CHAPTER 32

Understanding Voice Gateways

Cisco IP telephony gateways enable Cisco CallManager to communicate with non-IP telecommunications devices. Cisco CallManager supports several types of voice gateways.

This section covers the following topics:

Gateway Control Protocols and Trunk Interfaces

This section describes the gateway control protocols and trunk interface protocols that are supported for configuring gateways in Cisco CallManager.

- Gateway control protocols provide the internal interface between the voice gateway and Cisco CallManager.
- Trunk interfaces specify how the gateway interfaces with the PSTN or other external devices.

Gateway Control Protocols

Gateway control protocols provide communication and control between Cisco CallManager and the voice gateway.

The amount and type of information you configure in Cisco CallManager Administration versus what is configured on the gateway varies, depending on whether gateway control protocol is MGCP, H.323, or Skinny:

- **Media Gateway Control Protocol (MGCP)**—Gateways that support MGCP as of Cisco CallManager Release 3.1 include the Cisco VG200, Cisco 3600 and 2600, Cisco Catalyst 6000 8 Port Voice E1/T1 and Services modules, Cisco Catalyst 4000 Access Gateway Module, Cisco DE-30+, and Cisco DT-24+.

  When MGCP is used, the Cisco CallManager controls routing and tones and provides supplementary services to the gateway. MGCP provides call preservation (calls are maintained during failover and failback), redundancy, dial plan simplification (no dial peer configuration is required on the gateway), hookflash transfer, and tone on hold. MGCP-controlled gateways do not require an MTP to enable supplementary services such as hold, transfer, call pickup, and call park.

- **H.323**—The Cisco IOS integrated router gateways use H.323 protocol to communicate with Cisco CallManager. Intercluster trunks for connecting remote Cisco CallManagers across the IP WAN are also configured as H.323 gateways.

  Compared to MGCP, H.323 requires more configuration on the gateway, because the gateway must maintain the dial plan and route patterns.

- **Skinny Gateway Protocol**—Older Cisco voice gateways such as the AT-2, AT-4, AT-8, AS-2, AS-4, AS-8, and theCisco Catalyst 6000 24 Port FXS Analog Interface Module.

Trunk Interfaces

Device protocols specify the time-division multiplexing (TDM) signaling interface between the voice gateway and the PSTN or external non-IP telephony devices. Supported TDM interfaces vary by gateway model. The following list gives available interfaces:

- Foreign Exchange Office (FXO)—Use FXO ports for connecting to a central office or PBX. You can configure loop-start, ground-start, and E&M signaling interfaces, depending on the model selected.

Cisco CallManager assumes all loop start trunks lack positive disconnect supervision. We recommend that you configure trunks with positive disconnect supervision as ground start.

- Foreign Exchange Station (FXS)—Use FXS ports to connect to any Plain Old Telephone Service (POTS) device such as analog phones, fax machines, and legacy voice-mail systems.
- T1-PRI—Use this interface to designate North American ISDN Public Rate Interface with 23 bearer channels and one common channel signaling (CCS).
- E1-PRI—Use this interface to designate European ISDN Primary Rate Interface with 30 bearer channels, one CCS channel, and one framing channel.
- T1-CAS—Use this interface to designate T1 channel associated signaling (CAS), where each channel includes a dedicated signaling element. The supported signaling interface type is E&M.

Cisco Voice Gateways

Cisco CallManager supports several types of Cisco IP telephony gateways. These sections provide an overview of these supported gateways.

Standalone Voice Gateways

This section briefly describes the standalone, application-specific gateway models supported for use with Cisco CallManager.

Cisco Voice Gateway 200 Gateway

The Cisco IP Telephony VG200 provides a 10/100BaseT Ethernet port for connection to the data network. The following list gives available telephony 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 T1 PRI or T1-CAS ports for connecting to the PSTN
- 1 or 2 E1 PRI ports for connecting to the PSTN

- MGCP or H.323 interface to Cisco CallManager
  - MGCP mode supports T1/E1 PRI (user side only), T1-CAS, FXS, and FXO.
  - H.323 mode supports E1/T1 PRI (user side only), E1/T1-CAS, FXS, and FXO, and E&M, fax relay, G.711 modem.

