CHAPTER 10

Understanding Cisco Unified CallManager Voice Gateways

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

This section covers the following topics:

Cisco Voice Gateways

Cisco Unified CallManager supports several types of Cisco Unified Communications gateways. Gateways use call control protocols to communicate with the PSTN and other non-IP telecommunications devices, such as private branch exchanges (PBXs).

Trunk interfaces specify how the gateway communicates with the PSTN or other external devices by using time-division multiplexing (TDM) signaling. Cisco Unified CallManager and Cisco gateways use a variety of TDM interfaces, but supported TDM interfaces vary by gateway model. Refer to the *Cisco Unified Communications Solution Reference Network Design (SRND)* document for more information about selecting and configuring gateways. The following list gives available interfaces that Cisco Unified CallManager supports:

- Foreign Exchange Office (FXO)
- Foreign Exchange Station (FXS)
- T1 Channel Associated Signaling (CAS)
- Basic Rate Interface (BRI)
- T1 PRI—North American ISDN Primary Rate Interface (PRI)
- E1 PRI—European ISDN Primary Rate Interface (PRI)
- QSIG—Q signaling protocol that is based on ISDN standards
- Session Initiation Protocol (SIP)

Cisco Unified CallManager can use H.323 gateways that support E1 CAS, but you must configure the E1 CAS interface on the gateway.

For information about IP telephony protocols, see Understanding IP Telephony Protocols in the *Cisco Unified CallManager System Guide, Release 5.0(4)*.

These sections provide an overview of the following gateways that Cisco Unified CallManager supports:

- Standalone Voice Gateways, page 10-2
- Cisco Catalyst 4000 and 6000 Voice Gateway Modules, page 10-5
- H.323 Gateways, page 10-7

Standalone Voice Gateways

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

Cisco Voice Gateway 200

The Cisco Unified Communications 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 Digital Access T1 ports for connecting to the PSTN
- 1 or 2 Digital Access PRI ports for connecting to the PSTN
- MGCP or H.323 interface to Cisco Unified CallManager
  - MGCP mode supports T1/E1 PRI, T1 CAS, FXS, FXO. (Only the user side supports BRI.)
  - H.323 mode supports E1/T1 PRI, E1/T1 CAS, FXS, and FXO. H.323 mode supports E&M, fax relay, and G.711 modem.

The MGCP VG200 integration with legacy voice-messaging systems allows the Cisco Unified 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:

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

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-messaging systems, and speakerphones to register with a single Cisco Unified CallManager cluster.

Cisco VG248 Analog Phone Connectivity

The Cisco VG248 Analog Phone Gateway communicates with Cisco Unified 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-messaging system
- Group call pickup
- Voice-messaging system 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 Unified CallManager

Cisco VGC Phone Device Types

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

Fax and Modem Connectivity

The Cisco VG248 supports legacy fax machines and modems. When using fax machines, the Cisco VG248 uses either the Cisco fax relay or pass-through/up speed technology to transfer faxes across the network with high reliability.

You can connect any modem to the Cisco VG248 by using pass-through mode.

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 connect to it. It will also pass on SMDI call information from other Cisco VG248s, or from a legacy PBX, to the voice-messaging system. Any commands for message-waiting indicators get sent to Cisco Unified CallManager and to any other attached SMDI hosts.

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

- You can share a single voice-messaging system between Cisco Unified CallManager and a legacy PBX.
- Voice-messaging system and Cisco VG248 can function remotely in a centralized call-processing model.
- Multiple clusters can use a single voice-messaging system, by using one Cisco VG248 per cluster.
- Configure multiple voice-messaging systems in a single cluster because the Cisco VG248 generates SMDI call information rather than the Cisco Unified 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 Cisco Unified CallManager. The clock gets set by using the DefineDateTime Skinny message that Cisco Unified CallManager sends. After a power cycle or restart, the clock resets when the Cisco VG248 receives the DefineDateTime message from Cisco Unified CallManager and then resets no 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 Cisco Unified CallManager.

Refer to the "Gateway Configuration" section and the "Cisco Unified IP Phone Configuration" section of the *Cisco Unified CallManager Administration Guide, Release 5.0(4)* and to the *Cisco VG248 Analog Phone Gateway Software Configuration Guide* for more information.

Cisco VG224 Analog Phone Gateway

The Cisco VG224 Analog Phone Gateway has a standalone, 17-inch rack-mounted chassis with 24-FXS ports. This product allows on-premise analog telephones, fax machines, modems, and speaker phones to register with Cisco Unified CallManager.

