



Using the Cisco Voice Gateway 200

This chapter describes how to use the Cisco Voice Gateway 200 (VG200) with Cisco CallManager 3.0 or later. It introduces the Media Gateway Control Protocol (MGCP) and describes how MGCP supports redundant Cisco CallManager servers. This chapter then provides an overview of the configuration required to make this work. This chapter includes the following topics:

- Voice over IP Overview, page 1-1
- Using the Cisco VG200 with Cisco CallManager, page 1-6
- Using the Cisco VG200 with MGCP, page 1-7
- Using a Cisco VG200 Gateway with H.323, page 1-8
- Where to Go Next, page 1-14

Voice over IP Overview

Voice over IP (VoIP) enables your gateway to carry live voice traffic (for example, telephone calls and faxes) over an IP network. VoIP can consolidate voice and data traffic for more efficient use of bandwidth, reduce toll charges, and enable alternatives to expensive and proprietary PBX systems.

The Cisco VG200 is an important component of a VoIP solution, providing a simple and inexpensive interface between an Ethernet data network and the Public Switched Telephone Network (PSTN).

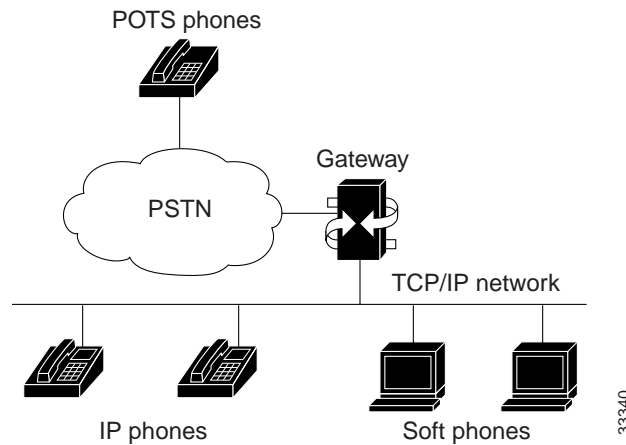
This section covers the following topics, describing in a general way how VoIP works and defining the standards supported by the Cisco VG200 used to implement a VoIP network.

- Implementing a VoIP Network, page 1-2
- The H.323 Standard, page 1-3
- Media Gateway Control Protocol, page 1-5

Implementing a VoIP Network

To implement VoIP, you must first establish a working IP network and ensure that the network is properly tuned to reduce congestion. Voice traffic is more sensitive to congestion than data traffic, meaning that users will notice more congestion when using VoIP telephones than when using IP for other purposes. For more information about configuring IP, refer to the “Configuring IP” chapter in the *Cisco IOS 12.0 Network Protocols Configuration Guide*.

You also need to thoroughly understand your company’s existing voice network and dial plan, which is the pattern of dialed numbers that provides access to different telephony endpoints. You can then begin the process of integrating your voice and data networks, which should be done cautiously and systematically to prevent disruption of existing telephone service.

