Product Overview

This chapter contains the following information about the Cisco MGCP IP Phone:

What Is Media Gateway Control Protocol?

Media Gateway Control Protocol (MGCP) is the Internet Engineering Task Force's (IETF's) standard for multimedia conferencing over IP. MGCP is an ASCII-based, application-layer control protocol (defined in RFC 2705) that can be used to establish, maintain, and terminate calls between two or more endpoints.

Like other VoIP protocols, MGCP 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.

One aspect of MGCP that differs from other VoIP protocols is that MGCP relies on a control server, or call agent (CA) to control call progression, tones to apply, and call characteristics. MGCP endpoints carry out instructions from the CA, which controls how calls proceed.

MGCP provides the capabilities to allow a Control Server to:

- Determine the location of the target endpoint.
- Determine the media capabilities of the target endpoint. Using Session Description Protocol (SDP), MGCP determines the lowest level of common service 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.
- Establish a session between the originating and target endpoint. If the call can be completed, MGCP establishes a session between the endpoints. MGCP also supports mid-call changes, such as the addition of another endpoint to the conference or the changing of a media characteristic or codec.

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

MGCP is a client-server protocol. The CA handles all aspects of setting up calls to and from endpoints. CAs or control servers provide the feature capabilities that a particular endpoint will be able to use. Endpoints connected to different CAs will likely have a different set of features they can use. Since all of the call control features are in the control server, each control server vendor decides which features are most important, and therefore different control server vendors differ in "essential features."

What Is the Cisco MGCP IP Phone?

The Cisco Media Gateway Control Protocol (MGCP) Internet Protocol (IP) Phone provides voice communication over an IP network. It functions much like a traditional phone, allowing you to place and receive telephone calls.

The Cisco MGCP IP phone works with a third-party CA that uses MGCP for call control and Basic Telephony eXtensible Markup Language (BTXML) for control of the phone's displays and feature keys. This document describes the phone features that are controlled by the phone. Refer to your CA documentation for descriptions of all other phone features, displays, and applications.

Cisco MGCP 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 MGCP 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.

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

Figure 1-1   Cisco MGCP IP Phone Physical Features

The main components of the Cisco MGCP IP Phone are defined in Table 1-1.

Table 1-1   Cisco MGCP IP Phone Features

Feature	Description
LCD screen	The IP Phone "desktop," which displays information such as the time, date, your phone number, caller ID, and line/call status. Refer to your call agent or service provider documentation for the LCD functionality.
Line or speed-dial buttons	Provide additional dialing capabilities, such as opening a new line, speed- dialing the number on the LCD screen, or ending a call. Refer to your call agent or service provider documentation for line and speed-dial button functionality.
Footstand adjustment	Adjusts the angle of the phone base.
Soft keys	Provide additional functions for your phone. Refer to your call agent or service provider documentation for soft keys functionality.
i button	Provides additional functions for your phone. Refer to your call agent or service provider documentation for *i* button functionality.
	Can provide online help for selected keys or features and network statistics about the active call. Displays a descriptor of the key pressed directly after pressing the *i* button. For example, pressing the *i* button, then up or down displays a screen instructing you how to scroll up and down on the LCD
Messages mode button	Refer to your call agent or service provider documentation for Messages mode button functionality.
Directory mode button	Refer to your call agent or service provider documentation for Directory mode button functionality.
Services mode button	Refer to your call agent or service provider documentation for Services mode button functionality.
Settings mode button	Provides access to phone settings such as contrast and ring type and to network configuration and status information.
Volume buttons	Increase or decrease the volume for the currently active voice receiver: handset, headset, or speakerphone. The volume keys also control the ringer volume (if onhook), and the contrast of the LCD.
Function toggles	Toggle the headset, mute, and speaker functions on and off.
Scroll key	Enables you to move among different options displayed on the LCD screen.
Dialing pad	Press the dialing pad buttons to dial a phone number. Dialing pad buttons work exactly like those on your existing telephone. Refer to your call agent or service provider documentation for any additional dialing pad functions.
Handset	Lift the handset and press the dialing pad numbers to place a call, answer a call, and operate other phone functions.

BTXML Support

Basic Telephony eXtensible Markup Language is supported on the Cisco MGCP IP phone. BTXML defines XML elements for controlling the user interface of an IP telephone. BTXML describes what information is displayed on the screen and how the user provides input using soft keys and hard keys.

For information about creating and using BTXML scripts on the Cisco MGCP IP phone, see the *Cisco IP Phone MGCP BTXML Version 2.0 Application Development Guide*.

