CHAPTER 2

Connecting the Endpoints

Overview

As discussed in Chapter 1, "Cisco TelePresence Solution Overview," there are many elements to Cisco TelePresence endpoint systems, including:

- TelePresence codecs (primary and secondary)
- Cisco Unified 7975G IP phone
- 65" plasma displays
- Cameras
- Microphones
- Speakers
- Auxiliary audio devices
- Auxiliary video devices

There are other elements, such as mounting brackets, furniture, cables, and power cords; the full assembly and connectivity instructions are covered in detail in the documentation.

The focus of this chapter is to provide an overview of how these main system elements are interconnected within CTS-1000, CTS-500, CTS-3000, and CTS-3200 systems, as well as how these interact with the network infrastructure. Such an overview helps lay a foundational context for the design chapters that follow.

Connecting a CTS-1000 System

The CTS-1000 includes:

- One Cisco TelePresence codec (a primary codec)
- One Cisco Unified 7975G IP phone
- One 65" plasma display
- One high-definition camera
- One microphone
- One speaker
- One input for auxiliary audio

- One input for auxiliary video which can be used for a document camera or PC

The Cisco TelePresence primary codec is the center of the CTS systems. Essentially, all components connect to it and it, in turn, connects to the network infrastructure.

Specifically, the Cisco Unified 7975G IP phone connects to the TelePresence primary codec via an RJ-45 cable that provides it network connectivity and 802.3af Power-over-Ethernet (PoE).

Another RJ-45 cable connects from the TelePresence primary codec to the camera, providing the camera with 802.3af PoE. A second cable from the primary codec to the camera provides video connectivity.

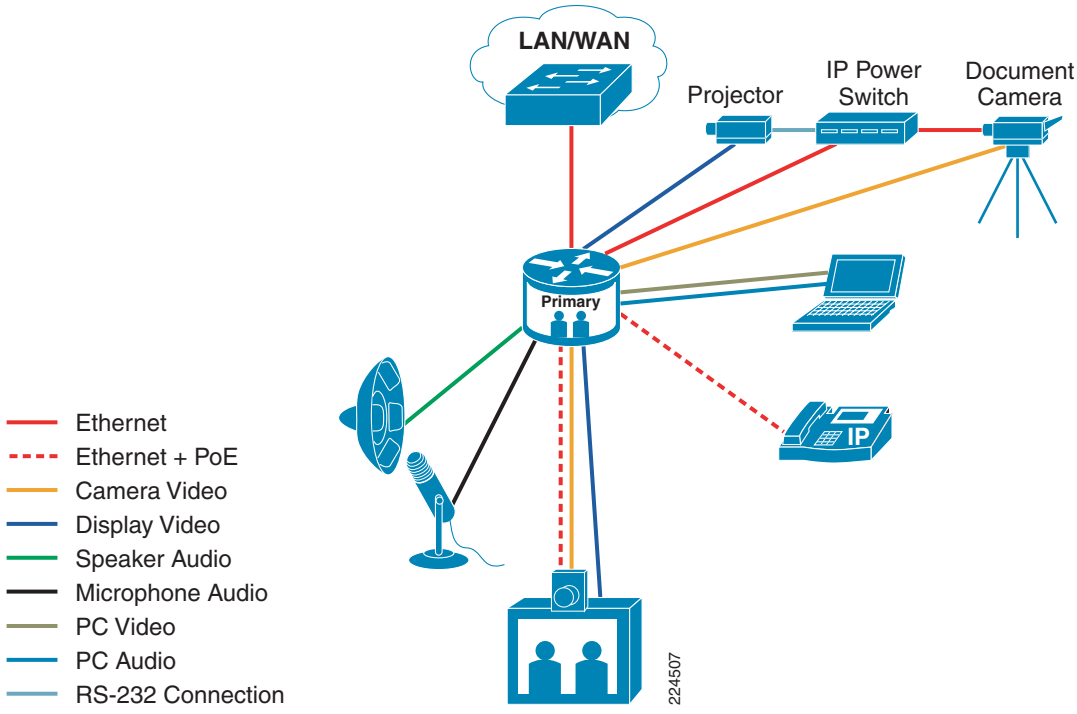
A video cable also connects the primary codec to the 65” plasma display. This cable is essentially an High Definition Multimedia Interface (HDMI) cable, but with a proprietary element for carrying management information instead of audio signals (as the audio signals are processed independently by the master codec).

Additionally, a speaker cable and a microphone cable connect the speaker and microphone to the primary codec, respectively.

The primary codec also has inputs for auxiliary audio and auxiliary video. Auxiliary video can come from a PC connection or from a document camera connection. An IP power switch (IPS) provides control for the on/off function of the document camera, attached projector, as well as the lighting shroud of the CTS unit via an Ethernet connection.

