



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, page 34-1](#)
- [Cisco Voice Gateways, page 34-4](#)
- [Gateways, Dial Plans, and Route Groups, page 34-19](#)
- [Gateway Failover and Failback, page 34-20](#)
- [Gateway Configuration Checklist, page 34-22](#)
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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 that you configure in Cisco CallManager Administration versus what is configured on the gateway vary, depending on whether gateway control protocol is MGCP, H.323, or Skinny:

- **Media Gateway Control Protocol (MGCP)**—When MGCP is used, 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 a media termination point (MTP) to enable supplementary services such as hold, transfer, call pickup, and call park. See [Table 34-1 on page 34-14](#) for gateway models that support MGCP.
- **H.323 Protocol**—The Cisco IOS integrated router gateways use the H.323 protocol to communicate with Cisco CallManager. See [Table 34-1 on page 34-14](#) for gateway models that support H.323 protocol.
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 use Skinny Gateway Protocol.
- **Skinny Client Control Protocol**—Cisco VG248 Gateway registers and communicates with the Cisco CallManager by using the Skinny Client Control Protocol, which also allows support for enhanced features, such as Message Waiting Indication. The Cisco VG248 Gateway supports clustering and failover.

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 that is selected.

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

- 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 Primary Rate Interface (PRI) with 23 bearer channels and one common channel signaling (CCS) channel.
- E1 PRI—Use this interface to designate European ISDN Primary Rate Interface (PRI) 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 types include E&M, ground start, and loop start.
- QSIG—Use this protocol to connect with digital PBXs that use common channel signaling protocol based on ISDN Q.931 standards. Cisco CallManager supports QSIG basic call and ID services for certain MGCP-controlled gateways and for PBXs.

Cisco supports basic call services for the following features:

- Calling Line Identification Presentation (CLIP)
- Connected Name Identification Presentation (CNIP)
- Connected Name Identification Presentation (CONP)
- Calling/Connected Line Identification Restriction (CLIR)

The type of PBX determines which QSIG ID services are supported. Refer to the *Cisco IP Telephony Network Design Guide* for details about compatible PBX software loads, configuration and operational caveats for QSIG protocol.

Cisco Voice Gateways

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

- [Standalone Voice Gateways, page 34-4](#)
- [Cisco Catalyst 4000 and 6000 Voice Gateway Modules, page 34-8](#)
- [Cisco Integrated Communications System 7750 Gateways, page 34-11](#)
- [H.323 Gateways, page 34-13](#)

Standalone Voice Gateways

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

Cisco Voice Gateway 200

The Cisco IP Telephony Voice Gateway (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+/DE-30+

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

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

Cisco Access Analog 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 Access AS gateways communicate with Cisco CallManager by using Skinny Gateway Protocol.

Cisco 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 specifies loop start.

Cisco Access AT gateways communicate with Cisco CallManager by using Skinny Gateway Protocol.

Cisco VG248 Analog Phone Gateway

The Cisco VG248 Analog Phone Gateway has a standalone, 19-inch rack-mounted chassis with 48-FXS ports. This product allows on-premise analog telephones, fax machines, modems, voice-mail systems, and speakerphones to register with a single Cisco CallManager cluster.

Cisco VG248 Analog Phone Connectivity

The Cisco VG248 Analog Phone Gateway communicates with the Cisco CallManager by using the Skinny Client Control Protocol to allow support for the following supplementary services features for analog phones:

- Call transfer
- Conference
- Call waiting (with calling party ID display)
- Hold (including switch between parties on hold)
- Music on hold
- Call forward all
- Send all calls to voice mail
- Group call pickup
- Voice-mail message waiting indication
- Speed dial (maximum of 9 speed dials)
- Last number redial
- Cisco fax relay
- Dynamic port and device status that is available from Cisco CallManager

Cisco VGC Phone Device Types

All Cisco VG248 ports and units appear as distinct devices in the Cisco CallManager with the device type “Cisco VGC Phone.” The Cisco CallManager recognizes and configures each port as a phone.