Cisco CallManager XML Support

The Cisco MGCP IP phone supports customer-written Cisco CallManager XML cards that can be accessed using buttons or softkeys on the phone. These cards can provide data such as stock quotes, calendars, and directory lookups. The XML cards can be accessed by the following methods:

- From the **Services** soft key, configured using the services_url parameter.

- By pressing the **directory** button and selecting **External Directory**, configured using the directory_url parameter.

- By specifying a bitmap to be used as the phone's logo (branding), configured using the logo_url parameter.

See Chapter 4, "Configuring the Cisco MGCP IP Phone" for information about configuring these parameters.

The Cisco MGCP IP phone supports Cisco CallManger XML up to version 3.0. It does not support the XML objects added in Cisco CallManager XML version 3.1:

- CiscoIPPhoneIconMenu

- CiscoIPPhoneExecute

- CiscoIPPhoneError

- CiscoIPPhoneResponse

The following exceptions apply to the Cisco MGCP IP phone:

- External directories cannot be appended to the main list of directories under the **directory** button. If external directories are provisioned for the Cisco MGCP IP phone, then they can be accessed by pressing the **directory** button and selecting the **External Directory** option.

- The Cisco MGCP IP phone removes white space when the Cisco CallManager XML cards are displayed. Multiple spaces are consolidated to a single space.

- Setting x and y coordinates for the CiscoIPPhoneImage object is not supported. The image always appears at location 0,0. Centering of the image is not supported if x and y are set to -1.

- The Cisco MGCP IP phone displays any valid title it receives. This differs from the Cisco CallManager phones in that the CiscoIPPhoneGraphicMenu object does not display a title even if it receives one and the CiscoIPPhoneImage object displays the previous menu item or "Services" rather than received titles.

- CallManager phones allow embedded carriage returns and line feeds in menu items. In the Cisco MGCP IP phone, carriage returns and line feeds are discarded.

- The Cisco MGCP IP phone always displays the full set of directory softkeys. For Cisco CallManager phones, the softkeys can change depending on what type of object it receives. This is due to support for Cisco CallManager 3.0.

- A parameter is sent along with the initial request for a Services or Directory URL which differentiates the Cisco MGCP IP phone from other types of phones.

For more information about using XML on your Cisco MGCP IP phone, see the following links or documents:

- Cisco IP Phone Service forum at the following URL:

  http://www.hotdispatch.com/cisco-ip-telephony

- Cisco Call Manager Services Developer Kit at the following URL:

  http://www.cisco.com/warp/public/570/avvid/voice_ip/cm_xml/cm_xmldown.shtml

- *Developing Cisco IP Phone Services* by Darrick Deel, Mark Nelson, and Anne Smith, ISBN
  1-58705-060-9

Supported Features

The MGCP phone supports the following features. Depending on the features that your CA supports, some of these may not be available on your phone.

Physical Features

- Hearing-aid compatible handset
- Headset compatibility
- Integrated two-port Ethernet switch that allows the telephone and a computer to share a single Ethernet jack
- Direct connection to a 10BaseT or 100BaseT Ethernet (RJ-45) network (half- or full-duplex connections are supported)
- Large (4.25 x 3 in.) display with adjustable contrast

Network Features

- Interoperability with third-party CAs.
- Up to six MGCP connections and call appearances.
- IP address assignment—Dynamic Host Configuration Protocol (DHCP) client or manually configured via a local setup menu
- Network startup using DHCP and Trivial File Transfer Protocol (TFTP)
- Domain Name System (DNS): The MGCP phone uses DNS lookups to locate its communication partners (for example, a TFTP server or a CA. If a CA is unreachable, the MGCP phone queries the DNS server for an alternate CA.)
- Dynamic DNS and TFTP servers: You can configure additional DNS and TFTP servers in the configuration files. 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.
- Telnet support—Allows the user to use telnet to connect directly to the Cisco MGCP IP Phone to debug and troubleshoot the phone. See "Configuring the Cisco MGCP IP Phone" section on page 4-1 for more information on configuration parameters.
- Ping support—Allows the user to use ping to see if a Cisco MGCP IP Phone is operational and how long the response time is from the phone.
- Traceroute support—Allows the user to use traceroute to see the path that a Cisco MGCP IP phone traverses in the route to its desired destination.

- Remote reset support—Allows a service provider to reset a phone from a remote site. This feature provides a key tool for restarting the phone's registration process with the provider's call agent or proxy and for receiving a new or updated configuration or firmware load from a designated TFTP server.