Finally, an RJ-45 cable provides 10/100/1000 Ethernet connectivity from the primary codec to the network infrastructure. These interconnections for a CTS-1000 system are illustrated in Figure 2-1.

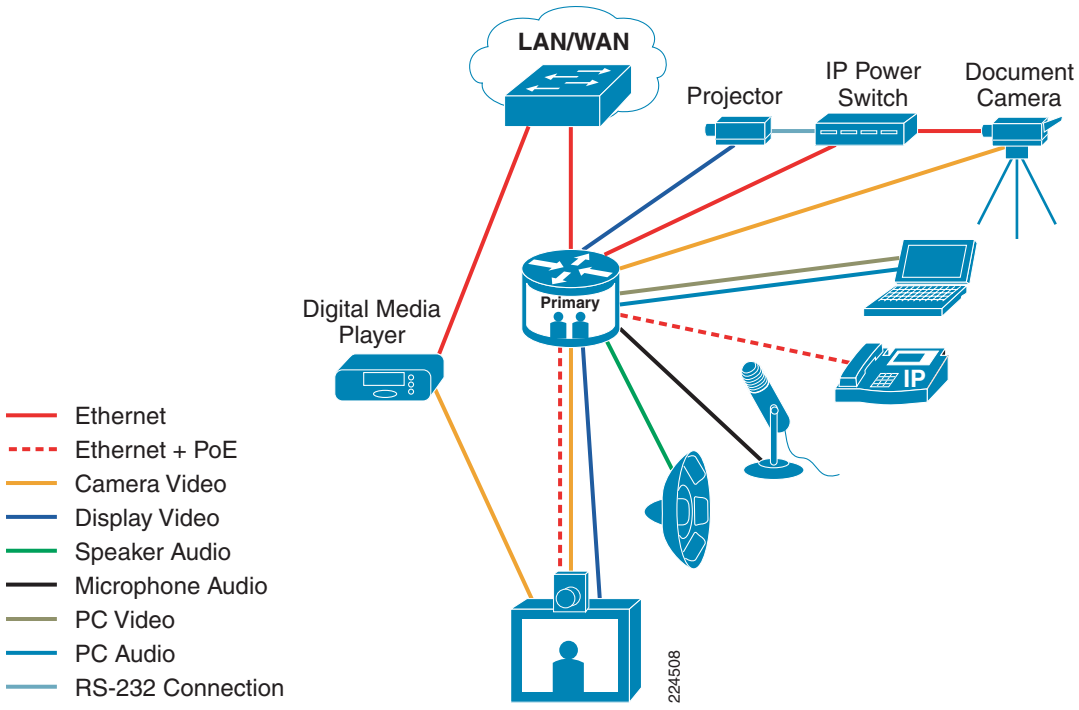
Figure 2-1 Connectivity Schematic for a CTS-1000 System



Connecting a CTS-500 System

The CTS-500 is a enclosed system that includes a 37” display with integrated codec, camera, microphone, and speaker. Its only connection point is an RJ-45 cable which connects it to the network. For purposes of this discussion, the CTS-500 is similar to the CTS-1000 in terms of connectivity, but has a connection for an optional digital media player. This can be used to display content when the CTS-500 is not being used for TelePresence Meetings. The connectivity of the CTS-500 is shown in Figure 2-2.

Figure 2-2 Connectivity Schematic for a CTS-500 System



Connecting a CTS-3000 System

The CTS-3000 system includes:

- One Cisco TelePresence primary codec
- Two Cisco TelePresence secondary codecs
- One Cisco Unified 7975G IP phone
- Three 65” plasma displays
- Three high-definition cameras
- Three microphones
- Three speakers
- One input for auxiliary audio
- One input for auxiliary video which can be used for a document camera or PC

As with the CTS-1000 system, the primary codec is the central part of the CTS-3000 system to which all other components interconnect.

Specifically, the Cisco Unified 7975G IP phone connects to the TelePresence primary codec via an RJ-45 cable that provides it network connectivity and 802.3af Power-over-Ethernet (PoE).

A video cable connects the primary codec to the center 65” plasma display; another of these cables connects the right display to the (right) secondary codec, and a third connects the left display to the (left) secondary codec. As with the CTS-1000 system, this cable is essentially an HDMI cable, but with a proprietary element for carrying management information instead of audio signals (as the audio signals are processed independently by the master codec). Each of these secondary codecs, in turn, are connected to the primary codec via a RJ-45 cable; however, no 802.3af PoE is required over these Ethernet links as the secondary codecs have independent power supplies.

Three cameras are mounted on the central display and each camera is connected to its respective codec:

- The left camera is connected to the (left) secondary codec.
- The center camera is connected to the primary codec.
- The right camera is connected to the (right) secondary codec.