This gateway supports the SCCP and SIP protocols.

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 Unified CallManager and supports the following capabilities:

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

MGCP BRI Call Connections

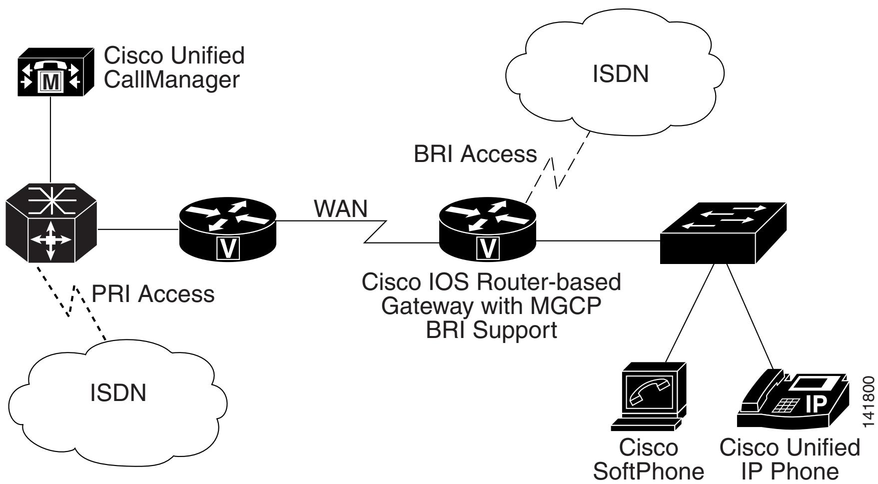
Previously, gateways used H.323 signaling to Cisco Unified CallManager to provide interfaces to the public switched telephone network (PSTN) for BRI ISDN connections. The following list gives drawbacks to using the H.323 protocol:

- Deploying and managing a large number of gateways in a private network represents a very time-consuming task because you must provision every H.323 gateway and its dial plan at the gateway.
- Voice clipping occurs when calls to IP phones use the H.323 gateway because the media cut-through times are very high.
- Calls disconnect if the controlling Cisco Unified CallManager fails during a call.

Now, Cisco Unified CallManager can use a Media Gateway Control Protocol (MGCP) gateway to handle BRI ISDN connections to the PSTN and to provide a centrally administered gateway interface. Cisco Unified CallManager uses logical connections to exchange MGCP and ISDN Q.931 messages with the gateway. This connection uses a User Datagram Protocol (UDP) logical connection for exchanging MGCP messages and a Transmission Control Protocol (TCP) connection for the backhaul ISDN Q.931 messages.

Figure 10-1 shows a typical scenario that centralizes call processing for remote-site BRI trunk gateways that connect to the PSTN. When a call arrives from or goes to the PSTN over the BRI trunk, the Cisco Unified CallManager and the gateway (based on an IOS router) exchange ISDN Q.931 messages across the WAN.

Figure 10-1 Topology Shows a Scenario by Using MGCP BRI Interfaces



For more information about MGCP BRI with Cisco Unified CallManager, refer to the *MGCP-Controlled Backhaul of BRI Signaling in Conjunction with* Cisco Unified CallManager document on the Cisco.com website.

Note The BRI gateway supports MGCP BRI backhaul for BRI trunk only. It does not support BRI phone or station. The IOS gateway supports BRI phones that use Skinny Client Control Protocol.

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 Unified 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 defines 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 Unified CallManager through a T1 CAS gateway, the device performs a hookflash procedure to transfer the call to another destination. Cisco Unified 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-messaging system, IVR, POTS
- Cisco fax relay
- MGCP interface to Cisco Unified 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-messaging systems by using SMDI and by associating the ports with voice-messaging 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 SCCP 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 Unified 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 Unified CallManager. You can configure this module with the following interface and service modules:

- 6 ports for FXS and FXO
- 2 T1/E1 ports for Digital Access PRI and Digital Access T1

Cisco Catalyst 4224 Voice Gateway Switch

The Cisco Catalyst 4224 Voice 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 Digital Access PRI, and Digital Access T1 services

The Cisco Catalyst 4224 Access Gateway Switch provides an MGCP or H.323 interface to Cisco Unified 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, 1751, 1760, 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 supported voice gateway features and configuration.