Fax and Modem Connectivity

The Cisco VG248 supports legacy fax machines and modems. When fax machines are used, the Cisco VG248 uses the Cisco fax-relay technology to transfer faxes across the network with high reliability by using less bandwidth than a voice call uses.

You can connect any modem to the Cisco VG248.

Voice-Mail Connectivity

The Cisco VG248 generates call information by using the Simplified Message Desk Interface (SMDI) format for all calls that are ringing on any of the 48 analog lines that are connected to it. It will also pass on SMDI call information from other Cisco VG248s, or from a legacy PBX, to the voice-mail system. Any commands for message-waiting indicators get sent to the Cisco CallManager and to any other attached SMDI hosts.

This mechanism allows for many new configurations when SMDI-based voice-mail systems are used, including

- You can share a single voice-mail system between Cisco CallManager and a legacy PBX.
- Voice mail and Cisco VG248 can function remotely in a centralized call-processing model.
- Multiple clusters can use a single voice-mail system, by using one Cisco VG248 per cluster.
- Configure multiple voice-mail systems in a single cluster because the Cisco VG248 generates SMDI call information rather than the Cisco CallManager.

Cisco VG248 Time Device

The Cisco VG248 contains a real-time clock that is persistent across power cycles and restarts. The real-time clock gets set for the first time when the device registers with the Cisco CallManager. The clock gets set by using the DefineDateTime Skinny message that the Cisco CallManager sends. After a power cycle or restart, the clock resets when the Cisco VG248 receives the DefineDateTime message from Cisco CallManager and then resets not more than once per hour thereafter.

Cisco VG248 Configuration File Updates

The Cisco VG248 queries the TFTP server to access the configuration files for the device. The configuration files update whenever you modify the configuration of the Cisco VG248 via the Cisco CallManager.

Refer to the [Gateway Configuration](#) section and the [Cisco IP Phone Configuration](#) section of the *Cisco CallManager Administration Guide* and to the *Cisco VG248 Analog Phone Gateway Software Configuration Guide* for more information.

Cisco IAD2400 Series Integrated Access Device

The Cisco IAD2420 integrated access device provides voice, data, and video services over internet protocol (IP) and asynchronous transfer mode (ATM) networks. By using the Cisco IAD 2420, service providers can deliver toll-quality voice and data services over circuit- or packet-switched networks. The Cisco IAD2420 provides an MGCP interface with Cisco CallManager and supports the following capabilities:

- Analog: FXS ports for POTS telephony devices, FXO ports for PSTN connections
- Digital: T1 PRI and T1 CAS services

Cisco Catalyst 4000 and 6000 Voice Gateway Modules

Several telephony modules for the Cisco Catalyst 4000 and 6000 family switches act as telephony gateways. You can use existing Cisco Catalyst 4000 or 6000 family devices to implement IP telephony in your network by using the following voice gateway modules:

- Install Catalyst 6000 voice gateway modules that are line cards in any Cisco Catalyst 6000 or 6500 series switch.
- 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 features as DT-24+/DE-30+)
 - Digital signal processor (DSP) resources for transcoding and conferencing
- MGCP interface to Cisco CallManager
- Connection to a voice-messaging system (using T1 CAS)

Users have the flexibility to use ports on a T1 module for T1 connections or as network resources for voice services. Similarly, the E1 module provides ports for E1 connections or as network resources. The ports can serve as T1/E1 interfaces, or the ports will support transcoding or conferencing.



Note

Either module supports DSP features on any port, but T1 modules cannot be configured for E1 ports, and E1 modules cannot be configured for T1 ports.

Similar to the Cisco MGCP-controlled gateways with FXS ports, the Cisco 6608 T1 CAS gateway supports hookflash transfer. Hookflash transfer is a signaling procedure that allows a device, such as a voice-messaging system, to transfer to another destination. While the device is connected to Cisco CallManager through a T1 CAS gateway, the device performs a hookflash procedure to transfer the call to another destination. Cisco CallManager responds to the hookflash by using a blind transfer to move the call. When the call transfer completes, the voice channel that connected the original call to the device gets released.



Note

Only E&M T1 ports support hookflash transfer.

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
- 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 speakerphones, and fax machines. You can also connect to legacy voice-mail systems by using SMDI and by associating the ports with voice-mail extensions.