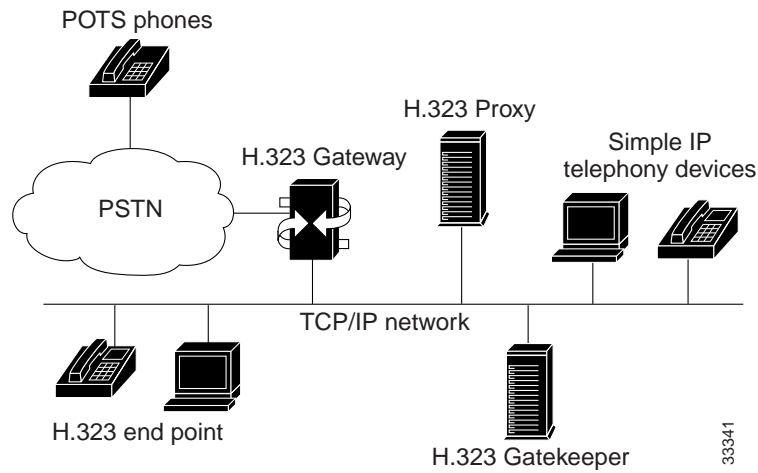
Figure 1-1 Typical VoIP Network

In a VoIP network, illustrated in Figure 1-1, it is possible for IP telephony devices to interoperate directly. However, to connect to the PSTN, an intermediary device, called a gateway, is required. A VoIP gateway, such as the Cisco VG200, allows users of IP phones and PC-based soft phones to exchange calls with users of plain old telephone service (POTS) phones on the PSTN. The gateway translates between the signals used on the PSTN and the IP packets used to transmit data on a TCP/IP network.

The H.323 Standard

The H.323 standard describes a method for converting between voice and data transmission formats and for managing connections between telephony endpoints. H.323 is actually a collection of protocols that define standard methods for interconnecting H.323 endpoints (sometimes called terminals) and POTS devices.

Figure 1-2 The H.323 Standard



H.323 endpoints can communicate among each other without the intervention of any other devices, however to connect them to the PSTN, to manage call setup and network bandwidth, and to simplify administration, the following devices are often used:

- H.323 gateway—Connects H.323 devices to POTS devices on the PSTN
- H.323 gatekeeper—Manages network bandwidth and provides a central point for call administration
- H.323 proxy—An H.323 endpoint that acts as an intermediary between H.323 endpoints and other IP telephony devices

To standardize communication among these devices and endpoints, H.323 utilizes the following protocols:

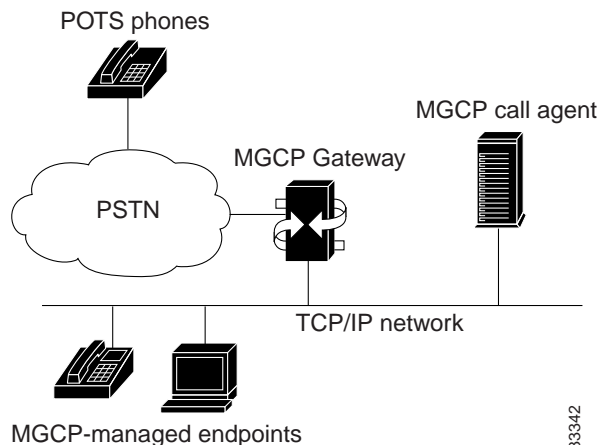
- H.245—Used to negotiate channel usage and capabilities
- H.225—Used for call signaling and call setup
- Registration, Admission and Status (RAS)—Used to communicate with the H.323 gatekeeper
- Real-Time Protocol/Real-Time Control Protocol (RTP/RTCP)—Used to create and transmit audio packets on the IP data network

H.323 is a peer-to-peer protocol, helping to establish and manage connections among H.323 endpoints. As with other peer-to-peer protocols, the many-to-many relationships become difficult to configure and maintain in large networks. MGCP was developed specifically to overcome the problem of scalability inherent in the H.323 protocol.

Media Gateway Control Protocol

MGCP establishes a master-slave protocol which, when compared to H.323, simplifies voice network administration and improves reliability and performance. In particular, it makes configuration and administration of gateway devices easy. With MGCP, gateways are defined by MGCP as slave devices in relation to the master, referred to as a *call agent*, which manages connections between endpoints and controls how gateways function.

Figure 1-3 MGCP



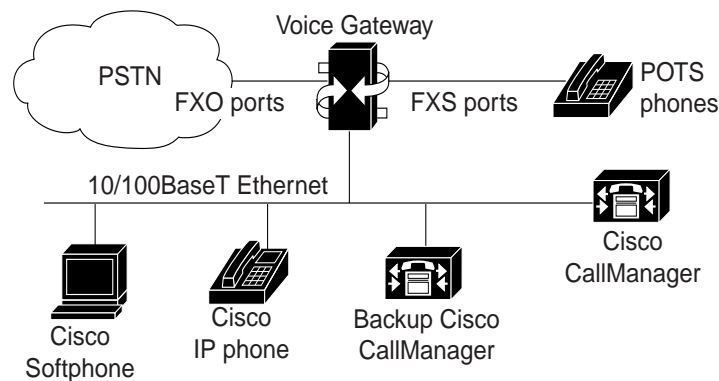
Because an MGCP gateway is a slave device, it derives most of the configuration it requires from the call agent. To configure an MGCP gateway, all you need to do is to identify the call agent on the gateway and identify the gateway to the call agent. In addition, MGCP optionally supports multiple call agents, which can eliminate a potential single point of failure in the voice network.

