Gateways

Revised: April 2, 2013; OL-30952-01

Gateways provide a number of methods for connecting a network of collaboration endpoints to the Public Switched Telephone Network (PSTN), a legacy PBX, or external systems. Voice and video gateways range from entry-level and standalone platforms to high-end, feature-rich integrated routers and chassis-based systems.

This chapter explains important factors to consider when selecting a Cisco gateway to provide the appropriate protocol and feature support for your voice and video network. The main topics discussed in this chapter include:

- Understanding Cisco Gateways, page 5-2
- Gateway Selection, page 5-3
- Gateways for Video Telephony, page 5-12
- Gateways Best Practices, page 5-15
- Fax and Modem Support, page 5-21

What's New in This Chapter

Table 5-1 lists the topics that are new in this chapter or that have changed significantly from previous releases of this document.

Table 5-1   New or Changed Information Since the Previous Release of This Document

New or Revised Topic	Described in	Revision Date
Added information on video gateways	Various sections throughout this chapter	April 2, 2013
Removed information on fax and modem support	Fax and Modem Support, page 5-21	April 2, 2013
Removed information on the Cisco Unified Videoconferencing 3500 Gateway	For this information, refer to the *Gateways* chapter of the *Cisco Unified Communications System 9.0 SRND*, available at http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/9x/gateways.html	April 2, 2013
No changes for Cisco Unified Communications System Release 9.0		June 28, 2012

Understanding Cisco Gateways

This section presents an overview of the following types of gateways:

TDM and IP Trunking Gateways

Until approximately 2006, the only way for an enterprise to connect its internal voice and video network to services outside the enterprise was by means of TDM gateways to the traditional PSTN. Cisco offers a full range of TDM and serial gateways with analog and digital connections to the PSTN as well as to PBXs and external systems. TDM connectivity covers a wide variety of low-density analog (FXS and FXO), low density digital (BRI), and high-density digital (T1, E1, and T3) interface choices.

Starting around 2006, new voice and video service options to an enterprise became available from service providers, often as SIP trunk services. Using a SIP trunk for connecting to PSTN and other destinations outside the enterprise involves an IP-to-IP connection at the edge of the enterprise's network. The same functions traditionally fulfilled by a TDM or serial gateway are still needed at this interconnect point, including demarcation, call admission control, QoS, troubleshooting boundary, security checks, and so forth. For voice and video SIP trunking connections, the Cisco Unified Border Element and the Cisco Video Communication Server (VCS) fulfills these functions as an interconnect point between the enterprise and the service provider network.

The remainder of this chapter discusses in detail Cisco TDM gateway platforms. A brief section also covers the Cisco serial gateways. Cisco Unified Border Element and Cisco Video Communication Server are discussed in greater detail in the chapter on Cisco Unified CM Trunks, page 6-1.

Cisco gateways enable voice and video endpoints to communicate with non-IP and external telecommunications devices. There are two types of Cisco gateways, analog and digital. Both types support voice calls, but only digital gateways support video.

Cisco Analog Gateways

There are two categories of Cisco analog gateways, station gateways and trunk gateways.

- Analog station gateways

  Analog station gateways connect Unified CM to Plain Old Telephone Service (POTS) analog telephones, interactive voice response (IVR) systems, fax machines, and voice mail systems. Station gateways provide Foreign Exchange Station (FXS) ports.

- Analog trunk gateways

  Analog trunk gateways connect Unified CM to PSTN central office (CO) or PBX trunks. Analog trunk gateways provide Foreign Exchange Office (FXO) ports for access to the PSTN, PBXs, or key systems, and E&M (recEive and transMit, or ear and mouth) ports for analog trunk connection to a legacy PBX. Analog Direct Inward Dialing (DID) and Centralized Automatic Message Accounting (CAMA) are also available for PSTN connectivity.

Cisco analog gateways are available on the following products and series:

- Cisco Voice Gateways VG204, VG224, and VG350
- Cisco Integrated Services Routers Generation 2 (ISR G2) 1900, 2900, and 3900 Series with appropriate PVDMs and service modules or cards

Cisco Digital Trunk Gateways

A Cisco digital trunk gateway connects Unified CM to the PSTN or to a PBX via digital trunks such as Primary Rate Interface (PRI), Basic Rate Interface (BRI), serial interfaces (V.35, RS-449, and EIA-530), or T1 Channel Associated Signaling (CAS). Digital T1 PRI and BRI trunks can be used for both video and audio-only calls.

Cisco digital trunk gateways are available on the following products and series:

- Cisco Integrated Services Routers Generation 2 (ISR G2) 1900, 2900, and 3900 Series with appropriate PVDMs and service modules or cards
- Cisco TelePresence ISDN GW 3241 and MSE 8321
- Cisco TelePresence Serial GW 3340 and MSE 8330

Cisco TelePresence ISDN Link

The Cisco TelePresence ISDN Link is a compact appliance for in-room ISDN and external network connectivity supporting Cisco TelePresence EX, MX, SX, and C Series endpoints. While traditional voice and video gateways are shared resources that provide connectivity between the IP network and the PSTN for many endpoints, each Cisco ISDN Link is paired with a single Cisco endpoint. For more information, refer to Cisco TelePresence ISDN Link documentation available at

http://www.cisco.com/en/US/products/ps12504/tsd_products_support_series_home.html

Gateway Selection

When selecting a gateway for your voice and video network, consider the following factors:

- Core Feature Requirements, page 5-3
- Gateway Protocols for Call Control, page 5-4
- Core Feature Requirements, page 5-6

Core Feature Requirements

Gateways used by voice and video endpoints must meet the following core feature requirements:

- Dual tone multifrequency (DTMF) relay capabilities
- Supplementary services support

  Supplementary services are basic telephony functions such as hold, transfer, and conferencing.

- Fax/modem support

  Fax over IP enables interoperability of traditional analog fax machines with IP telephony networks. The fax image is converted from an analog signal and is carried as digital data over the packet network.

- Redundancy support

  Cisco Unified Communications is based on a distributed model for high availability. Unified CM clusters provide for Unified CM redundancy. The gateways must support the ability to "re-home" to a secondary Unified CM in the event that a primary Unified CM fails. Some gateways may register to a Cisco VCS, in which case the gateway must support the ability to "re-home" to a secondary Cisco VCS if the primary fails.

Refer to the gateway product documentation to verify that any gateway you select for an enterprise deployment can support the preceding core requirements. Additionally, every collaboration implementation has its own site-specific feature requirements, such as analog or digital access, DID, and capacity requirements.