T.38 Fax Relay

Transporting real-time Group 3 fax documents over internet protocol (IP) uses the International Telecommunications Union Telecommunication Standardization Sector (ITU-T) recommendation T.38 Fax Relay. The T.38 standard defines the IP network protocol that Internet-aware T.38 fax devices and T.38 IP fax gateways use. The T.38 Fax Relay for VoIP H.323 feature provides standards-based, fax relay protocol support for Cisco and other vendor gateways.

The T.38 Fax Relay feature provides a standards-based, fax relay protocol that is available on several Cisco gateways. Because the T.38 Fax Relay protocol is standards based, Cisco gateways and gatekeepers can interoperate with third-party T.38 enabled gateways and gatekeepers in a mixed vendor network that requires real-time fax relay capabilities.

Cisco Unified CallManager handles the T.38 fax call by using a voice connection. When the originating gateway sends a fax, the gateway establishes an initial voice call. The terminating gateway detects the fax tone that the answering fax machine generates. The VoIP H.323 call stack then starts a T.38 mode request by using H.245 procedures. If the opposite end of the call acknowledges the T.38 mode request, the initial audio channel closes, and T.38 Fax Relay channel opens. When the fax transmission finishes, the call disconnects.

Transcoders should be provisioned for use with T.38 Fax Relay either when codec mismatches exist or when fast-connect procedures are employed.

Note Cisco IP Voice Media Streaming Application Service does not support T.38 data transmission. To prevent Cisco IP Voice Media Streaming Application Services from interfering with T.38 fax relay calls, the Cisco IP Voice Media Streaming Application service must be assigned to a Media Resource Group and that Media Resource Group must not be used in any gateways or trunks that are involved in T.38 calls.

Outbound FastStart Call Connections

Calls that are placed from IP phones over large WAN topologies can experience voice clipping when the called party goes off hook to answer the call. When H.323 trunks or gateways are separated from the Cisco Unified CallManager server, significant delays can occur because of the many H.245 messages that are exchanged when a call is set up.

With the FastStart feature, information that is required to complete a media connection between two parties gets exchanged during the H.225 portion of call setup, and this exchange eliminates the need for H.245 messages. The connection experiences one roundtrip WAN delay during call setup, and the calling party does not receive voice clipping when the called party answers the call.

Cisco Unified CallManager uses media termination points (MTP) for making an H.323 outbound FastStart call. Cisco Unified CallManager starts an outbound FastStart call by allocating an MTP and opening the receive channel. Next, the H.323 Fast Connect procedure sends the SETUP message with a FastStart element to the called endpoint. The FastStart element includes information about the receiving channel for the MTP.

The called endpoint accepts the H.323 Fast Connect procedure by sending a CALL PROCEEDING, PROGRESS, ALERT, or CONNECT message that contains a FastStart element. When Cisco Unified CallManager receives the FastStart element, it connects the media immediately and avoids the delays with the usual exchange of H.245 messages.

The called endpoint can refuse the H.323 Fast Connect procedure by not returning the FastStart element in any of the messages up to and including the CONNECT message. In this case, the Cisco Unified CallManager handles the call as a normal call and uses the MTP for subsequent media cut-through.

The Outbound FastStart feature requires an MTP. If an MTP is not available when the call is set up, the call continues without FastStart and with no supplementary services. If you want all calls to use FastStart only, you can set the service parameter called "Fail call if MTP allocation fails," which is located in the Cluster Wide Parameters (Device-H323) portion of the service parameters for the Cisco Unified CallManager service. When you set this parameter to True, the system rejects calls when no MTP is available.

Related Topic

H.323 Gateway Configuration Settings, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*

Voice Gateway Model Summary

Table 10-1 summarizes Cisco voice gateways that Cisco Unified CallManager supports with information about the supported signaling protocols, trunk interfaces, and port types.