Codec and Protocol Support

- Basic phone service including MGCP 0.1 and MGCP 1.0 headers
- G.711 u-law codec
- G.711 a-law codec
- G.729a codec
- DTMF out-of-band for G.729a codec compliant to RFC 2833
- Local Connection Options (LCOs):
    - G.729a codec: supports low-bandwidth access for multitenant deployment
    - Voice activity detection (VAD) support
    - Type of service (ToS) bit for Real-Time Transport Protocol (RTP)
- XML support
- Hypertext Transfer Protocol (HTTP)—The phone contains limited support for HTTP 1.1. The phone uses HTTP to retrieve Call Manager XML files.

Dialing and Messaging Features

- Message Waiting Indication—Lights to indicate that a new voice message is in a subscriber's mailbox. If the subscriber listens to the message but does not save or delete the message, the light remains on. If a subscriber listens to the new message or messages, and saves or deletes them, the light goes off. The message waiting indicator (MWI) is controlled by the voicemail server. The indication will be saved over a phone upgrade or reboot.
- Notified entity: A CA can direct an endpoint to send notify messages to an alternate destination.
- Call waiting, call transfer, call forward (unconditional, busy, no answer), announcement, music on hold, and volume control. (Must be supported by the CA.)
- Three-way calling using an external multipoint control unit (MCU). (Must be supported by the CA.)

Supported Protocols

The Cisco MGCP IP phone supports the following standard protocols:

- Domain Name System (DNS)—Used in the Internet for translating names of network nodes into addresses. The MGCP IP Phone uses DNS to resolve the host names of endpoints to IP addresses.
- Dynamic Host Control Protocol (DHCP)—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 MGCP IP phones to the network and become operational without having to manually assign an IP address and additional network parameters.

    The Cisco MGCP IP phone complies with the DHCP specifications documented in RFC 2131. By default, Cisco MGCP IP phones are DHCP-enabled.

- Internet Control Message Protocol (ICMP)—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 MGCP IP phone supports ICMP as it is documented in RFC 792.

- Internet Protocol (IP)—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 MGCP IP phone supports IP as it is defined in RFC 791.

- Real-Time Transport Protocol (RTP)—Transports real-time data (such as voice data) over data networks. RTP also has the ability to obtain Quality of Service (QoS) information.

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

- Session Description Protocol (SDP)—An ASCII-based protocol that describes multimedia sessions and their related scheduling information.

  The Cisco MGCP IP phone uses SDP for session description.

- Simple Network Time Protocol (SNTP)—Synchronizes computer clocks on an IP network. The Cisco MGCP IP phones use SNTP for their date and time support.

- Transmission Control Protocol (TCP)—Provides a reliable byte-stream transfer service between two endpoints on an Internet. The Cisco IP Phone 7960 supports TCP for Telnet and HTTP sessions.

- Trivial File Transfer Protocol (TFTP)—Allows files to be transferred from one computer to another over a network. The Cisco MGCP IP phone uses TFTP to download configuration files and software updates.

- User Datagram Protocol (UDP)—A simple protocol that exchanges data packets without acknowledgments or guaranteed delivery. MGCP can use UDP as the underlying transport protocol. If UDP is used, retransmissions are used to ensure reliability.

Character Support

The Cisco MGCP IP phone supports the ISO 8859-1 Latin1 characters. The following languages are supported:

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).

The following languages are not supported:

Zulu (zu) and other Bantu languages using Latin Extended-B letters, Arabic in North Africa, and Guarani (gn) missing GEIUY with ~ tilde.

Note   The info key text and the Settings menu are in English. These items are built into the phone image and cannot be changed.

ISO 8859-1 Latin1 characters can be used in the following areas:

- Caller ID information. When an MGCP message is received with ISO 8859-1 Latin1 characters in the caller ID strings, those caller ID strings are displayed on the Cisco SIP IP phone's LCD screen with the correct ISO 8859-1 Latin1 characters.

- Services menu applications written in CMXML. The customer can develop language-specific applications for a particular region. For example, an application that displayed the current weather in Sweden using Swedish language characters can be displayed on the Cisco MGCP IP phone. If the customer develops the same application for a Spanish town, they could translate the application into Spanish.

- Call control displays (external MGCP XML card deck). The XML deck used for MGCP call control is downloaded to the phone, and those XML cards can be translated into the local language.

> **Note** The Info key text and the test contained in the Settings menu are displayed in English.

- Line key labels. Line key labels are set using an MGCP message that sets the label to the string provided. Strings containing ISO 8859-1 Latin1 characters are displayed properly.

- Soft key labels. Soft key labels are set using XML cards. XML cards, including call controls cards and services applications, can provide soft keys with ISO 8859-1 Latin1 characters.