Gateway Protocols for Call Control

Cisco Unified Communications Manager (Unified CM) supports the following IP protocols for gateways:

- Session Initiation Protocol (SIP)
- H.323
- Media Gateway Control Protocol (MGCP)
- Skinny Client Control Protocol (SCCP)

Cisco Video Communication Server (VCS) supports the following IP protocols for gateways:

- Session Initiation Protocol (SIP)
- H.323

SIP is the recommended call signaling protocol because it aligns with the overall Cisco Collaboration solution and the direction of new voice and video products. However, protocol selection might depend on site-specific requirements and the current installed base of equipment. Existing deployments might be limited by the gateway hardware or require a different signaling protocol for a specific feature.

For example, Simplified Message Desk Interface (SMDI) is a standard for integrating voice mail systems to PBXs or Centrex systems. Connecting to a voice mail system via SMDI and using either analog FXS or digital T1 PRI would require either SCCP or MGCP protocol because H.323 or SIP devices do not identify the specific line being used from a group of ports. Use of H.323 or SIP gateways for this purpose means the Cisco Message Interface cannot correctly correlate the SMDI information with the actual port or channel being used for an incoming call.

Likewise, placement of certain Cisco video gateways within the network depends upon the existing call control architecture. Both the Cisco ISDN and serial gateways are optimized for video calls and were initially designed to work with the Cisco VCS. The Cisco TelePresence Serial Gateway 8330 and 3340 platforms are recommended to register with a Cisco VCS using H.323, as shown in Figure 5-1.

Figure 5-1   *Cisco TelePresence Serial Gateway Registered to Cisco VCS*
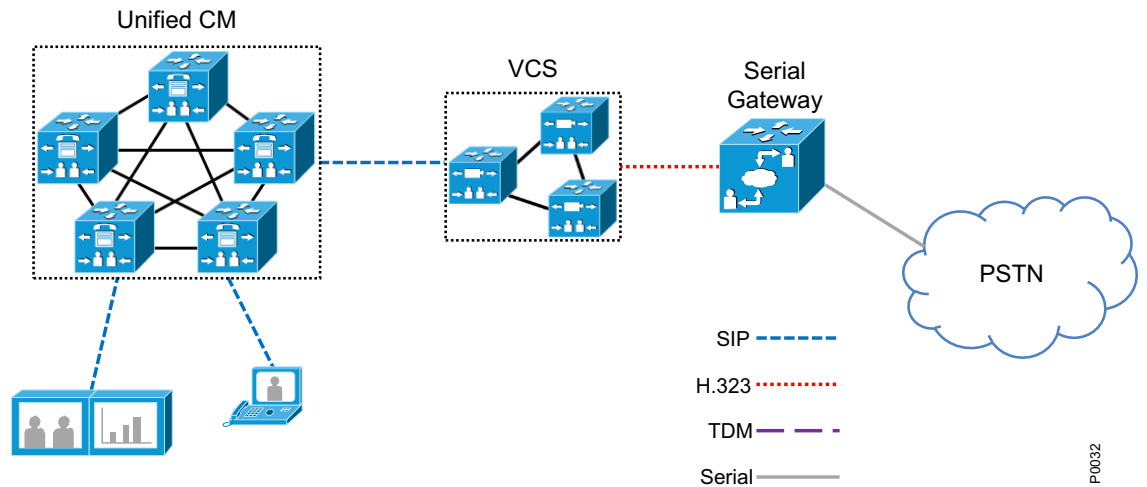
The Cisco TelePresence ISDN Gateway 8321 and 3241 support SIP beginning with version 2.2 and later. This means that the Cisco 8321 and 3241 gateways can either register to VCS using H.323 (as shown in Figure 5-2) or trunk directly to Unified CM using SIP (as shown in Figure 5-3).

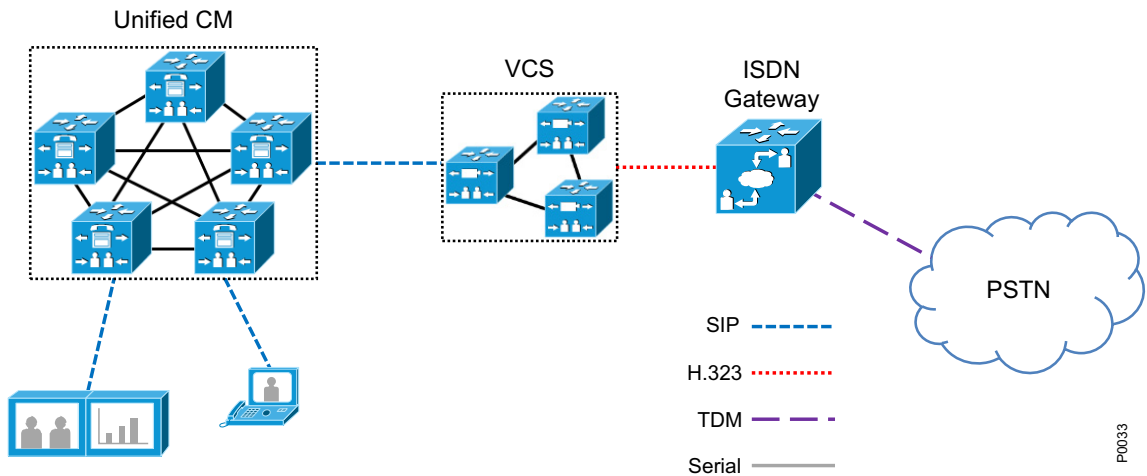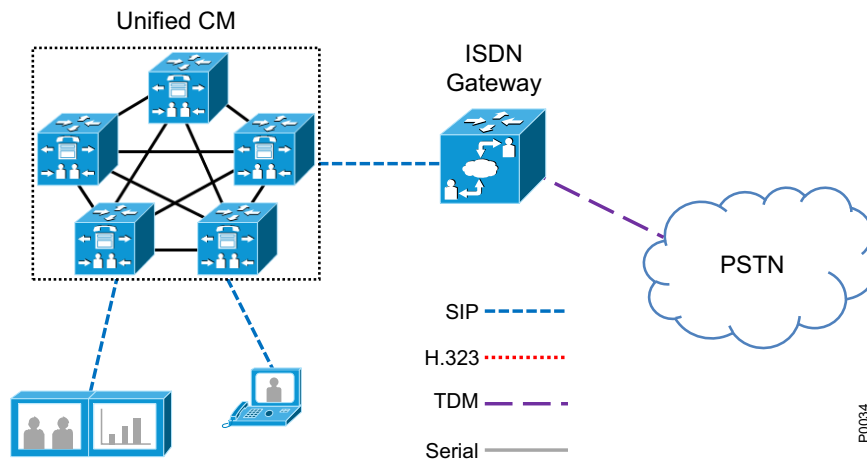*Figure 5-2*   *Cisco TelePresence ISDN Gateway Registered to Cisco VCS*

Figure 5-3   *Cisco TelePresence ISDN Gateway Registered to Cisco Unified CM*

In addition, the Unified CM deployment model being used can influence gateway protocol selection. (Refer to the chapter on Collaboration Deployment Models, page 10-1.)