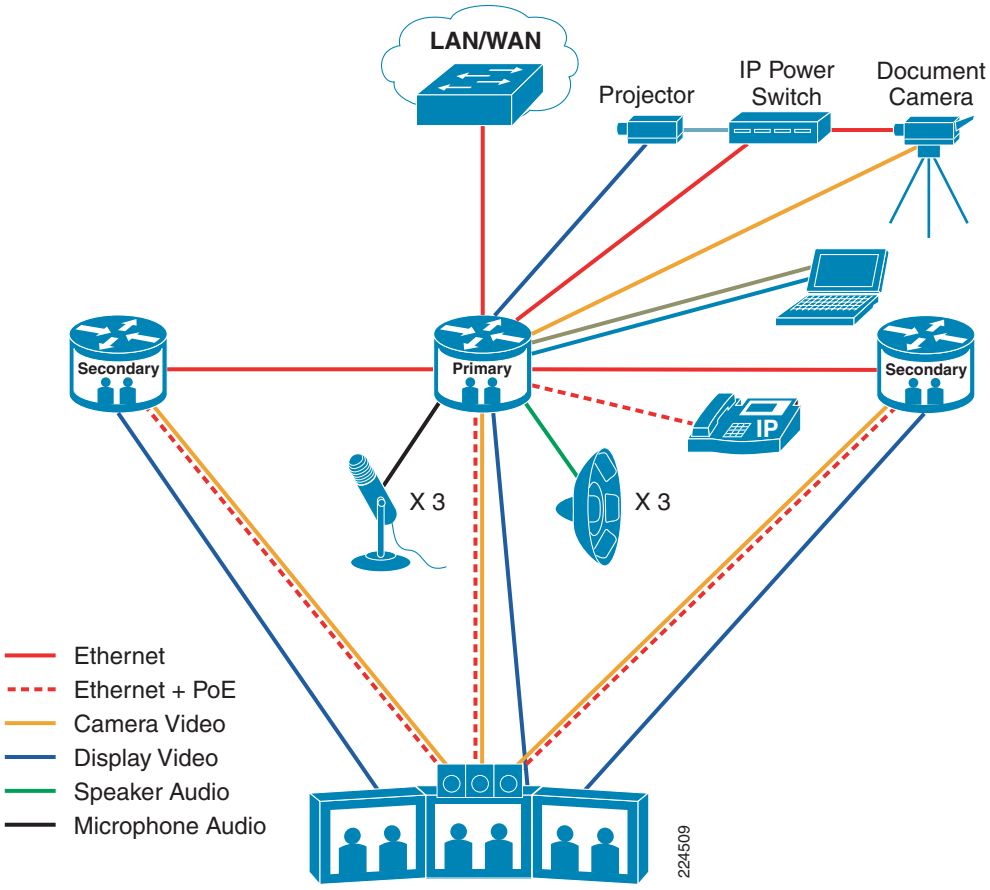
Each camera connects to its respective codec via two cables: a RJ-45 cable, which provides 802.3af PoE and network connectivity to the camera and a video cable to carry the video signals to the codec.

Additionally, three speaker cables and three microphone cables connect the (left, center, and right) speakers and (left, center, and right) microphones to the primary codec, respectively.

The primary codec also has inputs for auxiliary audio and auxiliary video. Auxiliary video can come from a PC connection or from a document camera connection. An IP power switch (IPS) provides control for the on/off function of the document camera, attached projector, as well as the lighting shroud of the CTS unit via an Ethernet connection.

Finally, an RJ-45 cable provides 10/100/1000 Ethernet connectivity from the primary codec to the network infrastructure. These interconnections for a CTS-3000 system are illustrated in Figure 2-3.

Figure 2-3 Connectivity Schematic for a CTS-3000 System



Connecting a CTS-3200 System

The CTS-3000 system includes:

- One Cisco TelePresence primary codec
- Two Cisco TelePresence secondary codecs
- One Cisco Unified 7975G IP phone
- Three 65” plasma displays
- Three high-definition cameras
- Nine microphones
- Three speakers
- One input for auxiliary audio
- One input for auxiliary video or another optional secondary codec for high-speed auxiliary video which can be used for a document camera or PC

As with the CTS-3000 system, the primary codec is the central part of the CTS-3200 system to which all other components interconnect.

Specifically, the Cisco Unified 7975G IP phone connects to the TelePresence primary codec via an RJ-45 cable that provides it with network connectivity and 802.3af Power-over-Ethernet (PoE).