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

Cisco Communication Media Module

The Cisco Communication Media Module (CMM), which is a Catalyst 6500 line card, provides T1 and E1 gateways that allow organizations to connect their existing TDM network to their IP communications network. The Cisco CMM provides connectivity to the PSTN also. You can configure the Cisco CMM, which provides an MGCP interface to Cisco CallManager, with the following interface and service modules:

- 6-port T1 interface module for connecting to the PSTN or a PBX
- 6-port E1 interface module for connecting to the PSTN or a PBX
- 24-port FXS interface module for connecting to POTS telephony devices

Cisco Catalyst 4000 Access Gateway Module

The Cisco Catalyst 4000 Access Gateway Module provides an MGCP or H.323 gateway interface to Cisco CallManager. You can configure this module with the following interface and service modules:

- 6 ports for FXS and FXO
- 2 T1/E1 ports for T1 PRI, T1 CAS, or E1 PRI

Cisco Catalyst 4224 Access Gateway Switch

The Cisco Catalyst 4224 Access Gateway Switch provides a single-box solution for small branch offices. The Catalyst 4224 provides switching, IP routing, and PSTN voice-gateway services by using onboard digital signal processors (DSPs). The Catalyst 4224 has four slots that you can configure with multiflex voice and WAN interface cards to provide up to 24 ports. These ports can support the following voice capabilities:

- FXS ports for POTS telephony devices
- FXO ports for PSTN connections
- T1 or E1 ports for T1 PRI, E1 PRI, and T1 CAS services

The Cisco Catalyst 4224 Access Gateway Switch provides an MGCP or H.323 interface to Cisco CallManager.

Cisco Integrated Communications System 7750 Gateways

The Cisco Integrated Communications System (ICS) 7750 provides an integrated communications platform for converged voice/data applications and services, including IP telephony, multiservice routing, and applications such as unified messaging, integrated web call centers, data/voice collaboration, and networked video.

The Cisco ICS 7750 uses multiservice route processor (MRP)/voice gateway cards based on Cisco IOS software to provide the functionality of multiservice routers and H.323-compliant voice gateways. The multiservice route processor (MRP) card, a voice- and data-capable router, supports both digital and analog trunks and WAN routing interfaces. Using the MRP, you can link remote Ethernet LANs to the public switched telephone network (PSTN) and existing private

branch exchanges (PBXs) as well as most common analog devices such as fax machines and teleconferencing stations. The MRP card accepts voice interface cards (VICs), WAN interface cards (WICs), and voice WAN interface cards (VWICs) for complete integration of voice and data networking. Refer to the [Cisco ICS 7750 System Description](#) for information about MRPs and supported VICs, WICs, and VWICs.

In release 2.0 and later, Analog Station Interface (ASI) cards add support for high-density analog foreign exchange station (FXS) ports (8- and 16-port versions are available).

Cisco ICS 7750 MRP Cards

The following Cisco ICS 7750 MRP cards provide gateway support in Cisco CallManager.

The MRP300 has flash memory and two slots for VIC, WIC, and VWIC modules that can provide the following features:

- T1/E1 PRI and T1 CAS (E&M only) connections
- FXS for analog POTS connections
- FXO for loop-start or ground-start trunks
- MGCP or H.323 interface to Cisco CallManager
- Digital signal processor (DSP) resources for transcoding and conferencing

Other models of the MRP300 include

- MRP3-8FXS—Contains an 8-port FXS module and an open slot for any VIC, WIC, or VWIC module that support digital and analog voice trunks and WAN routing.
- MRP3-16FXS—Contains a 16-port FXS module for analog phone connections.
- MRP3-8FXOM1—Combines the MRP300 card with a high-density analog module. The MRP3-8FXOM1 card has eight onboard FXO M1 ports and an open VIC/WIC/VWIC slot. The FXO M1 ports enable connections to a PSTN or to a PBX.