Core Feature Requirements

This section describes how each protocol (SCCP, H.323, MGCP, and SIP) supports the following gateway feature requirements:

- DTMF Relay, page 5-6
- Supplementary Services, page 5-7
- Unified CM Redundancy, page 5-10

DTMF Relay

Dual-Tone Multifrequency (DTMF) is a signaling method that uses specific pairs of frequencies within the voice band for signals. A 64 kbps pulse code modulation (PCM) voice channel can carry these signals without difficulty. However, when using a low bite-rate codec for voice compression, the potential exists for DTMF signal loss or distortion. An out-of-band signaling method for carrying DTMF tones across an IP infrastructure provides an elegant solution for these codec-induced symptoms.

SCCP Gateways

The Cisco VG248 carries DTMF signals out-of-band using Transmission Control Protocol (TCP) port 2002. Out-of-band DTMF is the default gateway configuration mode for the VG248.

H.323 Gateways

H.323 gateways, such as the Cisco 3900 Series products, can communicate with Unified CM using the enhanced H.245 capability for exchanging DTMF signals out-of-band. This capability is enabled through the command line interface (CLI) of the 3900 Series gateway and the **dtmf-relay** command available in its dial-peers.

MGCP Gateway

Cisco IOS-based platforms can use MGCP for Unified CM communication. Within the MGCP protocol is the concept of *packages*. The MGCP gateway loads the DTMF package upon start-up. The MGCP gateway sends *symbols* over the control channel to represent any DTMF tones it receives. Unified CM then interprets these signals and passes on the DTMF signals, out-of-band, to the signaling endpoint.

Note   Use the **fm-package** command to enable DTMF via RFC 2833, available as of Cisco IOS Release 12.4(6)T.

SIP Gateway

Cisco IOS and ISDN gateways can use SIP for Unified CM communication. They support various methods for DTMF, but only the following methods can be used to communicate with Unified CM:

- Named Telephony Events (NTE), or RFC 2833
- Unsolicited SIP Notify (UN) (Cisco IOS gateways only)
- Key Press Markup Language (KPML)

The method used for DTMF can be configured using the gateway CLI command **dtmf-relay** under the respective **dial-peer**.

For more details on DTMF method selection, see the section on Calls over SIP Trunks, page 7-11.

Supplementary Services

Supplementary services provide user functions such as hold, transfer, and conferencing. These are considered basic telephony features and are more common in voice calls than in video calls.

SCCP Gateways

The Cisco SCCP gateways provide full supplementary service support. The SCCP gateways use the Gateway-to-Unified CM signaling channel and SCCP to exchange call control parameters.

H.323 Gateways

H.323v2 implements Open/Close LogicalChannel and the emptyCapabilitySet features. The use of H.323v2 by H.323 gateways eliminates the requirement for an MTP to provide supplementary services. A transcoder is allocated dynamically only if required during a call to provide access to G.711-only devices while still maintaining a G.729 stream across the WAN.

Once an H.323v2 call is set up between a Cisco IOS gateway and an IP endpoint, using the Unified CM as an H.323 proxy, the endpoint can request to modify the bearer connection. Because the Real-Time Transport Protocol (RTP) stream is directly connected to the endpoint from the Cisco IOS gateway, a supported media codec can be negotiated.

Figure 5-4 and the following steps illustrate a call transfer between two IP phones:

1. If IP Phone 1 wishes to transfer the call from the Cisco IOS gateway to Phone 2, it issues a transfer request to Unified CM using SCCP.

2. Unified CM translates this request into an H.323v2 CloseLogicalChannel request to the Cisco IOS gateway for the appropriate SessionID.

3. The Cisco IOS gateway closes the RTP channel to Phone 1.

4. Unified CM issues a request to Phone 2, using SCCP, to set up an RTP connection to the Cisco IOS gateway. At the same time, Unified CM issues an OpenLogicalChannel request to the Cisco IOS gateway with the new destination parameters, but using the same SessionID.

5. After the Cisco IOS gateway acknowledges the request, an RTP voice bearer channel is established between Phone 2 and the Cisco IOS gateway.

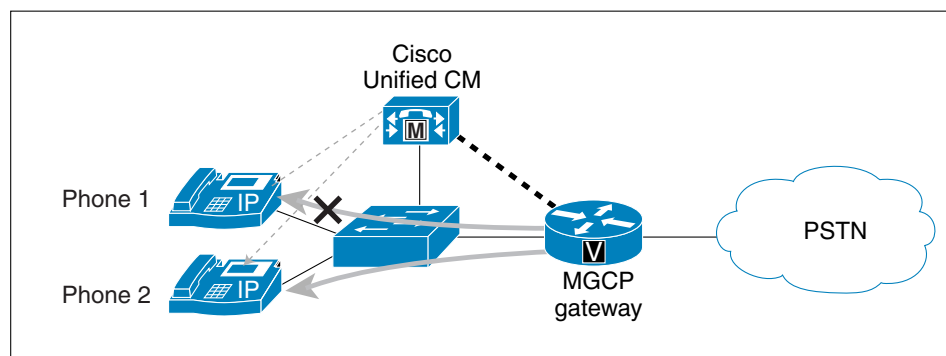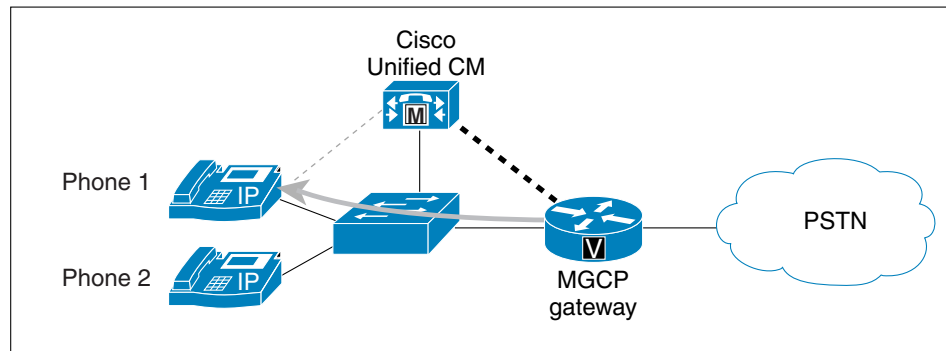*Figure 5-4*   *H.323 Gateway Supplementary Service Support*
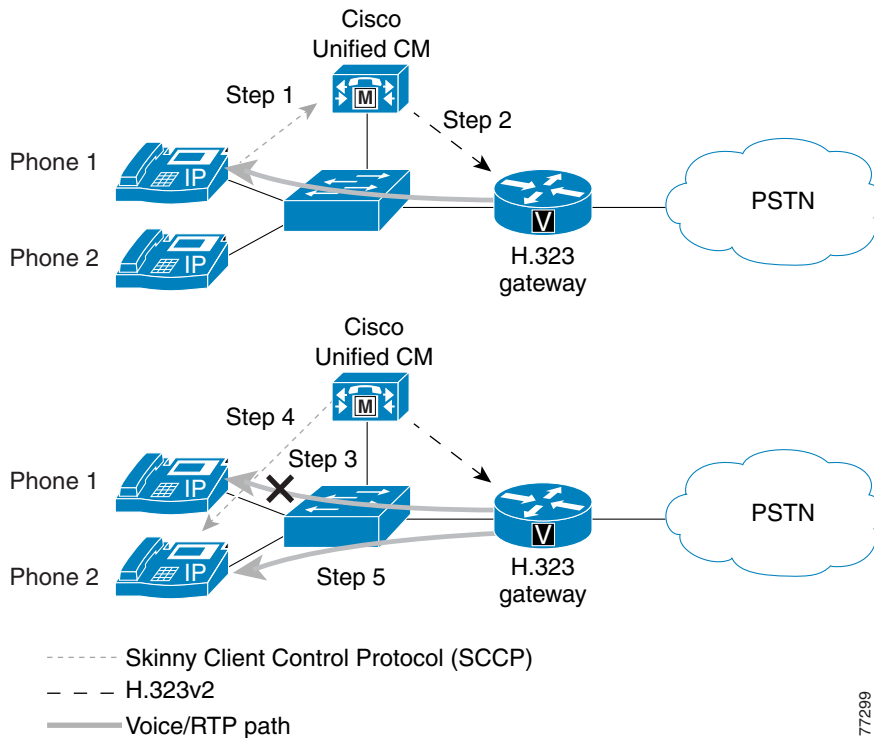
MGCP Gateway