The MGCP VG200 integration with legacy voice-mail systems allows the Cisco CallManager to associate a port with a voice mailbox and connection.

Cisco Access Digital Trunk Gateways DT-24+/DT30+

The Cisco Access Digital Gateways DT-24+/DE-30+ provide the following features:

- T1/E1 PRI (network or user side)
- T1-CAS connections (DT-24+) supporting E&M signaling with wink, immediate, or delay dial supervision; and loop start FXO and ground start circuit emulation.
- MGCP interface to Cisco CallManager

Cisco Analog Access Station Gateways

Station gateways let you connect the Cisco CallManager to POTS analog telephones, interactive voice response (IVR) systems, fax machines, and voice-mail systems. Station gateways provide FXS ports. The AS-2, AS-4, and AS-8 models accommodate two, four, and eight Voice over IP (VoIP) gateway channels, respectively.

Cisco AS gateways communicate with Cisco CallManager using SGCP.

Cisco Access Analog Trunk Gateways

Analog trunk gateways let you connect the Cisco CallManager to standard PSTN central office (CO) or PBX trunks. Trunk gateways provide FXO ports. The AT-2, AT-4, and AT-8 models accommodate two, four, and eight VoIP gateway channels. The signaling type is loop start.

Cisco AT gateways communicate with Cisco CallManager using SGCP.

Cisco Catalyst 4000 and 6000 Voice Gateway Modules

Several available telephony modules for the Cisco Catalyst 4000 and 6000 family switches act as telephony gateways enabling you to implement IP telephony in your network using existing Cisco Catalyst 4000 or 6000 family devices.

You can install Catalyst 6000 voice gateway modules that are line cards in any Cisco Catalyst 6000 or 6500 series switch. You can install The Catalyst 4000 access gateway module in any Catalyst 4000 or 4500 series switch.

Cisco Catalyst 6000 8 Port Voice T1/E1 and Services Module

The Cisco Catalyst 6000 8 Port Voice T1/E1 and Services Modules provide the following features:

- 8 ports for providing
  - Digital T1/E1 connectivity to the PSTN (T1/E1 PRI or T1-CAS with the same feature as DT-24+/DE-30+)
  - DSP resources for transcoding and conferencing
- MGCP interface to Cisco CallManager

Depending upon which port type is configured, the ports can serve as T1/E1 interfaces, or the ports will support transcoding or conferencing.

Note Either blade supports DSP features on any port, but T1 blades cannot be configured for E1 ports, and E1 blades cannot be configured for T1 ports.

Users have the flexibility to use each port for T1/E1 connections or as network resources for voice services.

Cisco Catalyst 4000 Access Gateway Module

The Cisco Catalyst 4000 Access Gateway Module provides the following telephony features:

- 6 ports for FXS, FXO or E&M
- 2 T1/E1 ports for T1 PRI, T1-CAS, or E1 PRI
- MGCP interface to Cisco CallManager

Cisco Catalyst 4224 Access Gateway Module

The Cisco Catalyst 4224 Access Gateway Module provides the following features:

- 8 ports for FXS
- Supported protocols and interface types including
  - T1-PRI, E1-PRI, T1-CAS, E1-CAS R2, ISDN BRI, and FXO
- MGCP interface to Cisco CallManager

Cisco Catalyst 6000 24 Port FXS Analog Interface Module

The Cisco Catalyst 6000 24 Port FXS Analog Interface Module provides the following features:

- 24 Port RJ-21 FXS module
- V.34/V.90 modem, voice mail, IVR, POTS
- Cisco fax relay (T.38 Phase 2)
- MGCP interface to Cisco CallManager

The Catalyst 6000 24 Port FXS Analog Interface Module provides 24 FXS ports for connecting to analog phones, conference room speaker phones, and fax machines. You can also connect to legacy voice-mail systems. Using SMDI, you can associate the ports with voice-mail extensions.