A video cable connects the primary codec to the center 65” plasma display, another cable connects the right display to the (right) secondary codec, and a third connects the left display to the (left) secondary codec. As with the CTS-3000 system, this cable is essentially an HDMI cable, but with a proprietary element for carrying management information instead of audio signals (as the audio signals are processed independently by the master codec). Each of these secondary codecs, in turn, are connected to the primary codec via an RJ-45 cable; however, no 802.3af PoE is required over these Ethernet links as the secondary codecs have independent power supplies.

Three cameras are mounted on the central display and each camera is connected to its respective codec:

- The left camera is connected to the (left) secondary codec.
- The center camera is connected to the primary codec.
- The right camera is connected to the (right) secondary codec.

Each camera connects to its respective codec via two cables:

- RJ-45 cable, which provides 802.3af PoE and network connectivity to the camera
- Video cable to carry the video signals to the codec

Additionally, three speaker cables connect the (left, center, and right) speakers to the primary codec, respectively.

One microphone cable connects the center microphone to the primary codec. The remaining eight microphones are connected to the audio extension box, which is in turn connected to the primary codec. The audio extension box also houses the HDMI splitter. The HDMI splitter connects to the auxiliary video output of the primary codec. Up to four displays or a projector and three displays can be connected to the HDMI ports on the audio extension box.

The primary codec also has inputs for low-speed (5 frames per second) auxiliary audio and auxiliary video inputs. Video input can come from a PC or optional document camera. An IP power switch (IPS) provides control for the on/off function of the document camera, attached projector, as well as the lighting shroud of the CTS unit via an Ethernet connection. Optionally, another secondary codec can be connected to the primary codec to provide high-speed (30 frames per second) auxiliary video input. The auxiliary codec is connected to the primary codec via the RJ-45 cable from the Ethernet port normally used for the document camera. Auxiliary audio is still connected to the primary codec.

A connection for an optional headset is also provided on the primary codec of the CTS-3200.

Finally, an RJ-45 cable provides 10/100/1000 Ethernet connectivity from the primary codec to the network infrastructure. These interconnections for a CTS-3200 system are illustrated in Figure 2-4 and Figure 2-5.

Figure 2-4 Connectivity Schematic for a CTS-3200 System with Low-Speed Auxiliary Input