The MGCP gateways provide full support for the hold, transfer, and conference features through the MGCP protocol. Because MGCP is a master/slave protocol with Unified CM controlling all session intelligence, Unified CM can easily manipulate MGCP gateway voice connections. If an IP telephony endpoint (for example, an IP phone) needs to modify the session (for example, transfer the call to another endpoint), the endpoint would notify Unified CM using SCCP. Unified CM then informs the MGCP gateway, using the MGCP User Datagram Protocol (UDP) control connection, to terminate the current RTP stream associated with the Session ID and to start a new media session with the new endpoint information. Figure 5-5 illustrates the protocols exchanged between the MGCP gateway, endpoints, and Unified CM.

Figure 5-5   MGCP Gateway Supplementary Service Support

Direct call from MGCP gateway to IP phone.
MTP is not required.

The MGCP gateway supports supplementary
services such as call transfer.

SIP Gateway

The Unified CM SIP trunk interface to Cisco SIP gateways supports supplementary services such as hold, blind transfer, and attended transfer. The support for supplementary services is achieved via SIP methods such as INVITE and REFER. The corresponding SIP gateway must also support these methods in order for supplementary services to work. For more details, refer to the following documentation:

- *Cisco Unified Communications Manager System Guide*

  http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_maintenance_guides_list.html

- *Cisco IOS SIP Configuration Guide*

  http://www.cisco.com/en/US/docs/ios/voice/sip/configuration/guide/12_4t/sip_12_4t_book.html

- Cisco TelePresence ISDN Gateway documentation

  http://www.cisco.com/en/US/products/ps11448/tsd_products_support_series_home.html

Unified CM Redundancy

An integral piece of the collaboration solution architecture is the provisioning of low-cost, distributed PC-based systems to replace expensive and proprietary legacy PBX systems. This distributed design lends itself to the robust fault tolerant architecture of clustered Unified CMs. Even in its most simplistic form (a two-system cluster), a secondary Unified CM should be able to pick up control of all gateways initially managed by the primary Unified CM.

SCCP Gateways

Upon boot-up, the Cisco VG224, VG248, and ATA 188 gateways are provisioned with Unified CM server information. When these gateways initialize, a list of Unified CMs is downloaded to the gateways. This list is prioritized into a primary Unified CM and secondary Unified CM. In the event that the primary Unified CM becomes unreachable, the gateway registers with the secondary Unified CM.

H.323 VoIP Call Preservation for WAN Link Failures

H.323 call preservation enhancements for WAN link failures sustain connectivity for H.323 topologies where signaling is handled by an entity that is different from the other endpoint, such as a gatekeeper that provides routed signaling or a call agent (such as Cisco Unified CM) that brokers signaling between the two connected parties. Call preservation is useful when a gateway and the other endpoint are located at the same site but the call agent is remote and therefore more likely to experience connectivity failures.

H.323 call preservation covers the following types of failures and connections.

Failure Types:

- WAN failures that include WAN links flapping or degraded WAN links.
- Cisco Unified CM software failure, such as when the ccm.exe service crashes on a Unified CM server.
- LAN connectivity failure, except when a failure occurs at the local branch.

Connection Types:

- Calls between two Cisco Unified CM controlled endpoints under the following conditions:
  - During Unified CM reloads.
  - When a Transmission Control Protocol (TCP) connection used for signaling H.225.0 or H.245 messages between one or both endpoints and Unified CM is lost or flapping.
  - Between endpoints that are registered to different Unified CMs in a cluster, and the TCP connection between the two Unified CMs is lost.
  - Between IP phones and the PSTN at the same site.
- Calls between a Cisco IOS gateway and an endpoint controlled by a softswitch, where the signaling (H.225.0, H.245 or both) flows between the gateway and the softswitch and media flows between the gateway and the endpoint:
  - When the softswitch reloads.
  - When the H.225.0 or H.245 TCP connection between the gateway and the softswitch is lost, and the softswitch does not clear the call on the endpoint.
  - When the H.225.0 or H.245 TCP connection between softswitch and the endpoint is lost, and the softswitch does not clear the call on the gateway.
- Call flows involving a Cisco Unified Border Element running in media flow-around mode that reload or lose connection with the rest of the network.

Note that, after the media is preserved, the call is torn down later when either one of the parties hangs up or media inactivity is detected. In cases where there is a machine-generated media stream, such as music streaming from a media server, the media inactivity detection will not work and then the call might hang. Cisco Unified CM addresses such conditions by indicating to the gateway that such calls should not be preserved, but third-party devices or the Cisco Unified Border Element would not do this.

Flapping is defined for this feature as the repeated and temporary loss of IP connectivity, which can be caused by WAN or LAN failures. H.323 calls between a Cisco IOS gateway and Cisco Unified CM may be torn down when flapping occurs. When Unified CM detects that the TCP connection is lost, it clears the call and closes the TCP sockets used for the call by sending a TCP FIN, without sending an H.225.0 Release Complete or H.245 End Session message. This is called *quiet clearing*. The TCP FIN sent from Unified CM could reach the gateway if the network comes up for a short duration, and the gateway will tear down the call. Even if the TCP FIN does not reach the gateway, the TCP keepalives sent from the gateway could reach Unified CM when the network comes up. Unified CM will send TCP RST messages in response to the keepalives because it has already closed the TCP connection. The gateway will tear down H.323 calls if it receives the RST message.