The FXS module provides legacy analog devices with connectivity into the IP network, enabling them to utilize the IP network infrastructure for toll-bypass applications and to communicate with devices such as IP phones and H.323 end stations. This module also supports fax relay, which enables compressed fax transmission over the IP WAN, preserving valuable WAN bandwidth for other data applications.

H.323 Gateways

H.323 devices comply with the H.323 communications standards and enable video conferencing over LANs and other packet-switched networks. You can add third-party H.323 devices or other Cisco devices that support H.323 (such as the Cisco 2600 series, 3600 series, or 5300 series gateways). You can also configure H.323 intercluster trunks to connect Cisco CallManagers in different clusters.

Cisco IOS H.323 Gateways

Cisco IOS H.323 gateways such as the Cisco 2600, 3600, 1750, 3810 V3, 7200 7500, AS5300, and VG200 provide full-featured routing capabilities as well as VoIP gateway functions. Refer to the documentation for each of these gateway types for information about support voice gateway features and configuration.

Intercluster Trunks

Use an intercluster trunk, an H.323 device, to connect two Cisco CallManagers in remote clusters. For information about configuring gatekeeper-controlled H.323 intercluster trunks for routing intercluster calls across a remote WAN link, refer to the *Cisco IP Telephony Network Design Guide*.

Voice Gateway Model Summary

Table 32-1 summarizes Cisco voice gateways supported by Cisco CallManager, with information about the gateway control protocols, trunk interfaces, and port types.

Table 32-1 Overview of Supported Voice Gateways, Protocols, Trunk Interfaces, and Ports

Gateway Model	Gateway Control Protocol	Trunk Interface	Port Types
Cisco IOS Integrated Routers			
Cisco 1750	H.323 (H.225)	FXS	POTS
		FXO	E&M
Cisco 3810 V3	H.323 (H.2250	T1-CAS	T1-CAS
		E1-CAS	E1-CAS
Cisco 2600	MGCP or H.323	FXS	POTS
		FXO	Loop start, ground start, E&M
		T1 PRI	T1 PRI
		T1-CAS	E&M
		E1 PRI	E1 PRI

Table 32-1 Overview of Supported Voice Gateways, Protocols, Trunk Interfaces, and Ports (continued)

Gateway Model	Gateway Control Protocol	Trunk Interface	Port Types
Cisco 3600	MGCP or H.323	FXS FXO T1 PRI T1-CAS E1 PRI	POTS Loop start, ground start, or E&M T1 PRI E&M E1 PRI
Cisco 7200	H.323 (H.225)	T1/E1 CAS T1/E1 PRI	T1/E1 CAS T1/E1 PRI
Cisco 7500	H.323 (H.225)	T1/E1 CAS T1/E1 PRI	T1/E1 CAS T1/E1 PRI
Cisco AS5300	H.323 (H.225)	T1/E1 CAS T1/E1 PRI	T1/E1 CAS T1/E1 PRI
Intercluster Trunk	H.323	Intercluster Trunk	Not applicable
Cisco Standalone Voice Gateways			
Cisco Voice Gateway 200 (VG200)	MGCP or H.323	FXO FXS T1-PRI T1-CAS E1 PRI	Loop start, ground start, E&M POTS T1-PRI E&M E1 PRI
Cisco Access Digital Trunk Gateway DE-30+	MGCP	E1-PRI	E1 PRI
Cisco Access Digital Trunk Gateway DT-24+	MGCP	T1-PRI T1-CAS	T1-PRI E&M, loop start, ground start

Table 32-1 Overview of Supported Voice Gateways, Protocols, Trunk Interfaces, and Ports (continued)