The Cisco MRP200 does not have onboard Flash memory and has two slots for VIC, WIC, and VWIC modules that provide:

- T1/E1 PRI and T1 CAS (E&M only) connections

- FXS for analog POTs connections
- FXO for loop start or ground start trunks
- MGCP or H.323 interface to Cisco CallManager
- Digital signal processor (DSP) resources for transcoding and conferencing

Cisco ICS 7750 ASI Cards

The following Cisco ICS 7750 ASI cards provide gateway support in Cisco CallManager:

- ASI81—Contains an 8-port FXS module and an open VIC, WIC, or VWIC slot. Although the ASI81 is similar to the MRP3-8FXS, the ASI81 does not have onboard Flash memory.
- ASI160—Contains a 16-port FXS module. Although the ASI160 is similar to the MRP3-16FXS, the ASI160 does not have onboard Flash memory.

The ASI cards provide the following features:

- FXS for analog POTS connections
- FXO for loop-start or ground-start trunks
- MGCP interface to Cisco CallManager

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).

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. Refer to the documentation for each of these gateway types for information about support voice gateway features and configuration.

Voice Gateway Model Summary

Table 34-1 summarizes Cisco voice gateways that Cisco CallManager supports with information about the gateway control protocols, trunk interfaces, and port types.

Table 34-1 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 FXO	POTS Loop start or ground start
Cisco 3810 V3	H.323 (H.225)	T1 CAS E1 CAS	T1 CAS E1 CAS
Cisco 2600 series	MGCP or H.323	FXS FXO T1/E1 PRI T1 CAS QSIG (Not all Cisco 2600 series gateways support QSIG. Refer to your gateway documentation for additional information.)	POTS Loop start or ground start T1/E1 PRI E&M T1/E1 PRI

Table 34-1 Overview of Supported Voice Gateways, Protocols, Trunk Interfaces, and Ports (continued)

Gateway Model	Gateway Control Protocol	Trunk Interface	Port Types
Cisco 3600 series	MGCP or H.323	FXS FXO T1/E1 PRI T1 CAS QSIG (Not all Cisco 3600 series gateways support QSIG. Refer to your gateway documentation for additional information.)	POTS Loop start or ground start T1/E1 PRI E&M T1/E1 PRI
Cisco 3725	MGCP or H.323	FXS FXO T1/E1 PRI T1 CAS QSIG	POTS Loop start or ground start T1/E1 PRI E&M T1/E1PRI
Cisco 3745	MGCP or H.323	FXS FXO T1/E1 PRI T1 CAS QSIG	POTS Loop start or ground start T1/E1 PRI E&M T1/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

Table 34-1 Overview of Supported Voice Gateways, Protocols, Trunk Interfaces, and Ports (continued)

Gateway Model	Gateway Control Protocol	Trunk Interface	Port Types
Cisco AS5300	H.323 (H.225)	T1/E1 CAS T1/E1 PRI	T1/E1 CAS T1/E1 PRI
Cisco Standalone Voice Gateways			
Cisco Voice Gateway 200 (VG200)	MGCP or H.323	FXO FXS T1/E1 PRI T1 CAS QSIG	Loop start or ground start POTS T1/E1 PRI E&M T1/E1 PRI
Cisco Access Digital Trunk Gateway DE-30+	MGCP	E1 PRI QSIG	E1 PRI E1 PRI
Cisco Access Digital Trunk Gateway DT-24+	MGCP	T1 PRI T1 CAS QSIG	T1 PRI E&M, loop start, ground start T1 PRI
Cisco Access Analog Trunk Gateway (AT-2, AT-4, AT-8)	Skinny Gateway Protocol	FXO	Loop start
Cisco Access Analog Station Gateway (AS-2, AS-4, AS-8)	Skinny Gateway Protocol	FXS	POTS
Cisco VG248 Analog Phone Gateway	Skinny Client Control Protocol	FXS	POTS
Cisco IAD2420	MGCP	FXS FXO T1 PRI T1 CAS QSIG	POTS Loop start or ground start T1 PRI E&M T1 PRI

Table 34-1 Overview of Supported Voice Gateways, Protocols, Trunk Interfaces, and Ports (continued)