Table 10-1 Overview of Supported Voice Gateways, Protocols, Trunk Interfaces, and Port Types

Gateway Model	Supported Signaling Protocols	Trunk Interfaces	Port Types	Notes
Cisco IOS Integrated Routers				
Cisco 1751 and Cisco 1760	H.323 and SIP	FXS	Loopstart or groundstart	Basic calls only
		FXO	Loopstart or groundstart	
		E&M		
		Analog DID		
		CAMA		
		BRI		
		BRI QSIG		Basic calls only
		T1 CAS (E&M, FXS, FXO)		
		E1 R2		
		T1/E1 QSIG		Basic calls only
		T1/E1 PRI		
	MGCP	FXS	Loopstart only	Basic calls only
		FXO	Loopstart or groundstart	No caller ID
		BRI		User side only; no QSIG support

Table 10-1 Overview of Supported Voice Gateways, Protocols, Trunk Interfaces, and Port Types (continued)

Gateway Model	Supported Signaling Protocols	Trunk Interfaces	Port Types	Notes
		T1 CAS (E&M)		
		T1/E1 QSIG		Supplementary Services
		T1/E1 PRI		
Cisco 2600, 2600XM series, 2691, 3700 series	H.323 and SIP	FXS	Loopstart or groundstart	
		FXO	Loopstart or groundstart	Basic calls only
		E&M		
		Analog DID		
		CAMA		
		BRI		
		BRI QSIG		Basic calls only
		T1 CAS (E&M, FXS, FXO)		
		T1 FGD		
		E1 CAS		
		E1 R2		
		T1/E1 QSIG		Basic calls only
		T1/E1 PRI		
		T1 PRI NFAS		
		T1 PRI (Megacom/SDN)		Per T1 port only; not per call
	MGCP	FXS	Loopstart only	Basic calls only
		FXO	Loopstart or groundstart	No caller ID
		BRI		Only 2600XM/2691 support MGCP BRI; User side only; no QSIG support
		T1 CAS (E&M)		
		T1/E1 QSIG		Supplementary Services
		T1/E1 PRI		
		T1 PRI (Megacom/SDN)		Per call
	SCCP	BRI		DoD STE BRI phones only; single-B-channel

Table 10-1 Overview of Supported Voice Gateways, Protocols, Trunk Interfaces, and Port Types (continued)

Gateway Model	Supported Signaling Protocols	Trunk Interfaces	Port Types	Notes
Cisco 3600 series	H.323 and SIP	FXS	Loopstart or groundstart	Basic calls only
		FXO	Loopstart or groundstart	
		E&M		
		Analog DID		
		CAMA		
		BRI		
		BRI QSIG		Basic calls only
		T1 CAS (E&M, FXS, FXO)		
		T1 FGD		
		E1 CAS		
		E1 R2		
		T1/E1 QSIG		Basic calls only
		T1/E1 PRI		
		T1 PRI NFAS		
		T1 PRI (Megacom/SDN)		Per T1 port only; not per call
	MGCP	FXS	Loopstart only	Basic calls only
		FXO	Loopstart or groundstart	No caller ID
		BRI		Only 3640/3660 and some interface cards support MGCP BRI; User side only; no QSIG support
		T1 CAS (E&M)		
		T1/E1 QSIG		Supplementary Services
		T1/E1 PRI		
		T1 PRI (Megacom/SDN)		Per call
Cisco 2800 and 3800 series	H.323 and SIP	FXS	Loopstart or groundstart	Basic calls only
		FXO	Loopstart or groundstart	
		E&M		

Table 10-1 Overview of Supported Voice Gateways, Protocols, Trunk Interfaces, and Port Types (continued)

Gateway Model	Supported Signaling Protocols	Trunk Interfaces	Port Types	Notes
		Analog DID		
		CAMA		
		BRI		
		BRI QSIG		Basic calls only
		T1 CAS (E&M, FXS, FXO)		
		T1 FGD		
		E1 CAS		
		E1 R2		
		T1/E1 QSIG		Basic calls only
		T1/E1 PRI		
		T1 PRI NFAS		
		T1 PRI (Megacom/SDN)		Per T1 port only; not per call
	MGCP	FXS	Loopstart only	Basic calls only
		FXO	Loopstart or groundstart	No caller ID
		BRI		User side only; no QSIG support
		T1 CAS (E&M)		
		T1/E1 QSIG		Supplementary Services
		T1/E1 PRI		
		T1 PRI (Megacom/SDN)		Per call
	SCCP	FXS		
Cisco 7200 series	H.323	T1 CAS (E&M, FXS, FXO)		
		T1 FGD		
		E1 R2		
		T1/E1 QSIG		Basic calls only
		T1/E1 PRI		
Cisco 5000 series	H.323 and SIP	T1 CAS (E&M, FXS, FXO)		
		T1 FGB		
		T1 FGD		
		E1 R2		

Table 10-1 Overview of Supported Voice Gateways, Protocols, Trunk Interfaces, and Port Types (continued)