Gateway Model	Gateway Control Protocol	Trunk Interface	Port Types
Cisco Access Analog Trunk Gateway (AT-2, AT-4, AT-8)	SGCP	FXO	Loop start
Cisco Access Analog Station Gateway (AS-2, AS-4, AS-8)	SGCP	FXS	POTS
Cisco Catalyst Voice Gateway Modules			
Catalyst 4000 Access Gateway Module (WS-X4604-GWY)	MGCP or H.323	FXS FXO T1 CAS	POTS Loop start, ground start, E&M
Cisco Catalyst 6000 8 Port Voice T1 and Services Module (WS-X6608-T1)	MGCP	T1-PRI T1-CAS	T1-PRI E&M, loop start, ground start
Cisco Catalyst 6000 8 Port Voice E1 and Services Module (WS-X6608-E1)	MGCP	E1-PRI	E1-PRI
Cisco Catalyst 6000 24 Port FXS Analog Interface Module	MGCP	Foreign Exchange Station (FXS)	POTS

Gateways, Dial Plans, and Route Groups

Use dial plans to access or call out to the PSTN; route groups; and group specific gateways. Remote Cisco CallManagers across the IP WAN are configured as intercluster (H.323) gateways.

The different gateways used within the Cisco IP Telephony Solutions have dial plans configured in different places:

- Configure dial plan information for both skinny and MGCP gateways in the Cisco CallManager.
- Typically configure H.323-based Cisco IOS software gateways dial plan configuration in Cisco CallManager to access that gateway and configure dial peers in the gateway to pass that call out the gateway.

The route group points to one or more gateways and can select the gateways for call routing based on preference. The route group can direct all calls to the primary device and then use the secondary devices when the primary is unavailable. This serves effectively as a trunk group. One or more route lists can point to the same route group. All devices in a given route group share the same characteristics such as path and digit manipulation. Route groups can perform digit manipulation that will override what was performed in the route pattern.

Configuration information associated with the gateway defines how the call is actually placed.

You can configure an H.323 gateway to be gatekeeper-controlled. This means that before a call is placed to an H.323 device it must successfully query the gatekeeper. Multiple clusters for inbound and outbound calls can share H.323 gateways, but MGCP and SGCP-based gateways are dedicated to a single Cisco CallManager cluster.

Gateway Failover and Failback

This section describes how Cisco voice gateways handle failover and failback.

MGCP Gateways

MGCP gateways receive a list of Cisco CallManagers according to the Cisco CallManager group, defined for the device pool assigned to the gateway. A Cisco CallManager group can contain one, two, or three Cisco CallManagers, listed in priority order, that the gateway uses there. If Cisco CallManager #1 goes down, then Cisco CallManager #2 is used. If #1 and #2 go down, then #3 is used.

Failback is the process of recovering a higher-priority Cisco CallManager when a gateway fails over to a secondary or tertiary Cisco CallManager. For Cisco MGCP gateways, higher priority Cisco CallManagers are periodically checked, statuses are taken and, when determined ready, marked as available again. The gateway then reverts to the highest available Cisco CallManager when all calls have gone idle, or within 24 hours, whichever occurs first. A failback may be forced by the administrator either by stopping the lower priority Cisco CallManager (calls are preserved), or by restarting the gateway (calls are terminated).

IOS H.323 Gateways

Using several enhancements to the **dial-peer** and **voice class** commands in Cisco IOS Release 12.1(2)T, Cisco IOS gateways can now support redundant Cisco CallManagers. A new command, **h225 tcp timeout** *seconds*, that has been added specifies the time it takes for the Cisco IOS gateway to establish an H.225 control connection for H.323 call setup. If the Cisco IOS gateway cannot establish an H.225 connection to the primary Cisco CallManager, it tries a second Cisco CallManager defined in another **dial-peer** statement. The Cisco IOS gateway shifts to the **dial-peer** statement with next highest **preference** setting.