Configuration of H.323 call preservation enhancements for WAN link failures involves configuring the **call preserve** command. If you are using Cisco Unified CM, you must enable the Allow Peer to Preserve H.323 Calls parameter from the Service Parameters window.

The **call preserve** command causes the gateway to ignore socket closure or socket errors on H.225.0 or H.245 connections for active calls, thus allowing the socket to be closed without tearing down calls using those connections.

MGCP Gateway

MGCP gateways also have the ability to fail over to a secondary Unified CM in the event of communication loss with the primary Unified CM. When the failover occurs, active calls are preserved.

Within the MGCP gateway configuration file, the primary Unified CM is identified using the **call-agent <hostname>** command, and a list of secondary Unified CM is added using the **ccm-manager redundant-host** command. Keepalives with the primary Unified CM are through the MGCP application-level keepalive mechanism, whereby the MGCP gateway sends an empty MGCP notify (NTFY) message to Unified CM and waits for an acknowledgement. Keepalive with the backup Unified CMs is through the TCP keepalive mechanism.

If the primary Unified CM becomes available at a later time, the MGCP gateway can "re-home," or switch back to the original Unified CM. This re-homing can occur either immediately, after a configurable amount of time, or only when all connected sessions have been released.

SIP Gateway

Redundancy with Cisco IOS SIP gateways can be achieved similarly to H.323. If the SIP gateway cannot establish a connection to the primary Unified CM, it tries a second Unified CM defined under another dial-peer statement with a higher preference.

By default the Cisco IOS SIP gateway transmits the SIP INVITE request 6 times to the Unified CM IP address configured under the dial-peer. If the SIP gateway does not receive a response from that Unified CM, it will try to contact the Unified CM configured under the other dial-peer with a higher preference.

Cisco IOS SIP gateways wait for the SIP 100 response to an INVITE for a period of 500 ms. By default, it can take up to 3 seconds for the Cisco IOS SIP gateway to reach the backup Unified CM. You can change the SIP INVITE retry attempts under the **sip-ua** configuration by using the command **retry invite** *<number>*. You can also change the period that the Cisco IOS SIP gateway waits for a SIP 100 response to a SIP INVITE request by using the command **timers trying** *<time>* under the **sip-ua** configuration.

One other way to speed up the failover to the backup Unified CM is to configure the command **monitor probe icmp-ping** under the **dial-peer** statement. If Unified CM does not respond to an Internet Control Message Protocol (ICMP) echo message (ping), the dial-peer will be shut down. This command is useful only when the Unified CM is not reachable. ICMP echo messages are sent every 10 seconds.

The Cisco ISDN Gateway can connect to Unified CM via SIP trunk starting with Unified CM release 9.0 and ISDN Gateway release 2.2 and later. The ISDN Gateway SIP configuration consists of entering an IP address, hostname, DNS A record, or DNS SRV record for outbound SIP connections. Redundancy can be achieved by utilizing DNS SRV records with appropriate weight and priority so that, if the primary Unified CM fails, the ISDN Gateway will send outbound SIP calls to the secondary Unified CM.

Gateways for Video Telephony

Video gateways terminate video calls into an IP telephony network or the PSTN. Video gateways are different from voice gateways because they have to interact with the ISDN or serial links that support video and convert that call to a video call on the IP network using protocols such as H.323 or SIP. Enterprises can consider separate gateways for voice calls and video calls, or they can have integrated gateways that route both voice and video calls.

The following key considerations can help you decide if you need separate gateways for voice and video or an integrated gateway:

- Dial plan — If the enterprise has the flexibility of a separate dial plan for video users, it can use separate video gateways that allow it to keep existing enterprise dial plans.

- Video users — If the enterprise has a large number of users who primarily use voice rather than video, then Cisco recommends using separate video gateways to service the video call users.

- Locations — If the enterprise has a large number of distributed locations with video users at many locations, then Cisco recommends using an integrated gateway to reduce total cost of ownership (TCO).

- Additional video capabilities such as video IVR, auto attendant, and bonding across trunks — Dedicated video gateways support advanced features that integrated gateways do not support.

- Protocol — Gateway protocol can be an important factor to align with enterprise policies and standards.

- Device management — Ease of maintenance, management, and troubleshooting can be an important factor. Dedicated gateways provide a better user interface (GUI) for management and configuration, while integrated gateways can provide better troubleshooting. However, these factors are dependent on the respective products.

Dedicated Video Gateways

Enterprises that have an extensive voice infrastructure with voice gateways can add dedicated video gateways so that users can make video calls through them to the PSTN. The Cisco ISDN Gateway and Serial Gateways are examples of dedicated video gateways. Although these products support audio-only calls, they were designed specifically with video users in mind. They support a wide range of video-centric protocols and features.

Figure 5-6 shows an enterprise deployment that can use existing protocols for its voice gateways and add video gateways so that Unified CM users can make voice and video calls to the PSTN.

Figure 5-6   Unified CM System with Separate PSTN Lines for Voice and IP Video Telephony
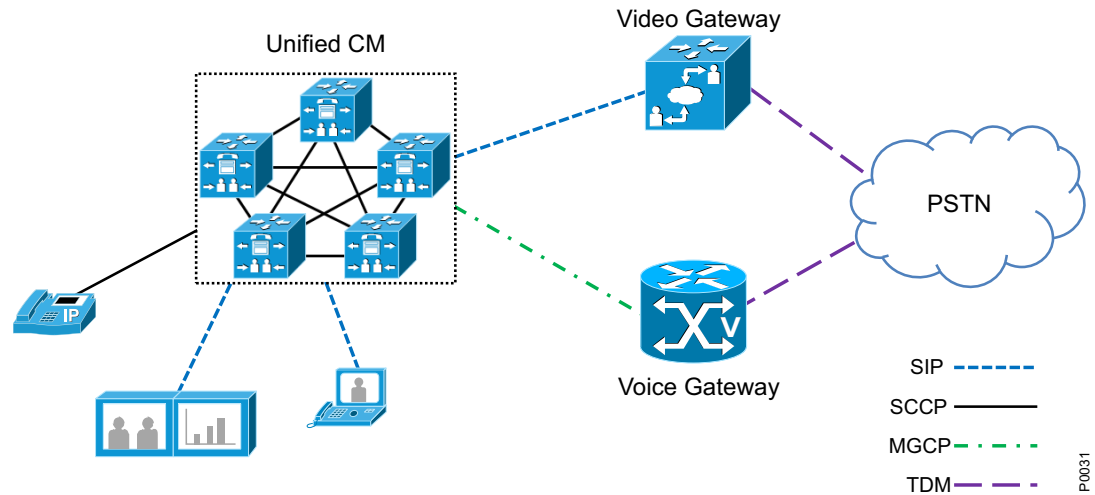
The Cisco video gateways, while excellent for video calls, do not support all of the telephony features that Cisco voice gateways offer. Cisco video gateways have the following characteristics:

- The Serial Gateway supports only H.323 for IP connectivity.
- The ISDN Gateway supports H.323 and SIP (starting with release 2.2) for IP connectivity.
- They support T1/E1-PRI, BRI, V.35, RS-449, and EIA-530.
- They support H.261, H.263, H.263+, and H.264 video codecs.
- They support G.711, G.722, G.722.1, and G.728; but they do not support G.729 audio.
- They support H.320, H.233, H.234, H.235 (AES), H.239, H.221, FTP, RTP, HTTP, HTTPS, DHCP, SNMP, and NTP.