Using the Cisco VG200 with Cisco CallManager

When used with Cisco CallManager 3.0 or later as a call agent, the Cisco VG200 functions as an MGCP gateway. The Cisco VG200 supports MGCP only on its Foreign Exchange Station (FXS) and Foreign Exchange Office (FXO) analog ports, while it supports H.323 on its T1 digital ports.

As shown in Figure 1-4, a Cisco VG200 gateway allows you to connect standard POTS telephones (connected directly to the gateway or anywhere on the PSTN) with Cisco IP SoftPhones, Cisco IP Phones, or any H.323-compliant telephony devices. The Cisco VG200 provides a 10/100BaseT Ethernet port for connection to the data network and a number of different telephony interfaces, depending on the hardware and software configuration you choose.

Figure 1-4 Using the Cisco VG200 with Cisco CallManager



Currently, the Cisco VG200 supports the following types of analog ports and connections:

- 1 to 4 FXO ports for connecting to a central office or PBX
- 1 to 4 FXS ports for connecting to POTS telephony devices
- 1 or 2 BRI ports for connecting to ISDN
- 1 or 2 E1/T1 digital ports:
 - For connecting to the PSTN using FXO emulation
 - For connecting to a T1 channel bank using FXS emulation

- For connecting to a PBX by means of a trunk (tie) line using Ear and Mouth (E&M) emulation

These ports can be used to integrate a VoIP network with POTS devices, PBXs, ISDN, or the PSTN.

The key to integrating voice and data networks using a Cisco VG200 gateway is Cisco CallManager, which handles call requests and establishes connections between IP devices placing and receiving calls.

When you use a Cisco VG200 gateway with H.323, Cisco CallManager can be used like an H.323 proxy. You can identify the Cisco CallManager server as the only H.323 endpoint known to the Cisco VG200 gateway on its Ethernet (LAN) port, so you do not have to configure and maintain a complex dialing plan on the gateway. This also allows the use of less intelligent IP telephony devices within an H.323 voice network, because not every device has to function as an H.323 endpoint.

You configure the Cisco VG200 gateway with information about Cisco CallManager, using the Cisco IOS command line interface (CLI). The procedures and commands required to perform this configuration are described in Chapter 3, “Configuring Voice over IP.”

When used with MGCP, you must also configure the Cisco CallManager server with information about each port on a Cisco VG200 gateway, using the Cisco CallManager Administrator, a Web-based graphic user interface (GUI). The *Cisco CallManager Administration Guide* describes the procedures for performing this configuration.

Using the Cisco VG200 with MGCP

To use a Cisco VG200 gateway with MGCP, you must use Cisco CallManager as the call agent, and you currently must use an analog voice network module and FXS or FXO voice interface cards. You use the Cisco IOS CLI to enable MGCP on the gateway and to identify the Cisco CallManager server. Cisco CallManager then assumes control over establishing and tearing down connections between IP endpoints on your network and endpoints connected through the PSTN.

When using MGCP with a Cisco VG200 gateway, all dial-plan related configuration elements are controlled by Cisco CallManager, and should not be configured in the Cisco VG200 gateway for MGCP-managed endpoints.

When you use a Cisco VG200 gateway with Cisco CallManager, you can use its redundant call agent feature to eliminate a potential single point of failure in the VoIP network. This feature requires that you have two or three Cisco CallManager servers available on your network and that you identify each of these servers using CLI commands. You identify the primary Cisco CallManager server as the MGCP call agent, and identify the second and third Cisco CallManager servers as backup servers.