The following example shows the configuration for H.323 gateway failover:

```
interface Loopback0
    ip address 1.1.1.1 255.255.255.0
    voip-gateway voip bind srcaddr 1.1.1.1
dial-peer voice 101 voip
    destination-pattern 1111
    session target ipv4:10.1.1.101
    preference 0
    voice class h323 1
dial-peer voice 102 voip
    destination-pattern 1111
    session target ipv4:10.1.1.102
    preference 1
    voice class h323 1
voice class h323 1
    h225 timeout tcp establish 3
```

Note To simplify troubleshooting and firewall configurations, we recommend that you use the new `voip-gateway voip bind srcaddr` command for forcing H.323 always to use a specific source IP address in call setup. Without this command, the source address used in the setup might vary depending on protocol (RAS, H.225, H.245 or RTP).

SGCP Gateways

SGCP gateways are identical to MGCP gateways in terms of Cisco CallManager redundancy, failover and failback.

Gateway Configuration Checklist

Table 32-2 provides an overview of the steps required to configure gateways in Cisco CallManager, along with references to related procedures and topics.

Table 32-2 Gateway Configuration Checklist

Configuration Steps		Procedures and Related Topics
Step 1	Install and configure the gateway or voice gateway module in the network.	Refer to the installation and configuration documentation for the model of gateway you are configuring.
Step 2	Gather the information you need to configure the gateway to operate with Cisco CallManager and to configure the trunk interface to the PSTN or external non-IP telephony device.	For Gateway Configuration Settings, refer to the *Cisco CallManager Administration Guide*. For Port Configuration Settings, refer to the *Cisco CallManager Administration Guide*.
Step 3	On the gateway, perform any required configuration steps.	Refer to the voice feature software configuration documentation or Cisco IOS documentation for the model of gateway you are configuring.
Step 4	Add and configure the gateway in Cisco CallManager Administration.	For Adding Gateways to Cisco CallManager, refer to the *Cisco CallManager Administration Guide*.
Step 5	Add and configure ports on the gateway.	For Port Configuration Settings, refer to the *Cisco CallManager Administration Guide*.
Step 6	For FXS ports, add directory numbers, if appropriate.	For Adding a Directory Number and Directory Number Configuration Settings, refer to *Cisco CallManager Administration Guide*.

Table 32-2 Gateway Configuration Checklist (continued)

Configuration Steps		Procedures and Related Topics
Step 7	Configure the dial plan for the gateway for routing calls out to the PSTN or other destinations. This can include setting up a route group, route list, and route pattern for the Gateway in Cisco CallManager or, for some gateways, configuring the dial plan on the gateway itself.	For Dial Plan Architecture and Configuration, refer to the *Cisco IP Telephony Network Design Guide.* See the "Dial Plan Architecture" section on page -5.
Step 8	Reset the gateway to apply the configuration settings.	For Resetting and Restarting Gateways, refer to the *Cisco CallManager Administration Guide*.

Tips

To get to the default web pages for gateway devices, you can use the IP address of that gateway. Make your hyperlink url = http://www.x.x.x.x/, where x.x.x.x. is the dot-form IP address of the device. The web page for each gateway contains device information and the real time status of the gateway.

Where to Find More Information

Related Topics

- Adding Gateways to Cisco CallManager, *Cisco CallManager Administration Guide*
- Gateway Configuration Settings, *Cisco CallManager Administration Guide*
- Port Configuration Settings, *Cisco CallManager Administration Guide*
- Directory Number Configuration Settings, *Cisco CallManager Administration Guide*

Additional Cisco Documentation

- *Cisco IP Telephony Network Design Guide*

  http://www.cisco.com/univercd/cc/td/doc/product/voice/ip_tele/network/
- Cisco Voice Gateway 200 (VG200) documentation on CCO

  http://www.cisco.com/univercd/cc/td/doc/product/voice/c_access/vg_200/