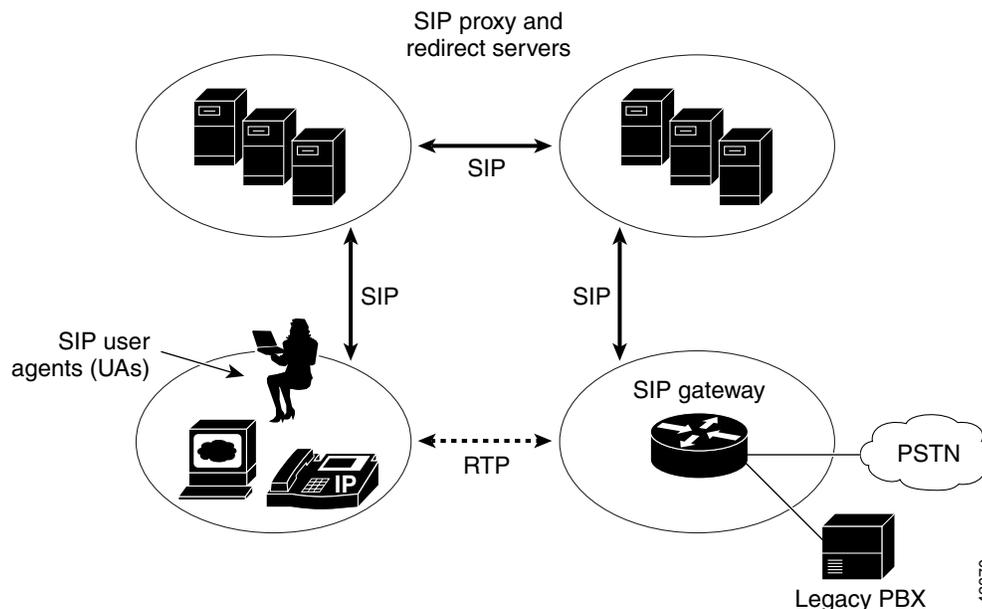
Typically, a SIP endpoint is capable of functioning as both a UAC and a UAS, but functions only as one or the other per transaction. Whether the endpoint functions as a UAC or a UAS depends on the UA that initiated the request.

From an architecture standpoint, the physical components of a SIP network can also be grouped into two categories: clients and servers. [Figure 1-4](#) illustrates the architecture of a SIP network.

**Note**

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

**Figure 1-4 SIP Architecture**



## SIP Clients

SIP clients include the following:

- Phones

Phones act as either a UAS or a UAC. Softphones (PCs that have phone capabilities installed) and Cisco SIP IP phones can initiate SIP requests and respond to requests.

- Gateways

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 side.

## SIP Servers

SIP servers include the following:

- Proxy server

A proxy server receives SIP client messages and forwards them to the next SIP server in the network. Proxy servers can provide functions such as authentication, authorization, network access control, routing, reliable request retransmission, and security.

- Redirect server

A 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 next hop or hops that a message should take, and then the client contacts the next-hop server or UAS directly. Location services are often included with redirect servers and provide contact information and address bindings for called parties.

- Registrar server

A registrar server processes requests from UACs for registration of their current location. Registrar servers are often colocated with a redirect or proxy server.

## BTXML Support

The Cisco SIP IP phone supports Basic Telephony eXtensible Markup Language. BTXML defines XML elements for controlling the user interface of an IP telephone. It describes what information is displayed on the screen and how to provide input using softkeys and hard keys. User-interface control is internal to the phone; there is no external BTXML user interface control.

## Cisco IP Phone Services

The Cisco SIP IP phone supports Cisco IP Phone Services that you configure to provide data such as stock quotes, calendars, and directory lookups. Phone users access this information using phone buttons such as the Services or Directories buttons. See [Chapter 4, “Managing Cisco SIP IP Phones,”](#) for information about configuring these cards.

**Note**

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The Cisco SIP IP phone supports Cisco IP Phone Services version 3.0 only.