Gateway Model	Supported Signaling Protocols	Trunk Interfaces	Port Types	Notes
		T1/E1 QSIG		Basic calls only
		T1/E1 PRI		
Cisco Standalone Voice Gateways				
Cisco VG224 Analog Gateway	H.323, MGCP, and SIP	FXS		Basic calls only
	SCCP	FXS		Supplementary Services
Cisco VG248 Analog Gateway	SCCP	FXS		Supplementary Services
Cisco VG200 Gateway	H.323 and SIP	FXS	Loopstart or groundstart	Basic calls only
		FXO	Loopstart or groundstart	
		E&M		
		Analog DID		
		CAMA		
		BRI		
		BRI QSIG		
		T1 CAS (E&M, FXS, FXO)		
		T1 FGD		
		E1 CAS		
		E1 R2		
		T1/E1 QSIG		Basic calls only
		T1/E1 PRI		
		T1 PRI NFAS		
		T1 PRI (Megacom/SDN)		Per T1 port only; not per call
	MGCP	FXS	Loopstart only	Basic calls only
		FXO	Loopstart or groundstart	No caller ID
		BRI		Only 2600XM/2691 support MGCP BRI; User side only; no QSIG support
		T1 CAS (E&M)		
		T1/E1 QSIG		Supplementary Services

Table 10-1 Overview of Supported Voice Gateways, Protocols, Trunk Interfaces, and Port Types (continued)

Gateway Model	Supported Signaling Protocols	Trunk Interfaces	Port Types	Notes
		T1/E1 PRI		
		T1 PRI (Megacom/SDN)		Per call
Cisco Access Digital Trunk Gateway DE-30+	MGCP	E1 PRI		
		E1 QSIG		Supplementary Services
Cisco Access Digital Trunk Gateway DT-24+	MGCP	T1 CAS (E&M)		
		T1 PRI		
		T1 QSIG		Supplementary Services
Cisco Access Analog Trunk Gateway (AT-2, AT-4, AT-8)	SCCP	FXO	Loop start	
Cisco Access Analog Station Gateway (AS-2, AS-4, AS-8)	SCCP	FXS		Supplementary Services
Cisco Catalyst Voice Gateway Modules				
Cisco Communication Media Module (WS-X6600-24FXS)	MGCP or H.323	FXS		Basic calls only
Cisco Communication Media Module (WS-X6600-6T1)	H.323	T1 CAS (E&M)		
		T1 PRI		
		T1 QSIG		Basic calls only
		T1 PRI NFAS		
	MGCP	T1 CAS (E&M)		
		T1 QSIG		Supplementary Services
		T1 PRI		
Cisco Communication Media Module (WS-X6600-6E1)	H.323	E1 PRI		
		E1 QSIG		Basic calls only
		E1 R2		
	MGCP	E1 PRI		
		E1 QSIG		Supplementary Services
Cisco Catalyst 4000 Access Gateway Module (WS-X4604-GWY)	H.323	FXS	Loopstart or groundstart	Basic calls only

Table 10-1 Overview of Supported Voice Gateways, Protocols, Trunk Interfaces, and Port Types (continued)

Gateway Model	Supported Signaling Protocols	Trunk Interfaces	Port Types	Notes
		FXO	Loopstart or groundstart	
		T1 CAS		
		T1/E1 QSIG		Basic calls only
		T1/E1 PRI		
	MGCP	FXS		Basic calls only
		FXO		
		T1 CAS (E&M)		
		T1/E1 QSIG		Supplementary Services
		T1/E1 PRI		
Cisco Catalyst 4224 Voice Gateway Switch	H.323	FXS		Basic calls only
		FXO		
		BRI		
		T1 CAS		
		E1 R2		
		T1/E1 QSIG		
		T1/E1 PRI		
Cisco Catalyst 6000 8-Port Voice T1/E1 and Services Module (WS-X6608-T1)	MGCP	T1 CAS (E&M)		
		T1 PRI		
		T1 QSIG		Supplementary Services
Cisco Catalyst 6000 8-Port Voice T1/E1 and Services Module (WS-X6608-E1)	MGCP	E1 PRI		
		E1 QSIG		Supplementary Services
Cisco Catalyst 6000 24-Port FXS Analog Interface Module (WS-X6624-FXS)	MGCP	FXS	Loopstart only	Basic calls only

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 Cisco Unified Communications Solutions have dial plans that are configured in different places:

- Configure dial plan information for both Skinny and MGCP gateways in the Cisco Unified CallManager.
- Configure dial plans in Cisco Unified 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. Cisco Unified CallManager restricts the gateways that you can include in the same route group and the route groups that you can include in the same route list. For more information about routing, see the "Route Plan Overview" section on page 7-4.