Once the redundant call agent feature is configured, if the primary Cisco CallManager server becomes unavailable, the Cisco VG200 gateway will register with the backup Cisco CallManager server. The Cisco VG200 gateway monitors MGCP packets sent by the Cisco CallManager and when no such traffic is detected, it sends keepalive packets to which the Cisco CallManager server should respond. If the Cisco VG200 gateway does not detect any packets from the Cisco CallManager for a specified period, it will try to establish a new connection with the backup Cisco CallManager server.

Another feature of the Cisco VG200 is called *switchback*, which refers to the way that the gateway reestablishes a connection with the primary Cisco CallManager server when the primary server becomes available again. You can configure the Cisco VG200 to reestablish connection immediately, or to wait for a specified length of time to ensure greater stability in the voice network.

Using a Cisco VG200 Gateway with H.323

The Cisco VG200 functions as an H.323 gateway with the following voice interface types, which are currently not supported with MGCP:

- E1 (R2 and E&M analog emulation)
- T1-CAS (FXO, FXS, and E&M emulation)
- E1-PRI
- T1-PRI
- ISDN/BRI
- E&M

Note that H.323 configuration is more complicated compared to MGCP and does not provide for redundant call agents. The following sections describe how to use the Cisco VG200 with H.323 with various interface and signalling types:

- T1-CAS Analog Port Emulation, page 1-9
- FXO Emulation, page 1-10
- FXS Emulation, page 1-11
- E&M Emulation, page 1-11
- Linking PBX Users with E&M Trunk Lines, page 1-12
- Direct-Inward Dialing on a BRI Port, page 1-13

T1-CAS Analog Port Emulation

T1 is a digital signaling standard that uses time-division multiplexing (TDM) to divide the available bandwidth of 1.544 Mbps into twenty-four 64-kbps timeslots. E1 is a similar standard, commonly used in Europe and many parts of Asia, which provides 2.048 Mbps bandwidth, divided into 32 time slots, with 30 time slots available for voice conversations.

A DS-0 group is a group or collection of DS-0 timeslots. Each of these DS-0 groups can be used to emulate an FXO analog connection to the PSTN, an FXS analog connection to a T1 channel bank, or an E&M connection for enabling a trunk (tie) line.

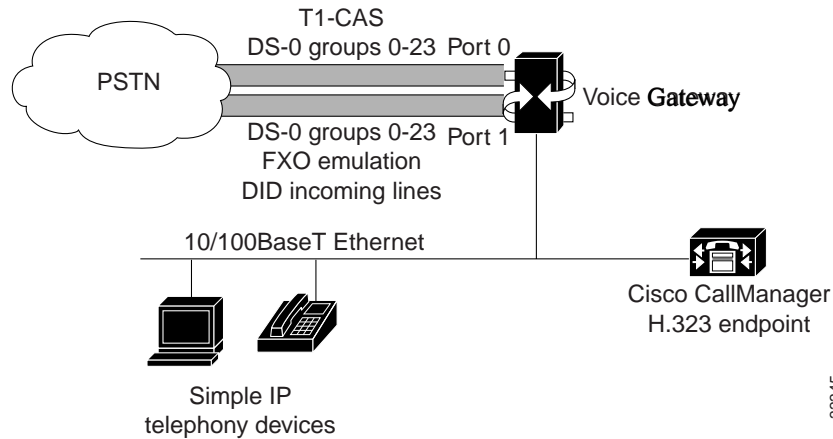
When you configure your DS-0 groups, you can configure DS-0 hunt groups, which allow any available voice port to be used for a range of phone numbers. This eliminates the chance that a single DS-0 could fail, or that a DS-0 is busy at the time of a call.

To use a Cisco VG200 gateway to interconnect a VoIP network and a T1 connection, you configure from 1 to 24 DS-0 groups for each T1 connection, and then define a voice port for each DS-0 group. The Cisco VG200 currently supports T1 connections using channel associated signaling (CAS).