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For more information about using XML on your Cisco SIP IP phone, refer to the following:

- IP Telephony  
<http://www.hotdispatch.com/cisco-ip-telephony>
- Cisco CallManager Services Developer Kit  
[http://www.cisco.com/pegi-bin/dev\\_support/access\\_level/product\\_support](http://www.cisco.com/pegi-bin/dev_support/access_level/product_support)
- *Developing Cisco IP Phone Services* by Darrick Deel, Mark Nelson, and Anne Smith, ISBN 1-58705-060-9

## Network Capabilities

The Cisco Unified IP Phone 7960G and 7940G supports the following networking capabilities:

- Telnet support  
Allows you to use Telnet to connect directly to the phone to debug and troubleshoot the phone.
- Ping support  
Allows you to ping from the phone to see if it is operational and assess how long the response time from the phone is.
- Traceroute support  
Allows you to see the path that the signal traverses in the route to its desired destination.

## Configuration Features

With the Cisco Unified IP Phone 7960G and 7940G, you can do the following:

- Configure an Ethernet port mode and speed
- Register with or unregister from a proxy server or backup proxy server
- Specify a TFTP boot directory
- Configure a label for phone-identification display purposes
- Configure a name for caller identification purposes for each active line on a phone
- Configure a 12- or 24-hour user interface time display
- Lock and unlock the phone
- Configure local call forwarding
- Configure autoconnect (intercom)
- Configure speed dialing
- Configure the same directory number (DN) on several lines
- Configure the phone to collect terminating-call data for troubleshooting and billing purposes
- Clear the message-waiting indicator from the console (by means of the command-line interface or by using Telnet in privileged access mode)

- Configure voice activity detection (VAD)

## Signaling Support

The following signaling and transport features are supported:

- G.711 (mu-law and a-law) and G.729a audio compression
- In-band dual tone multifrequency (DTMF) support for codecs for touch-tone dialing
- Out-of-band DTMF signaling (RFC 2833)

## Dial-Plan and Messaging Support

The Cisco SIP IP phone supports a dial plan that enables automatic dialing and generation of a selectable secondary dial tone. The Cisco SIP IP phone also supports the following:

- Local directory configuration (save and recall) and automatic dial completion

Each time a call is successfully made or received, the number is stored in a local directory that is maintained on the phone. The maximum number of entries is 32. Entries are aged out according to usage and age. The oldest entry that has been called the least number of times is overwritten first. This feature cannot be changed by the phone user; however, up to 20 entries can be “locked” (using the Locked softkey) so that they will never be deleted.

- Message waiting indication using unsolicited NOTIFY

When a caller leaves a voice message, a light indicates that a new voice message is in your mailbox. If you listen to the message but do not save or delete the message, the light remains on. If you listen to the new message and save or delete it, the light goes off. The message waiting indicator is controlled by the voice-mail server. The indication is saved over a phone upgrade or reboot.

- Dial-plan support for the asterisk (\*) and pound (#) characters

In previous versions of the dial plan, the “\*” and “#” characters were not matched if specified as dialed digits. The “\*” character was a wildcard equal to “match one or more characters.” The “#” character was interpreted as “dial immediately.” Support was added for the “\*” character by adding “\” as an escape character and the “#” character as a digit in the dial plan. Backward compatibility for the “#” and “\*” characters is supported.

- Distinctive ringing

If the INVITE message from the server contains an Alert-Info header, distinctive ringing is invoked. The format of the header is “Alert-info: x.” The value of “x” can be any number. This header is received only by the phone and is not generated by the phone.

Distinctive ringing is supported when the phone is idle or during a call. In the idle mode, the phone rings with a different cadence. The selected ringing type plays twice with a short pause in between. In call-waiting mode, two short beeps are generated instead of one long beep.

- Configurable and user-selectable dial tone

In earlier releases, specifying a comma (,) in the dial plan caused the phone to play the default secondary dial tone. With this release, specific user-specified tones can play. You can specify up to three different secondary dial tones in a single dial-plan match template. Tones play in the order in which they are listed.

- Ring-back tone for calls on hold

The parameter is selectable from the Settings > Call Preferences menu. When the parameter is enabled, the phone rings if the handset is placed on-hook and there is also a call on hold.
- Dial-plan update support using the Event header in NOTIFY messages.
- Local (180 Ringing) or remote (183 Session Progress) call progress tone.
- Call redirection information support using the Diversion header.

## Routing and Proxy Support

The Cisco SIP IP phone supports the following routing and proxy features:

- Proxy routing

The route attribute in the dial-plan template file can be used to indicate to which proxy—default, emergency, fully qualified domain name (FQDN)—the call should be initially routed. For example, to configure an emergency proxy, specify the value of the route attribute as “emergency.”
- Backup SIP proxy

When the primary proxy does not respond to the INVITE message sent by the phone after the configured number of retries, the phone sends the INVITE to the backup proxy. This is independent from the proxy defined in the route attribute in the dial-plan template used.