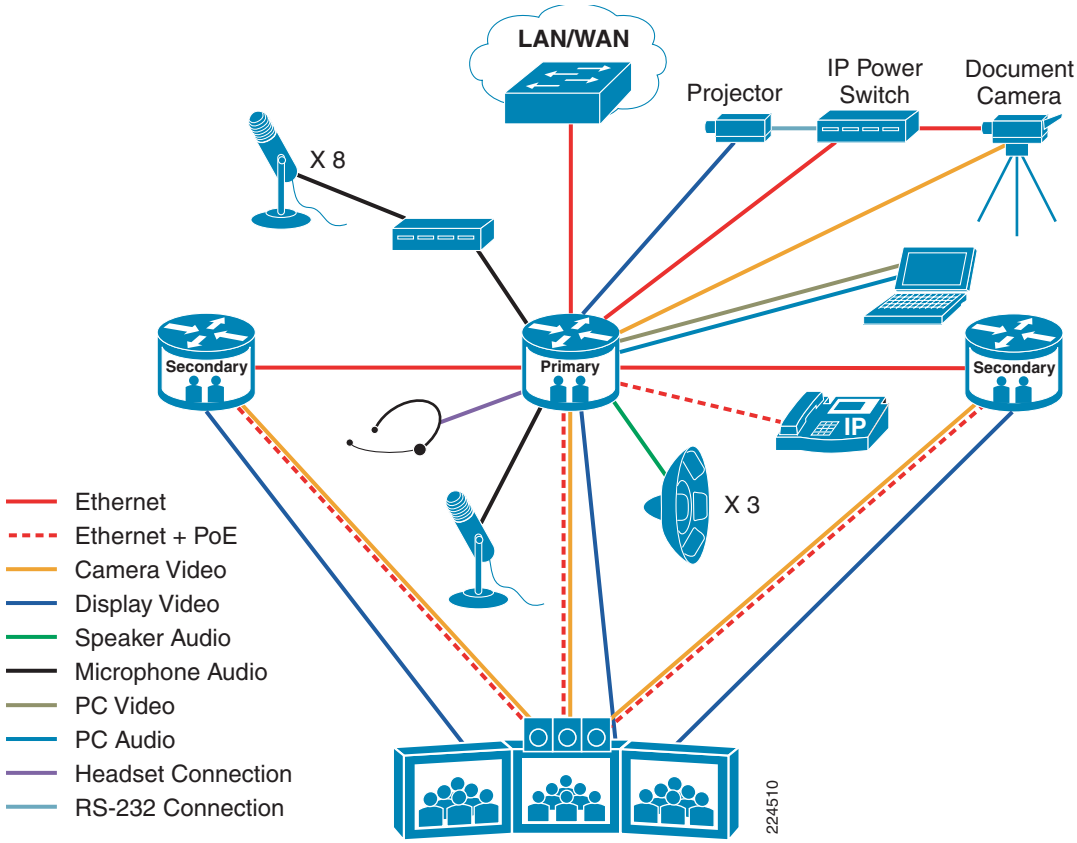
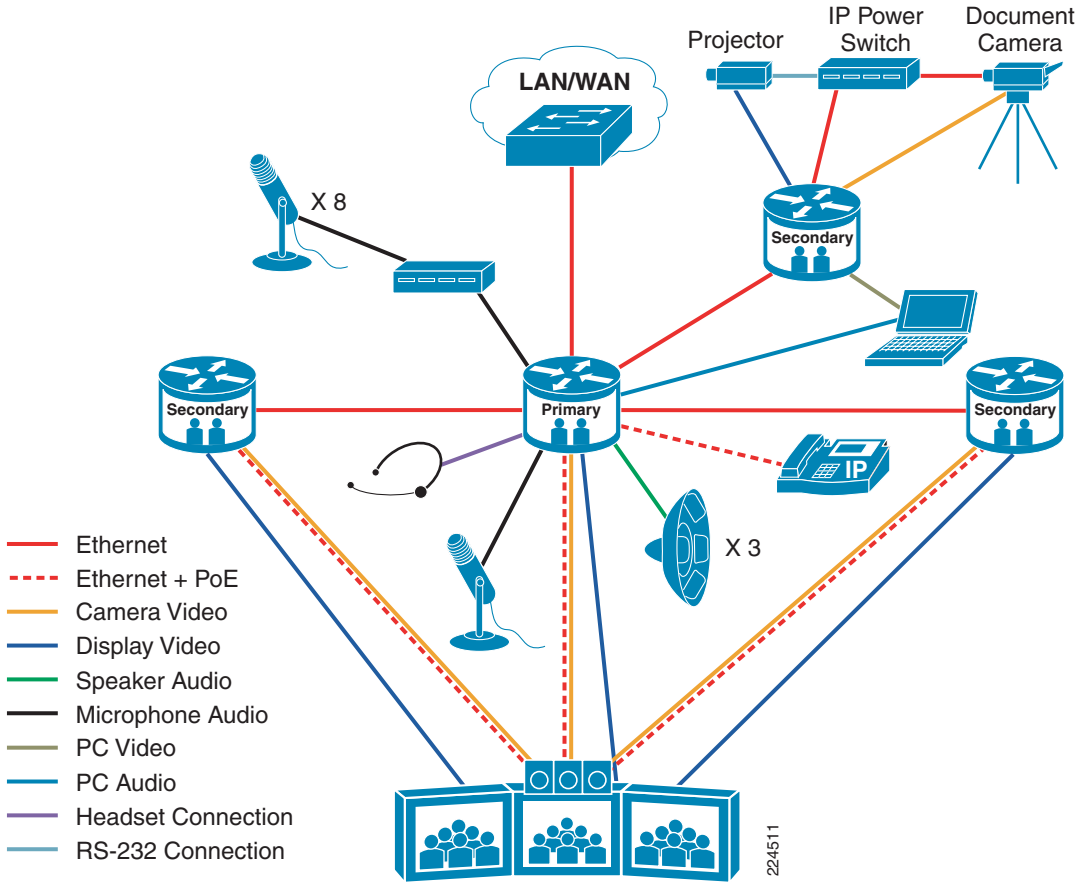


Figure 2-5 Connectivity Schematic for a CTS-3200 System with High-Speed Auxiliary Input

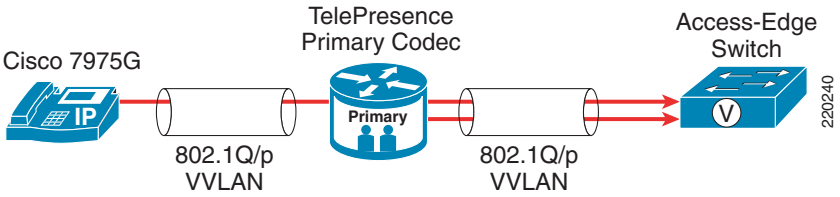


Cisco TelePresence Network Interaction

The primary codec is the interface between the CTS endpoint system and the network infrastructure. The primary codec connects to the network access edge switch via a RJ-45 10/100/1000 port. The access edge Catalyst switch that it connects to provides IP services, 802.1Q/p VLAN services, QoS services, and security services to the TelePresence endpoint.