As a result of these differences in the products, the Cisco TDM and Serial Gateways are not recommended as replacements for Cisco voice gateways. IP Telephony customers who want to add video to their communications environment should deploy both types of gateways and use the Cisco voice gateways for all voice calls and use the Cisco video gateways for video calls only. Customers might also have to procure separate circuits for voice and video from their PSTN service provider, depending on which model of Cisco gateway is deployed.

Also consider how calls will be routed across the IP network to a remote gateway for the purpose of providing toll bypass, and how calls will be re-routed over the PSTN in the event that the IP network is unavailable or does not have enough bandwidth to complete the call. More specifically, do you want to invoke automated alternate routing (AAR) for video calls?

Integrated Video Gateways

Although not recommended, enterprises may consider an integrated device for voice and video gateway functionality. This provides the enterprise the advantages of managing fewer devices and keeping the dial plan simple. The gateway processes the call as a voice call if it is voice and as a video call if it is video.

Cisco IOS, ISDN, and Serial Video gateway have the following characteristics:

- Provide H.323 and SIP support (except Serial Gateway, which is H.323 only)
- Supports H.261, H.263, H.263+, and H.264 video codec
- Provides extensive called and calling transformation capabilities
- Provides extensive logging and troubleshooting capabilities

The following considerations apply for deploying Cisco IOS, ISDN, and Serial Video gateways:

- Consider the capacity needed on PSTN links for additional video calls.
- Consider the need of devices to use content sharing such as Binary Flow Control Protocol (BFCP), and the additional bandwidth that will be used on the IP network.
- Consider if users need features such as far-end camera control or DTMF that is used for conferences that the gateway needs to support.

Configuring the Gateways in Unified CM

You can configure a Cisco TelePresence ISDN Gateway in either of the following ways:

- Configure a SIP trunk pointing to the ISDN gateway (as shown in Figure 5-3), and add appropriate Unified CM route patterns pointing to the SIP trunk.
- Configure a SIP trunk from Unified CM to Cisco VCS. Have the ISDN gateway (or Serial gateway in this case) register to the VCS using H.323 (as shown in Figure 5-2).

The Cisco TelePresence Serial Gateway cannot be trunked directly to Unified CM. It must register to Cisco VCS, which in turn has a SIP trunk to Unified CM.

Either way, the goal is have all inbound calls received by the gateways sent to Unified CM so that Unified CM can decide how to route the calls. See the chapter on Cisco Unified CM Trunks, page 6-1, for more details on how to configure the SIP trunk between Unified CM and VCS.

Call Signaling Timers

Due to the delay inherent in H.320 bonding, video calls can take longer to complete than voice calls. Several timers in Unified CM are tuned, by default, to make voice calls process as fast as possible, and they can cause video calls to fail. Therefore, you must modify the following timers from their default values in order to support H.320 gateway calls:

- H.245TCSTimeout
- Media Exchange Interface Capability Timer
- Media Exchange Timer

Cisco recommends that you increase each of these timers to 25 by modifying them under the Service Parameters in Unified CM Administration. Note that these are cluster-wide service parameters, so they will affect calls to all types of devices, including voice calls to existing Cisco voice gateways.

Bearer Capabilities of Cisco IOS Voice Gateways

H.323 calls use the H.225/Q.931 Bearer Capabilities Information Element (bearer-caps) to indicate what type of call is being made. A voice-only call has its bearer-caps set to "speech" or "3.1 KHz Audio" while a video call has its bearer-caps set to "Unrestricted Digital Information." Some devices do not support Unrestricted Digital Information bearer-caps. Calls to these devices might fail if Unified CM attempts the call as a H.323 video call.

Unified CM decides which bearer-caps to set, based on the following factors:

- Whether the calling and/or called devices are video-capable
- Whether the region in Unified CM is configured to allow video for calls between those devices

Unified CM supports retrying the video call as audio, and this feature can be enabled through configuration. When Unified CM makes a video call with bearer-caps set to "Unrestricted Digital" and the call fails, Unified CM then retries the same call as an audio call with the bearer-caps set to "speech."

When using H.323, Cisco IOS gateways can service calls as voice or video, based on the bearer capabilities it receives in the call setup. When using SIP, the gateway translates the ISDN capabilities into SDP for call negotiations.

If the Cisco voice gateway uses MGCP to communicate with Unified CM, the problem will not occur because Unified CM does not support video on its MGCP protocol stack and because, in MGCP mode, Unified CM has complete control over the D-Channel signaling to the PSTN.

Gateways Best Practices

This section addresses the following best practices with regard to gateways:

Tuning Gateway Gain Settings

Connecting a Cisco Unified Communications network to the PSTN through gateways requires that you properly address media quality issues arising from echo and signal degradation due to power loss, impedance mismatches, delay, and so forth. For this purpose, you must establish a Network Transmission Loss Plan (NTLP), which provides a complete picture of signal loss in all expected voice paths. Using this plan, you can identify locations where signal strength must be adjusted for optimum loudness and effective echo cancellation. Note that not all carriers use the same loss plan, and that the presence of cellular networks adds further complexity in creating the NTLP. Cisco does not recommend adjusting input gain and output attenuation on gateways without first completing such an NTLP. For more information, refer to *Echo Analysis for Voice Over IP*, available at

http://www.cisco.com/en/US/docs/ios/solutions_docs/voip_solutions/EA_ISD.pdf

Routing Inbound Calls from the PSTN

Use one of the following methods to route inbound calls from the PSTN:

- Assign a single directory number to each user for both video and voice calls. This method is not recommended because all calls would have to be received from the PSTN on a video gateway, including audio-only calls. This would waste valuable video gateway resources and be hard to scale.

- Assign at least two different directory numbers to each video-enabled device in the Unified CM cluster, with one line for audio and another line for video. With this method, the outside (PSTN) caller must dial the correct number to enable video.