The Cisco SIP IP phone attempts to register with the backup proxy. All interactions with the backup proxy, such as authentication challenges, are treated the same as the interactions with the primary proxy. The backup proxy is used only with new INVITE messages that fail to communicate with the primary proxy. Once the backup proxy is used, it is active for the duration of the call.

The location of the backup SIP proxy can be defined as an IP address in the default configuration file.
- Emergency SIP proxy

An optional emergency SIP proxy can be configured with the route attribute of the template tag in the dial-plan template file. When an emergency SIP proxy is configured and a call is initiated, the phone generates an INVITE message to the address specified in the proxy\_emergency parameter. The emergency proxy is used for the entire call.

The location of the emergency proxy can be defined as an IP address in the default configuration file.
- Support of the DNS SRV query

The Domain Name Server (DNS) SRV query is used to locate servers for a given service. SIP on Cisco IP phones uses a DNS SRV query to determine the IP address of the SIP proxy or redirect server. The query string generated is in compliance with RFC 2782 and prepends the protocol label with an underscore (\_), as in “\_protocol.\_transport.” The addition of the underscore reduces the risk of the same name being used for unrelated purposes.

In compliance with RFC 2782 and the draft-ietf-sip-srv-01 specification, the system can remember multiple IP addresses and use them properly. In the draft-ietf-sip-srv-01 specification, it is assumed that all proxies returned for the SRV record are equivalent such that the phone can register with any of the proxies and initiate a call using any other proxy.
- Network Address Translation (NAT) and outbound proxy

Enable or disable NAT with the nat\_enable parameter. You can configure the address of the NAT or firewall server using the nat\_address parameter.

You can configure the IP address and port number of the outbound proxy server. When outbound proxy is enabled, all SIP requests are sent to the outbound proxy server instead of to the proxyN\_address. All responses continue to reconcile the normal Using processing rules. The media stream is not routed through the outbound proxy.

You can enable or disable NAT and outbound proxy modes independently. The received= tag is added to the Via header of all responses if there is no received= tag in the uppermost Via header and the source IP address is different from the IP address in the uppermost Via header. Responses are sent back to the source under the following conditions:

- If a received= tag is in the uppermost Via header, the response is sent back to the IP address contained in the received= tag.
- If there is no received= tag and the IP address in the uppermost Via header is different from the source IP address, the response is sent back to the source IP address. Otherwise the response is sent back to the IP address in the uppermost Via header.

**Note**

For information on how to use the standard telephony features and URL dialing, refer to the documents listed in the [“Related Documentation” section on page ix](#).

## Supported Languages and Character Set

The Cisco SIP IP phone supports the ISO 8859-1 Latin1 characters and the following languages: French (fr), Spanish (es), Catalan (ca), Basque (eu), Portuguese (pt), Italian (it), Albanian (sq), Rhaeto-Romanic (rm), Dutch (nl), German (de), Danish (da), Swedish (sv), Norwegian (no), Finnish (fi), Faroese (fo), Icelandic (is), Irish (ga), Scottish (gd), English (en), Afrikaans (af), and Swahili (sw).

It does not support the following languages: Zulu (zu) and other Bantu languages that use Latin Extended-B letters; Arabic in North Africa; and Guarani (gn), which is missing the letters G, E, I, U, and Y with tildes (~).

**Note**

The XML cards, information text, and menus are all in English. These items are built into the phone image and cannot be changed.

You can use ISO 8859-1 Latin1 characters in the following areas:

- Caller ID information. When a SIP message is received with ISO 8859-1 Latin1 characters in the caller ID strings, those caller ID strings are displayed on the phone LCD with the correct ISO 8859-1 Latin1 characters.
- Services menu applications written in Cisco IP Phone Services. You can develop language-specific applications for a particular region. For example, an application that displayed the current weather in Sweden using Swedish characters can be displayed on the Cisco SIP IP phone. If you develop the same application for a Spanish locale, the application can be translated into Spanish.
- Line key labels. The line keys can be configured to support the Latin1 characters. You can specify the line key name in the configuration file, and it displays correctly. The Latin1 characters cannot be used in the linex\_name parameter, but can be used in the linex\_shortcode and linex\_displayname parameters. If the proxy supports Latin1 characters in the To/From headers, they can be used in the linex\_name parameter as well.