Additionally, the primary codec provides a RJ-45 connection to the Cisco Unified 7975G IP phone, to which it supplies 802.3af PoE. When the IP phone boots up, it sends a Cisco Discovery Protocol (CDP) message to the primary codec. The codec receives this CDP message and passes it on to the access edge switch, supplementing it with its own CDP advertisement. The access edge switch and codec exchange CDP messages and the switch (if configured according to best practice recommendations for IP telephony deployments) places the primary codec and the 7975G IP phone in a 802.1Q Voice VLAN (VVLAN), wherein 802.1Q/p Class of Service (CoS) markings are trusted. The primary Codec passes 802.1Q tags between the 7975G IP phone and the network access edge switch, extending the VVLAN all the way to the IP phone. This 802.1Q/p VVLAN assignment is illustrated in Figure 2-6.

Figure 2-6 Voice VLAN Extension Through Cisco TelePresence Primary Codec



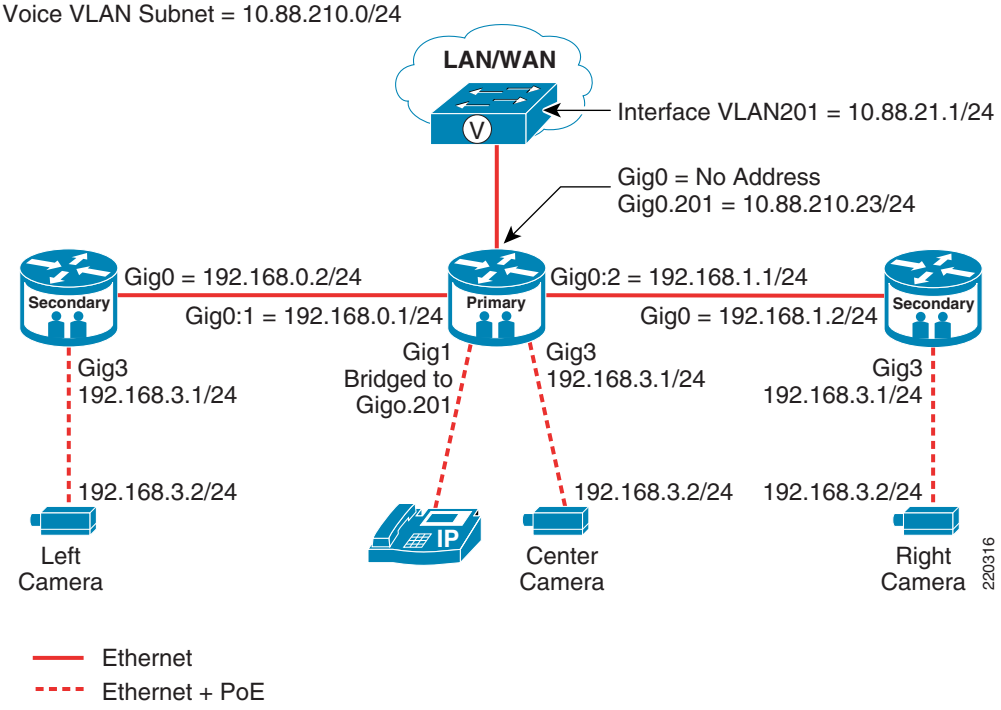
Note The above network interaction assumes that CDP is enabled and Voice VLANs are configured. If this is not the case, then then the network interaction begins with the DHCP requests described next.

If configured for dynamic IP addressing, the 7975G IP phone and the primary codec each generate a Dynamic Host Configuration Protocol (DHCP) request to the network and are supplied with IP addresses (one for the IP phone and another for the primary codec). The DHCP server may also provide the IP phone and primary codec the IP address of the download server, via DHCP option 150, from which they download their configuration files and firmware loads. This function is often provided by the Cisco Unified Communications Manager (CUCM) server. Alternatively, either or both of the devices may be configured with a static IP address and TFTP server address.

Additionally, it is important to note that the TelePresence systems utilize a private network for internal communications between the primary and secondary codecs, as well as between codecs and cameras. By default the internal address range used is 192.168.0.0/24 through 192.168.4.0/24; however, if the TelePresence codec receives a 192.168.x.x address from the network, then the internal private network will switch to 10.0.0.0/24 through 10.0.4.0/24. A default internal network IP address assignment is illustrated in Figure 2-7.

Figure 2-7 Default TelePresence Internal IP Addressing Scheme

Example:
Voice VLAN ID = 201
Voice VLAN Subnet = 10.88.210.0/24



Note Even though only 192.168.0.0/24 through 192.168.3.0/24 are illustrated in Figure 2-7, 192.168.4.0/24 is reserved within the system for future (internal) use.