- For video calls, have outside callers dial the main number of the video gateway. Cisco ISDN and Serial gateways offer an integrated auto-attendant that prompts the caller to enter the extension number of the party they are trying to reach. Unified CM will then recognize that it is a video call when ringing the destination device. This method relieves the caller from having to remember two different DID numbers for each called party, but it adds an extra step to dialing an inbound video call.

> ✎
> **Note**   The outside video endpoints must support DTMF in order to enter the extension of the called party at the IVR prompt.

The following example illustrates the second method:

A user has a Cisco Unified IP Phone with video capabilities enabled. The extension of the IP Phone is 51212, and the fully qualified DID number is 1-408-555-1212. To reach the user from the PSTN for a voice-only call, people simply dial the DID number. The CO sends calls to that DID number through T1-PRI circuit(s) connected to a Cisco Voice Gateway. When the call is received by the gateway, Unified CM knows that the gateway is capable of audio only, so it negotiates only a single audio channel for that call. Conversely, for people to reach the user from the PSTN for a video call, they must dial the main number of the video gateway and then enter the user's extension. For example, they might dial 1-408-555-1000. The CO would send calls to that number through the T1-PRI circuit(s) connected to a Cisco ISDN video gateway. When the call is received by the gateway, an auto-attendant prompt asks the caller to enter the extension of the person they are trying to reach. When the caller enters the extension via DTMF tones, Unified CM knows that the gateway is capable of video, so it negotiates both audio and video channels for that call.

Gateway Digit Manipulation

The Cisco TelePresence ISDN Gateways 8321 and 3241 and the Cisco TelePresence Serial Gateways 8330 and 3340 all have capabilities for digit manipulation. It is possible to set up multiple dial plan rules on these video gateways. These rules match based on calling and/or called number and work in either the IP-to-PSTN or PSTN-to-IP direction. When an inbound call matches a configured dial plan rule, the ISDN or Serial gateway can take one of the following actions:

- Reject the call
- Enter the Auto Attendant
- Place a call to a number (or IP address, hostname, or URI in the case of a PSTN-to-IP call)

When the action is to place a call to a number, the original called number or parts of it can be used in the new number to call.

For more details, refer to the following documentation:

- Cisco TelePresence ISDN Gateway documentation, available at

  http://www.cisco.com/en/US/products/ps11448/tsd_products_support_series_home.html

- Cisco TelePresence Serial Gateway documentation, available at

  http://www.cisco.com/en/US/products/ps11605/tsd_products_support_series_home.html

Routing Outbound Calls to the PSTN

Use one of the following methods to route outbound calls to the PSTN:

- Assign different access codes (that is, different route patterns) for voice and video calls. For example, when the user dials 9 followed by the PSTN telephone number they are trying to reach, it could match a route pattern that directs the call out a voice gateway. Similarly, the digit 8 could be used for the route pattern that directs calls out a video gateway.

- Assign at least two different directory numbers on each video-enabled device in the Unified CM cluster, with one line for audio and another line for video. The two lines can then be given different calling search spaces. When users dial the access code (9, for example) on the first line, it could be directed out a voice gateway, while dialing the same access code on the second line could direct the call out a video gateway. This method alleviates the need for users to remember two different access codes but requires them to press the correct line on their phones when placing calls. However not all Cisco video endpoints support multiple lines at this time, in which case prefixes would be the preferred method for routing outbound calls to the PSTN.

Video Gateway Call Bandwidth

The Cisco TelePresence ISDN Gateway dial plan rules can be configured so that calls with a certain prefix are limited to a maximum amount of bandwidth on the ISDN connection to the PSTN. This is useful to ensure that a single call cannot monopolize the entire PRI link. When you configure a service prefix in the gateway, you can choose one of the following maximum speeds:

- 128 kbps
- 192 kbps
- 256 kbps
- 320 kbps
- 384 kbps
- 512 kbps
- 768 kbps
- 1152 kbps
- 1472 kbps

Calls from an IP endpoint toward the PSTN can include the service prefix at the beginning of the called number in order for the gateway to decide which service to use for the call. Optionally, you can configure the default prefix to be used for calls that do not include a service prefix at the beginning of the number. This method can become quite complex because users will have to remember which prefix to dial for the speed of the call they wish to make, and you would have to configure multiple route patterns in Unified CM (one for each speed).

Note   Two global settings on the Cisco TelePresence ISDN Gateway can be used to set a minimum or maximum bandwidth value for incoming and outgoing ISDN calls. The dial plan cannot override this value with a higher maximum bandwidth; however, a dial plan can impose a lower bandwidth for particular calls.

Automated Alternate Routing (AAR)

When the IP network does not have enough bandwidth available to process a call, Unified CM uses its call admission control mechanism to determine what to do with the call. Unified CM performs one of the following actions with the call, depending on how you have configured it:

- Fail the call, playing busy tone to the caller and displaying a Bandwidth Unavailable message on the caller's screen

- Retry a video call as an audio-only call

- Use automated alternate routing (AAR) to re-route the call over an alternative path, such as a PSTN gateway

The Retry Video Call as Audio option takes effect only on the terminating (called) device, thus allowing the flexibility for the calling device to have different options (retry or AAR) for different destinations.

If a video call fails due to bandwidth limitations but automated alternate routing (AAR) is enabled, Unified CM will attempt to reroute the failed call as a video call to the AAR destination. If AAR is not enabled, the failed call will result in a busy tone and an error message will be sent to the caller. (See Figure 5-7.)

Figure 5-7   Possible Scenarios for a Video Call
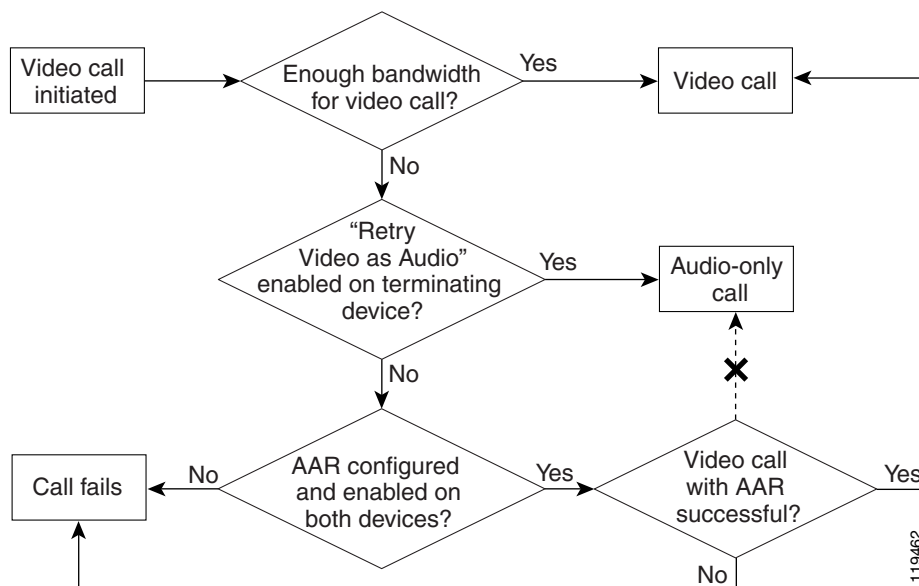
To provide AAR for voice or video calls, you must configure the calling and called devices as members of an AAR group and configure an External Phone Number Mask for the called device. The External Phone Number Mask designates the fully qualified E.164 address for the user's extension, and the AAR group indicates what digits should be prepended to the External Phone Number Mask of the called device in order for the call to route successfully over the PSTN.