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 and can override what was configured in the route pattern.

You can configure H.323 trunks, *not* H.323gateways, 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 Unified CallManager" section in the Cisco Unified CallManager System Guide, Release 5.0(4) 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 Unified CallManager cluster.

Related Topics

- Dependency Records for Gateways and their Route Groups and Directory Numbers, page 10-16
- Cisco Voice Gateways, page 10-1

Dependency Records for Gateways and their 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 Unified 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. If the dependency records are not enabled for the system, the dependency records summary window displays a message.

For more information about Dependency Records, refer to the "Accessing Dependency Records", "Deleting Gateways", and "Removing a Directory Number from a Phone" sections in the *Cisco Unified CallManager Administration Guide, Release 5.0(4)*.

Related Topics

- Gateways, Dial Plans, and Route Groups, page 10-16
- Cisco Voice Gateways, page 10-1

Gateway Failover and Fallback

This section describes how these Cisco voice gateways handle Cisco Unified CallManager failover and fallback situations. See the following topics:

- MGCP Gateways, page 10-17
- IOS H.323 Gateways, page 10-17
- Cisco VG248 Analog Phone Gateway, page 10-18

MGCP Gateways

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

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

Note Skinny Client Control Protocol (SCCP) gateways handle Cisco Unified CallManager redundancy, failover, and fallback in the same way as MGCP gateways.

IOS H.323 Gateways

Cisco IOS gateways also handle Cisco Unified 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 Unified CallManagers. The 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 Unified CallManager, it tries a second Cisco Unified 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 FastEthernet0/0
    ip address 10.1.1.10 255.255.255.0
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 (SCCP) for clustering and failover.

Transferring Calls Between Gateways

Using Cisco Unified CallManager Administration, you can configure gateways as OnNet (internal) gateways or OffNet (external) gateways by using Gateway Configuration or by setting a clusterwide service parameter. Used in conjunction with the clusterwide service parameter, Block OffNet to OffNet Transfer, the configuration determines whether calls can be transferred over a gateway.

To use the same gateway to route both OnNet and OffNet calls, associate the gateway with two different route patterns. Make one gateway OnNet and the other OffNet with both having the Allow Device Override check box unchecked.

Configuring Transfer Capabilities Using Gateway Configuration

Using Cisco Unified CallManager Administration Gateway Configuration, you can configure a gateway as OffNet or OnNet. The system considers the calls that come to the network through that gateway OffNet or OnNet, respectively. Use the Gateway Configuration window field, Call Classification, to configure the gateway as OffNet, OnNet, or Use System Default. See Table 10-2 for description of these settings.

The Route Pattern Configuration window provides a drop-down list box called Call Classification, which allows you to configure a route pattern as OffNet or OnNet. When Call Classification is set to OffNet and the Allow Device Override check box is unchecked, the system considers the outgoing calls that use this route pattern as OffNet (if configured as OnNet and check box is unchecked, then outgoing calls are considered OnNet).

The same gateway can be used to route both OnNet and OffNet calls by associating the gateway with two different route patterns: one OnNet and the other OffNet, with both having the Allow Device Override check box unchecked. For outgoing calls, the outgoing device setting classifies the call as either OnNet or OffNet by determining whether the Allow Device Override check box is checked.

In route pattern configuration, if the Call Classification is set as OnNet, the Allow Device Override check box is checked, and the route pattern is associated with an OffNet gateway, the system considers the outgoing call OffNet.

Table 10-2 Gateway Configuration Call Classification Settings

Setting Name	Description
OffNet	This setting identifies the gateway as being an external gateway. When a call comes in from a gateway that is configured as OffNet, the outside ring gets sent to the destination device.
OnNet	This setting identifies the gateway as being an internal gateway. When a call comes in from a gateway that is configured as OnNet, the inside ring gets sent to the destination device.
Use System Default	This setting uses the Cisco Unified CallManager clusterwide service parameter Call Classification.

Configuring Transfer Capabilities by Using Call Classification Service Parameter

To configure all gateways to be OffNet (external) or OnNet (internal), perform the following two steps:

1. Use the Cisco Unified CallManager clusterwide service parameter Call Classification.
2. Configure individual gateways to Use System Default in the Call Classification field that is on the Gateway Configuration window.