Gateway Model	Gateway Control Protocol	Trunk Interface	Port Types
Cisco Catalyst Voice Gateway Modules			
Cisco Catalyst 4000 Access Gateway Module (WS-X4604-GWY)	MGCP or H.323	FXS FXO T1 CAS T1/E1 PRI QSIG	POTS Loop start or ground start E&M T1/E1 PRI T1/E1 PRI
Cisco Catalyst 4224 Voice Gateway Switch	MGCP or H.323	FXS FXO T1/E1 PRI T1 CAS QSIG	POTS Loop start or ground start T1/E1 PRI E&M T1/E1 PRI
Cisco Catalyst 6000 8-Port Voice T1/E1 and Services Module (WS-X6608-T1) (WS-X6608-E1)	MGCP	T1/E1 PRI T1 CAS QSIG	T1/E1 PRI E&M, loop start, ground start T1/E1 PRI
Cisco Catalyst 6000 24-Port FXS Analog Interface Module (WS-X6624-FXS)	MGCP	FXS	POTS
Cisco Communication Media Module (WS-X6600-24FXS) (WS-X6600-6T1) (WS-X6600-6E1)	MGCP	FXS T1 PRI T1 CAS E1 PRI	POTS T1 PRI E&M E1 PRI

Table 34-1 Overview of Supported Voice Gateways, Protocols, Trunk Interfaces, and Ports (continued)

Gateway Model	Gateway Control Protocol	Trunk Interface	Port Types
Cisco ICS Gateways			
Cisco ICS77XX-MRP3 _{xx}	MGCP or H.323	FXS	POTS
Cisco ICS77XX-MRP3-8FXO-M1		FXO	Loop start or ground start
Cisco ICS77XX-MRP3-8FXS		T1/E1 PRI	T1/E1 PRI
		T1 CAS	E&M
	QSIG	T1/E1 PRI	
Cisco ICS77XX-MRP3-16FXS	MGCP or H.323	FXS	POTS
Cisco ICS77XX-MRP2 _{xx}	MGCP or H.323	FXS	POTS
		FXO	Loop start or ground start
		T1/E1 PRI	T1/E1 PRI
		T1 CAS	E&M
		QSIG	T1/E1 PRI
Cisco ICS77XX-ASI81	MGCP or H.323	FXS	POTS
		FXO	Loop start or ground start
		T1/E1 PRI	T1/E1 PRI
		T1 CAS	E&M
		QSIG	T1/E1 PRI
Cisco ICS77XX-ASI160	MGCP or H.323	FXS	POTS

Gateways, Dial Plans, and Route Groups

Gateways use dial plans to access or call out to the PSTN, route groups, and group-specific gateways. The different gateways that are used within the Cisco IP Telephony Solutions have dial plans that are configured in different places:

- Configure dial plan information for both Skinny and MGCP gateways in the Cisco CallManager.
- Configure dial plans in Cisco CallManager to access the H.323-based Cisco IOS software gateways. Configure dial peers in the H.323-based gateways to pass the call out of the gateway.

The route group points to one or more gateways and can choose the gateways for call routing based on preference. The route group can serve as a trunk group by directing all calls to the primary device and then using the secondary devices when the primary is unavailable. 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 that is associated with the gateway defines how the call is actually placed.

You can configure H.323 trunks, *not* H.323 gateways, to be gatekeeper-controlled trunks. This means that before a call is placed to an H.323 device, it must successfully query the gatekeeper. See the [“Gatekeeper and Trunk Configuration in Cisco CallManager” section on page 7-12](#) for more information.

Multiple clusters for inbound and outbound calls can share H.323 trunks, but MGCP and Skinny-based gateways remain dedicated to a single Cisco CallManager cluster.

Dependency Records for Route Groups and Directory Numbers

To find route groups or directory numbers that a specific gateway or gateway port is using, click the Dependency Records link that is provided on the Cisco CallManager Administration Gateway Configuration window. The Dependency Records Summary window displays information about route groups

and directory numbers that are using the gateway or port. To find out more information about the route group or directory number, click the route group or directory number, and the Dependency Records Details window displays.

For more information about Dependency Records, refer to [Accessing Dependency Records](#), [Deleting Gateways](#), and [Deleting a Directory Number](#) in the *Cisco CallManager Administration Guide*.