**Note**

The *i* button text and the Settings menus are in English. These items are built into the phone image and cannot be changed.

## Supported Protocols

The Cisco SIP IP phone supports the protocols listed in [Table 1-1](#).

**Table 1-1 Supported Protocols**

Protocol	Description
AVT	Audio video transport. Handles payload negotiation.
DHCP	Dynamic Host Configuration Protocol. Dynamically allocates and assigns IP addresses. DHCP allows you to move network devices from one subnet to another without administrative attention. It allows connection of Cisco SIP IP phones to the network so that they become operational without having to manually assign an IP address and additional network parameters. DHCP option 60 allows the Cisco SIP IP phone to identify itself with vendor-specific information. The Cisco SIP IP phone complies with DHCP specifications documented in RFC 2131. By default, the phone is DHCP-enabled.
DNS	Domain Name System. Translates names of network nodes into addresses. SIP uses DNS to resolve the host names of endpoints into IP addresses.
Dynamic DNS and TFTP	You can configure additional DNS and TFTP servers. Upon bootup, the phone first goes to the default TFTP server to download the configuration files. If a new dynamic TFTP server is specified in the files, the phone requests a new set of files from the specified server. If new DNS addresses are specified in the files, the phone uses those addresses for lookups.
HTTP	Hypertext Transfer Protocol. The phone contains limited support for HTTP 1.1. The Cisco SIP IP phone uses HTTP to retrieve Cisco CallManager XML files.
ICMP	Internet Control Message Protocol. 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 IP phone supports ICMP as defined in RFC 792.
IP	Internet Protocol. 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 defined in RFC 791.
RTP	Real-Time Transport Protocol. Supports transport of real-time data (such as voice) over data networks. RTP also has the ability to obtain quality-of-service (QoS) information. The Cisco SIP IP phone supports RTP as a media channel.

**Table 1-1 Supported Protocols (continued)**

Protocol	Description
SDP	Session Description Protocol. An ASCII-based protocol that describes multimedia sessions and their related scheduling information. Third-party call control is supported using delayed media negotiation, which is SDP data that is not completely advertised in the initial call setup. SDP also supports endpoints specified as fully qualified domain names (FQDNs). The Cisco SIP IP phone uses SDP for session description.
SNTP	Simple Network Time Protocol. Synchronizes computer clocks on an IP network. Current date and time are supported using SNTP including time zone and daylight saving time. The Cisco SIP IP phone uses SNTP for date and time support.
TCP	Transmission Control Protocol. Provides a reliable byte-stream transfer service between endpoints on the Internet. The Cisco SIP IP phone supports TCP for Telnet sessions only.
TFTP	Trivial File Transfer Protocol. 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.
ToS	Type of service. An indication of how an upper-layer protocol requires a lower-layer protocol to treat its messages. In SNA subarea routing, ToS definitions are used by subarea nodes to determine the optimal route to establish a given session. A ToS definition comprises a virtual route number and a transmission priority field. Also called class of service (CoS).
UDP	User Datagram Protocol. Exchanges data packets without acknowledgments or guaranteed delivery. A SIP network can use UDP as the underlying transport protocol. If UDP is used, retransmissions are used to ensure reliability. UDP fragmentation is supported. The Cisco SIP IP phone supports UDP as defined in RFC 768 for SIP signaling.
VAD	Voice activity detection. When enabled on a voice port or a dial peer, silence is not transmitted over the network, only audible speech. Sound quality is slightly degraded, but the connection monopolizes much less bandwidth.

## Where to Go Next

Use the following information to determine where to go next in this guide:

- See [Chapter 2, “Installing Cisco Unified IP Phone 7960G and 7940G Hardware on the Desktop or Wall,”](#) for placement of the phone on the desktop or wall and cabling information.
- See [Chapter 3, “Initializing Cisco Unified IP Phones,”](#) for information on installing firmware, customizing configuration files, and connecting the phone to power sources and the network.
- See [Chapter 4, “Managing Cisco SIP IP Phones,”](#) for information on upgrading firmware and performing other management tasks.
- See [Chapter 5, “Monitoring Cisco SIP IP Phones,”](#) for information on debugging and on viewing network statistics.