Similarly, if the TelePresence system is using 10.0.0.0/24 through 10.0.3.0/24 for its internal networking address range, then 10.0.4.0/24 is reserved within the system for future (internal) use.

It is important to note three key points regarding the internal networking of TelePresence systems:

- From the network's perspective, the TelePresence primary codec appears as a single endpoint device with a single IP address (but remember, the 7975G IP Phone also appears as a separate endpoint device with its own IP address).
- The internal components (such as secondary codecs and cameras) do not receive a default gateway. Therefore, they cannot route beyond the primary codec.
- If the primary codec is using 192.168.0.0/24 through 192.168.4.0/24 as its internal networking addresses (which is the default), then it is not able to connect to external servers or endpoints that are using these same addresses (as it will attempt to reach such addresses via its internal network, not its external default gateway). Conversely, if the primary codec has been assigned an IP address from the network in the 192.168.x.x range, then it uses internal networking addresses in the range of 10.0.0.0/24 through 10.0.4.0/24, and similarly, is not able to connect to external servers or endpoints that may be using these same addresses. Table 2-1 summarizes the IP addressing best practices for networks supporting TelePresence.

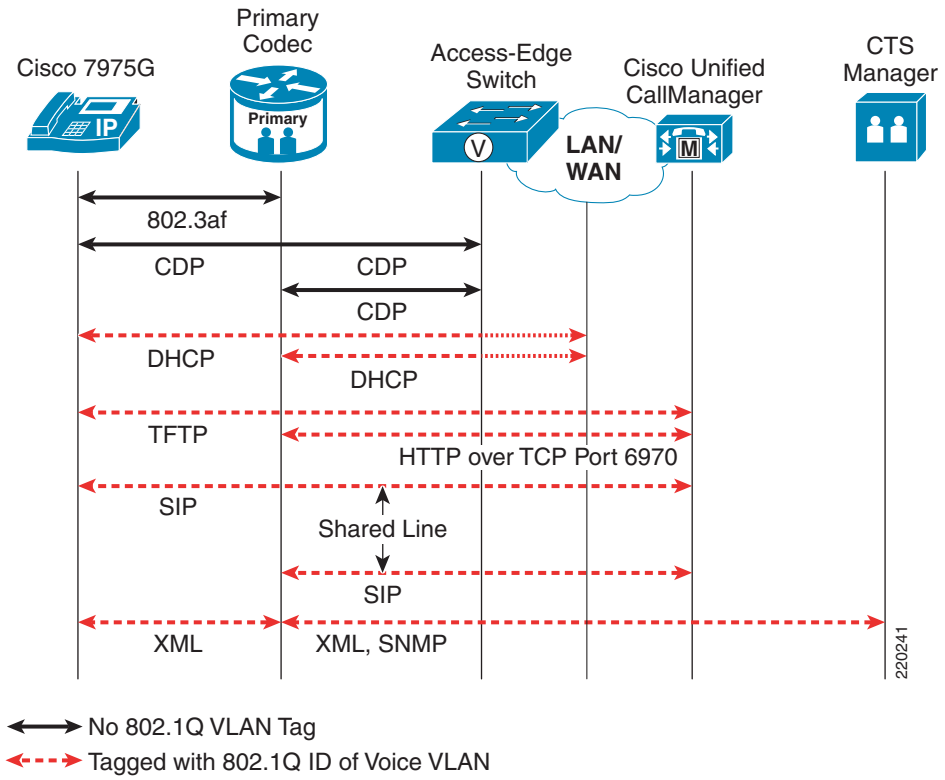
Table 2-1 TelePresence Network IP Addressing Best Practices

For Environments Where the CTS Uses 192.168.x.x for its Internal Communications. Avoid Using the Following Subnets:	For Environments Where the CTS Uses 10.x.x.x for its Internal Communications. Avoid Using the Following Subnets:
192.168.0.0/24	10.0.0.0/24
192.168.1.0.24	10.0.1.0/24
192.168.2.0.24	10.0.2.0/24
192.168.3.0.24	10.0.3.0/24
192.168.4.0.24	10.0.4.0/24

Provided there are no IP addressing issues, as described above, the IP phone then initiates a Trivial File Transfer Protocol (TFTP) session with the Cisco Unified Communications Manager (CUCM) to download its configuration and firmware files. The primary codec initiates an HTTP session over TCP port 6970 for its configuration and firmware files. Note that DNS may also be required to translate the CUCM hostname to an IP address.