Blocking Transfer Capabilities by Using Service Parameters

Block transfer provides a way of restricting transfer between external devices, so fraudulent activity gets prevented. You can configure the following devices as OnNet (internal) or OffNet (external) to Cisco Unified CallManager:

- H.323 gateway
- MGCP FXO trunk
- MGCP T1/E1 trunk
- Intercluster trunk
- SIP trunk

If you do not want OffNet calls to be transferred to an external device (one that is configured as OffNet), set the Cisco Unified CallManager clusterwide service parameter, Block OffNet to OffNet Transfer, to True.

If a user tries to transfer a call on an OffNet gateway that is configured as blocked, a message displays on the user phone to indicate that the call cannot be transferred.

Related Topics

- Route Pattern Configuration, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
- Gateway Configuration, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
- Trunk Configuration, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*

Gateway Configuration Checklist

Table 10-3 provides an overview of the steps that are required to configure gateways in Cisco Unified CallManager, along with references to related procedures and topics.

Table 10-3 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 Unified CallManager.	Gateway Configuration Settings, *Cisco Unified CallManager Administration Guide, Release 5.0(4)* Port Configuration Settings, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
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 Unified CallManager Administration.	Adding Gateways to Cisco Unified CallManager, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
Step 5	Add and configure ports on the gateway or add and configure the Cisco VG248 Analog Phone Gateway.	Port Configuration Settings, *Cisco Unified CallManager Administration Guide, Release 5.0(4)* Adding a Cisco VG248 Analog Phone Gateway, *Cisco Unified CallManager Administration Guide, Release 5.0(4)* Cisco Unified IP Phone Configuration, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
Step 6	For FXS ports, add directory numbers, if appropriate.	Directory Number Configuration Overview, *Cisco Unified CallManager Administration Guide, Release 5.0(4)* Directory Number Configuration Settings, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*

Table 10-3 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 configuration can include setting up a route group, route list, and route pattern for the Gateway in Cisco Unified CallManager or, for some gateways, configuring the dial plan on the gateway itself.	*Cisco Unified Communications Solution Reference Network Design* *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
Step 8	Reset the gateway to apply the configuration settings.	Resetting and Restarting Gateways, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*

Tip To get to the default web pages for many 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.

MGCP BRI Gateway Configuration Checklist

Table 10-4 provides an overview of the steps that are required to configure a BRI gateway in Cisco Unified CallManager, along with references to related procedures and topics.

Table 10-4 MGCP BRI Gateway Configuration Checklist

Configuration Steps		Procedures and Related Topics
Step 1	Install and configure the gateway and voice modules 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 Unified CallManager and to configure the trunk interface to the PSTN or external non-IP telephony device.	Gateway Configuration Checklist, page 10-20 Adding a BRI Port to an MGCP Gateway, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
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 Unified CallManager Administration.	Gateway Configuration, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
Step 5	Add and configure ports on the gateway.	Gateway Configuration, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*

Table 10-4 MGCP BRI Gateway Configuration Checklist (continued)

Configuration Steps		Procedures and Related Topics
Step 6	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 Unified CallManager or, for some gateways, configuring the dial plan on the gateway itself.	*Cisco Unified CallManager Administration Guide, Release 5.0(4)* *Cisco Unified Communications Solution Reference Network Design (SRND)*
Step 7	Reset the gateway to apply the configuration settings.	*Cisco Unified CallManager Administration Guide, Release 5.0(4)*

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 specifies 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

- Understanding IP Telephony Protocols, *Cisco Unified CallManager System Guide, Release 5.0(4)*
- Understanding Cisco Unified CallManager Trunk Types, *Cisco Unified CallManager System Guide, Release 5.0(4)*
- Route Plan Overview, *Cisco Unified CallManager System Guide, Release 5.0(4)*
- Gatekeepers and Trunks, *Cisco Unified CallManager System Guide, Release 5.0(4)*
- Gateway Configuration, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
- Adding Gateways to Cisco Unified CallManager, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
- Gateway Configuration Settings, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
- Port Configuration Settings, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*
- Directory Number Configuration Settings, *Cisco Unified CallManager Administration Guide, Release 5.0(4)*

Additional Cisco Documentation

- *Cisco Unified Communications Solution Reference Network Design*
- *Configuring Cisco Unified Communications 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*