You can use DS-0 groups with FXO analog port emulation to connect to a PSTN central office (CO). When connecting T1-CAS to the PSTN, either the Cisco VG200 gateway or the Cisco CallManager must be able to identify the dialed number (DNIS) of each call and route it to the appropriate endpoint on the IP network.

FXO Emulation

Figure 1-5 Using FXO Analog Emulation with T1-CAS



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To make this work you must obtain Direct Inward Dialing (DID) service from your local exchange carrier (LEC) on the DS-0 groups used for incoming calls. DID service is provided only on incoming connections, so to use it, you must divide your T1 channels into incoming and outgoing DS-0 groups.

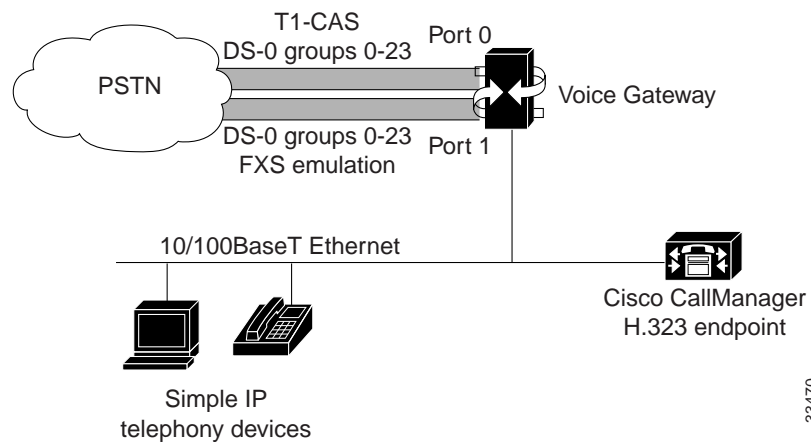
You can configure your dial plan on a Cisco VG200 gateway, using Cisco IOS CLI commands to identify the destination endpoint and coder-decoder (CODEC) for each incoming call on each DS-0 group. This is a straightforward task, but this method of configuration does not take full advantage of the Cisco CallManager and can only be used for H.323 endpoints.

You can configure a Cisco VG200 gateway to direct all incoming calls to the Cisco CallManager, and configure your dial plan using the Cisco CallManager Administrator. This simplifies administration and lets you connect simple IP telephony devices (such as the Cisco IP Phone 7960 and Cisco IP SoftPhone) to the PSTN. This configuration is described in Chapter 3, “Configuring Voice over IP.”

FXS Emulation

As shown in Figure 1-6, you can use FXS analog signalling with the T1-CAS port option to interconnect the PSTN and a Voice over IP network. In this configuration, you define the DS-0 groups as FXS and create a dialing plan to ensure that calls can be routed and set up between endpoints on the VoIP network and the POTS devices on the T1 channel bank.

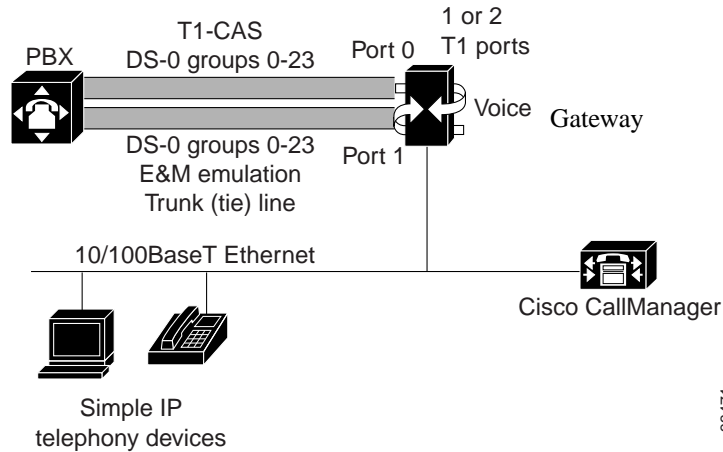
Figure 1-6 Using FXS Analog Emulation with T1-CAS



E&M Emulation

As shown in Figure 1-7, you can use the T1-CAS port on a Cisco VG200 gateway with E&M analog emulation. E&M signalling emulation lets you use a Cisco VG200 gateway to interconnect H.323 devices on a VoIP network with POTS devices over a standard trunk (tie) line.