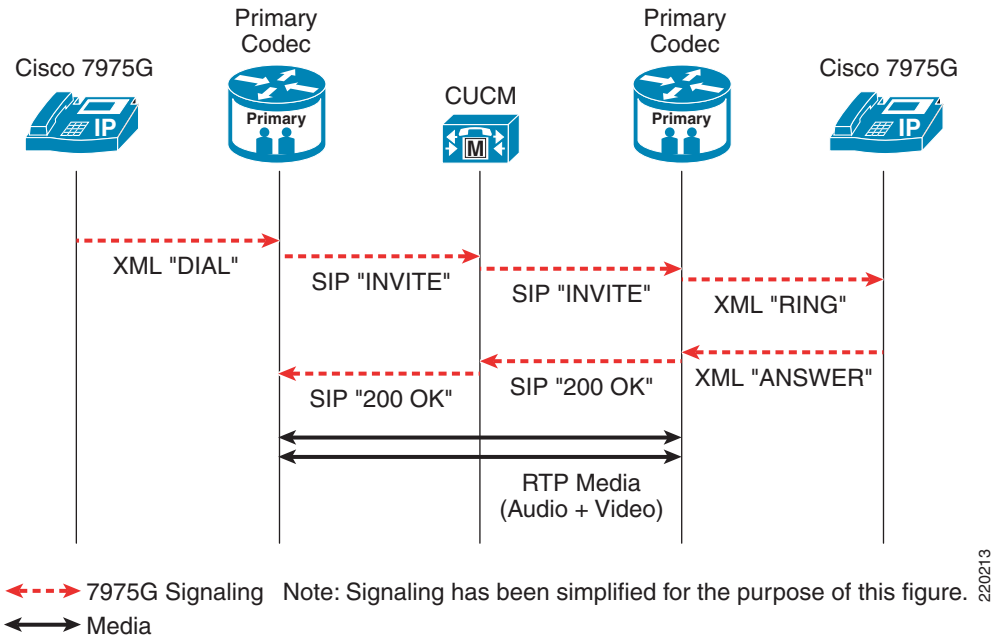
The primary codec then communicates with CUCM via Session Initiation Protocol (SIP). The Cisco 7975G IP Phone also communicates with CUCM via SIP, identifying itself as a shared line with the primary codec. Additional messaging occurs between the 7975G IP phone, the TelePresence primary codec, and the Cisco TelePresence Manager via Extensible Markup Language (XML), as well as Simple Network Management Protocol (SNMP). These network protocol interactions are illustrated in Figure 2-8.

Figure 2-8 Cisco TelePresence Network Control, Management, and Signaling Protocols



Once the TelePresence system has completed these protocol interactions, it is ready to place and receive calls. When a call is initiated, the Cisco 7975G IP phone sends an XML Dial message to its primary codec, which forwards the request as a SIP Invite message to the Cisco Unified Communications Manager. CUCM, in turn, forwards the SIP Invite message to the destination TelePresence codec, which forwards the message as an XML Ring message to its 7975G associated IP phone. The TelePresence primary codec can be set to automatically answer the incoming call or can be set to send an incoming call alert to the 7975G IP phone. If set to auto-answer, the codec answers the call immediately and sends a SIP OK message to CUCM. If auto-answer is not enabled, when the user presses the Answer softkey on the 7975G IP phone, the 7975G IP phone replies with a XML Answer message to the receiving TelePresence primary codec, and the codec in turn sends a SIP 200 OK message to CUCM. CUCM relays this SIP 200 OK message to the originating TelePresence primary codec and the call is established. Real-time media, both audio and video, is then passed between the TelePresence primary codecs over Real Time Protocol (RTP). The signaling and media paths for Cisco TelePresence are illustrated in Figure 2-9.

Figure 2-9 Cisco TelePresence Signaling and Media Paths



CTS-1000 and CTS-500 systems send only one audio and one video stream (excluding auxiliary audio and video inputs for the moment). On the other hand, CTS-3000 and CTS-3200 primary codecs process three separate audio and three separate video streams. However, these codecs do not send three separate audio streams and three separate video streams over the network. Rather, CTS-3000 and CTS-3200 primary codecs multiplex the three audio streams into one RTP stream and three video streams into one RTP stream, and hence send only a single audio and a single video stream over the network. These streams, in turn, are de-multiplexed by the receiving codec. The multiplexing of audio and video streams performed by the CTS-3000 primary codecs is illustrated in Figure 2-10. Auxiliary audio and video inputs are also multiplexed into the same audio and video streams. Therefore, in the case of the CTS-1000 or CTS-500, the primary video and auxiliary video are multiplexed into one outgoing video stream; likewise the primary audio and auxiliary audio are multiplexed into one outgoing audio stream. In the case of the CTS-3000 or CTS-3200, the auxiliary video is treated as the 4th video channel and multiplexed in with the rest of the video; likewise the auxiliary audio is treated as the 4th audio channel and multiplexed in with the rest of the audio.

Figure 2-10 CTS-3000 Multiplexing of Audio and Video Streams