Gateway Failover and Failback

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

MGCP Gateways

To handle Cisco CallManager failover situations, MGCP gateways receive a list of Cisco CallManagers that is arranged according to the Cisco CallManager group and defined for the device pool that is assigned to the gateway. A Cisco CallManager group can contain one, two, or three Cisco CallManagers that are listed in priority order for the gateway to use. If the primary Cisco CallManager in the list fails, the secondary Cisco CallManager gets used. If the primary and secondary Cisco CallManagers fail, the tertiary Cisco CallManager gets used.

Failback describes the process of recovering a higher priority Cisco CallManager when a gateway fails over to a secondary or tertiary Cisco CallManager. Cisco MGCP gateways periodically take status of higher priority Cisco CallManagers. When a higher priority Cisco CallManager is ready, it gets marked as available again. The gateway reverts to the highest available Cisco CallManager when all calls go idle or within 24 hours, whichever occurs first. The administrator can force a failback either by stopping the lower priority Cisco CallManager whereby calls get preserved, by restarting the gateway which preserves calls, or by resetting Cisco CallManager which terminates calls.



Note

Skinny gateways handle Cisco CallManager redundancy, failover, and failback in the same way as MGCP gateways.

IOS H.323 Gateways

Cisco IOS gateways can now handle Cisco CallManager failover situations. By using several enhancements to the **dial-peer** and **voice class** commands in Cisco IOS Release 12.1(2)T, Cisco IOS gateways can support redundant Cisco CallManagers. A new command, **h225 tcp timeout seconds**, specifies the time that 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 that is defined in another **dial-peer** statement. The Cisco IOS gateway shifts to the **dial-peer** statement with the 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, Cisco recommends 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 that is used in the setup might vary and depends on protocol (RAS, H.225, H.245, or RTP).

Cisco VG248 Analog Phone Gateway

The Cisco VG248 Analog Phone Gateway supports the Skinny Client Control Protocol for clustering and failover.

Gateway Configuration Checklist

[Table 34-2](#) provides an overview of the steps that are required to configure gateways in Cisco CallManager, along with references to related procedures and topics.

Table 34-2 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 that you are configuring.
Step 2	Gather the information that 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.	Gateway Configuration Settings , <i>Cisco CallManager Administration Guide</i> Port Configuration Settings , <i>Cisco CallManager Administration Guide</i>
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 that you are configuring.
Step 4	Add and configure the gateway in Cisco CallManager Administration.	Adding Gateways to Cisco CallManager , <i>Cisco CallManager Administration Guide</i>
Step 5	Add and configure ports on the gateway or Add and configure the Cisco VG248 Analog Phone Gateway.	Port Configuration Settings , <i>Cisco CallManager Administration Guide</i> Adding a Cisco VG248 Analog Phone Gateway , <i>Cisco CallManager Administration Guide</i> Cisco IP Phone Configuration , <i>Cisco CallManager Administration Guide</i>

Table 34-2 Gateway Configuration Checklist (continued)

Configuration Steps	Procedures and Related Topics
Step 6 For FXS ports, add directory numbers, if appropriate.	Adding a Directory Number , <i>Cisco CallManager Administration Guide</i> Directory Number Configuration Settings , <i>Cisco CallManager Administration Guide</i>
Step 7 Configure the dial plan for the gateway for routing calls out to the PSTN or other destinations. This configuration 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.	<i>Cisco IP Telephony Network Design Guide</i> <i>Cisco CallManager Administration Guide</i>
Step 8 Reset the gateway to apply the configuration settings.	Resetting and Restarting Gateways , <i>Cisco CallManager Administration Guide</i>

**Tip**

To get to the default web pages for gateway devices, you can use the IP address of that gateway. Make your hyperlink url = `http://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*
- *Cisco ICS 7750 System Description*
- *Configuring Cisco IP Telephony Voice Gateways*
- *Implementing Fax Over IP on Cisco Voice Gateways*
- *Cisco VG248 Analog Phone Gateway Software Configuration Guide*
- *Cisco VG248 Analog Phone Gateway Hardware Installation Guide*