Figure 1-7 Using E&M Analog Emulation with T1-CAS



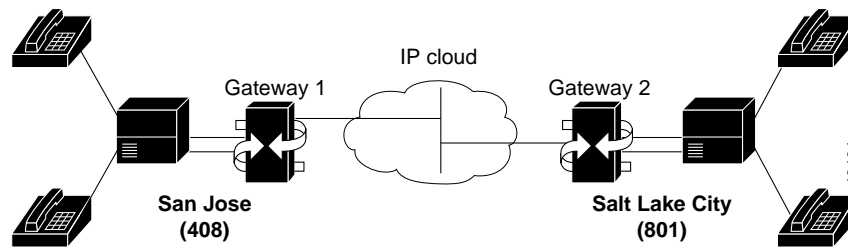
Linking PBX Users with E&M Trunk Lines

The following example shows how to configure VoIP to link PBX users with E&M trunk lines.

In this example, a company wants to connect two offices: one in San Jose, California and the other in Salt Lake City, Utah. Each office has an internal telephone network using PBX, connected to the voice network by an E&M interface. Both the Salt Lake City and the San Jose offices are using E&M Port Type II, with four-wire operation and Immediate Start signaling. Each E&M interface connects to the gateway using two voice interface connections. Users in San Jose dial “8-569” and then the extension number to reach a destination in Salt Lake City. Users in Salt Lake City dial “4-527” and then the extension number to reach a destination in San Jose.

Figure 1-8 illustrates the topology of this connection example.

Figure 1-8 Linking PBX Users with E&M Trunk Lines Example



This example assumes that the company already has established a working IP connection between its two remote offices. PBXs should be configured to pass all DTMF signals to the gateway. Cisco recommends that you do not configure “store-and-forward” tone.



Note

If you change the gain or the telephony port, make sure that the telephony port still accepts DTMF signals.

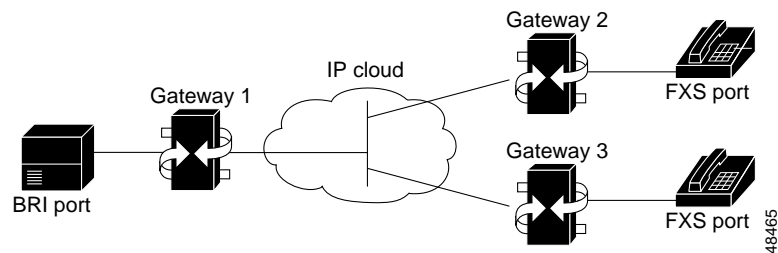
Direct-Inward Dialing on a BRI Port

The following example shows how to configure a BRI port for direct-inward dialing (DID). This configuration allows the called number information from the ISDN Q.931 setup message to be used for routing on an ISDN line.

In this example, a call comes in to gateway 1 on the BRI port. The DID information allows the gateway to route the call based on the called number. If the called number is 2xxx, the call is routed to gateway 2000, and if the called number is 3xxx, the call is routed to gateway 3000.

Figure 1-9 illustrates the topology of this connection example.

Figure 1-9 Configuring DID on a BRI Port



Where to Go Next

Now that you understand something about the different ways that you can use a Cisco VG200 gateway, you should proceed to Chapter 2, “Basic Configuration.” Chapter 2 describes the procedures required to configure your Cisco VG200 gateway, however you choose to use it. Even if you are familiar with other Cisco products, you should refer to Chapter 2 because the configuration of a Cisco VG200 gateway is a little different from Cisco routers.

After completing the basic configuration required to use a Cisco VG200 gateway, you can refer to Chapter 3, “Configuring Voice over IP,” for detailed instructions about configuring a Cisco VG200 gateway for MGCP or H.323.