For example, assume that user A is in the San Jose AAR group and user B is in the San Francisco AAR group. User B's extension is 51212, and the External Phone Number Mask is 6505551212. The AAR groups are configured to prepend 91 for calls between the San Jose and San Francisco AAR groups. Thus, if user A dials 51212 and there is not enough bandwidth available to process the call over the IP WAN between those two sites, Unified CM will take user B's External Phone Number Mask of 6505551212, prepend 91 to it, and generate a new call to 916505551212 using the AAR calling search space for user A.

By default, all video-capable devices in Unified CM have the Retry Video Call as Audio option enabled (checked). Therefore, to provide AAR for video calls, you must disable (uncheck) the Retry Video Call as Audio option. Additionally, if a call admission control policy based on Resource Reservation Protocol (RSVP) is being used between locations, the RSVP policy must be set to Mandatory for both the audio and video streams.

Furthermore, Unified CM looks at only the called device to determine whether the Retry Video Call as Audio option is enabled or disabled. So in the scenario above, user B's phone would have to have the Retry Video Call as Audio option disabled in order for the AAR process to take place.

Finally, devices can belong to only one AAR group. Because the AAR groups determine which digits to prepend, AAR groups also influence which gateway will be used for the rerouted call. Depending on your choice of configuration for outbound call routing to the PSTN, as discussed in the previous section, video calls that are rerouted by AAR might go out a voice gateway instead of a video gateway. Therefore, carefully construct the AAR groups and the AAR calling search spaces to ensure that the correct digits are prepended and that the correct calling search space is used for AAR calls.

While these considerations can make AAR quite complex to configure in a large enterprise environment, AAR is easier to implement when the endpoints are strictly of one type or the other. When endpoints are capable of both audio and video calls (such as Cisco Unified IP Phone 9971 or a Cisco TelePresence System EX90), the configuration of AAR can quickly become unwieldy. Therefore, Cisco recommends that large enterprise customers who have a mixture of voice and video endpoints give careful thought to the importance of AAR for each user, and use AAR only for select video devices such as dedicated videoconference rooms or executive video systems. Table 5-2 lists scenarios when it is appropriate to use AAR with various device types.

Table 5-2   *When to Use AAR with a Particular Device Type*

Device Type	Device is used to call:	Enable AAR?	Comments
IP Phone	Other IP Phones and video-capable devices	Yes	Even when calling a video-capable device, the source device is capable of audio-only, thus AAR can be configured to route calls out a voice gateway.
Cisco Jabber or Cisco Unified IP Phone 9971	Other video-capable devices only	Yes	Because the device is used strictly for video calls, you can configure the AAR groups accordingly.
	IP Phones and other video-capable devices	No	It will be difficult to configure the AAR groups to route audio-only calls differently than video calls.

Table 5-2   When to Use AAR with a Particular Device Type (continued)

Device Type	Device is used to call:	Enable AAR?	Comments
H.323 or SIP client	Other video-capable devices only	Yes	Because the device is used strictly for video calls, you can configure the AAR groups accordingly.
	IP Phones and other video-capable devices	No	It will be difficult to configure the AAR groups to route audio-only calls differently than video calls

Least-Cost Routing

Least-cost routing (LCR) and tail-end hop-off (TEHO) are very popular in VoIP networks and can be used successfully for video calls as well. In general, both terms refer to a way of configuring the call routing rules so that calls to a long-distance number are routed over the IP network to the gateway closest to the destination, in an effort to reduce toll charges. Unified CM supports this feature through its rich set of digit analysis and digit manipulation capabilities, including:

- Partitions and calling search spaces
- Translation patterns
- Route patterns and route filters
- Route lists and route groups

Configuring LCR for video calls is somewhat more complicated than for voice calls, for the following reasons:

- Video calls require their own dedicated gateways, as discussed previously in this chapter
- Video calls require much more bandwidth than voice calls

With respect to dedicated gateways, the logic behind why you might or might not decide to use LCR for video calls is very similar to that explained in the section on Automated Alternate Routing (AAR), page 5-18. Due to the need to have different types of gateways for voice and video, it can become quite complex to configure all the necessary partitions, calling search spaces, translation patterns, route patterns, route filters, route lists, and route groups needed for LCR to route voice calls out one gateway and video calls out another.

With respect to bandwidth requirements, the decision to use LCR depends on whether or not you have enough available bandwidth on your IP network to support LCR for video calls to/from a given location. If the current bandwidth is not sufficient, then you have to determine whether the benefits of video calls are worth the cost of either upgrading your IP network to make room for video calls or deploying local gateways and routing calls over the PSTN. For example, suppose you have a central site with a branch office connected to it via a 1.544-Mbps T1 circuit. The branch office has twenty video-enabled users in it. A 1.544-Mbps T1 circuit can handle at most about four 384-kbps video calls. Would it really make sense in this case to route video calls up to the central site in order to save on toll charges? Depending on the number of calls you want to support, you might have to upgrade your 1.544-Mbps T1 circuit to something faster. Is video an important enough application to justify the additional monthly charges for this upgrade? If not, it might make more sense to deploy a Cisco video gateway at the branch office and not bother with LCR. However, placing local Cisco video gateways at each branch office is not inexpensive either, so ultimately you must decide how important video-to-PSTN calls are to your business. If video is not critical, perhaps it is not worth upgrading the bandwidth or buying video

gateways but, instead, using the Retry Video Call as Audio feature to reroute video calls as voice-only calls if they exceed the available bandwidth. Once a call is downgraded to voice-only, local gateway resources and bandwidth to perform LCR become more affordable and easier to configure.

Fax and Modem Support

For information on fax and modem support across Cisco gateways refer to the following documentation:

- The *Gateways* chapter of the *Cisco Unified Communications System 9.0 SRND*, available at

  http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/9x/gateways.html

- *Fax, Modem, and Text Support over IP Configuration Guide*, available at

  http://www.cisco.com/en/US/docs/ios-xml/ios/voice/fax/configuration/15-mt/vf-15-mt-book.html

Fax and Modem Support