CHAPTER 7

Call Processing Overview

Overview

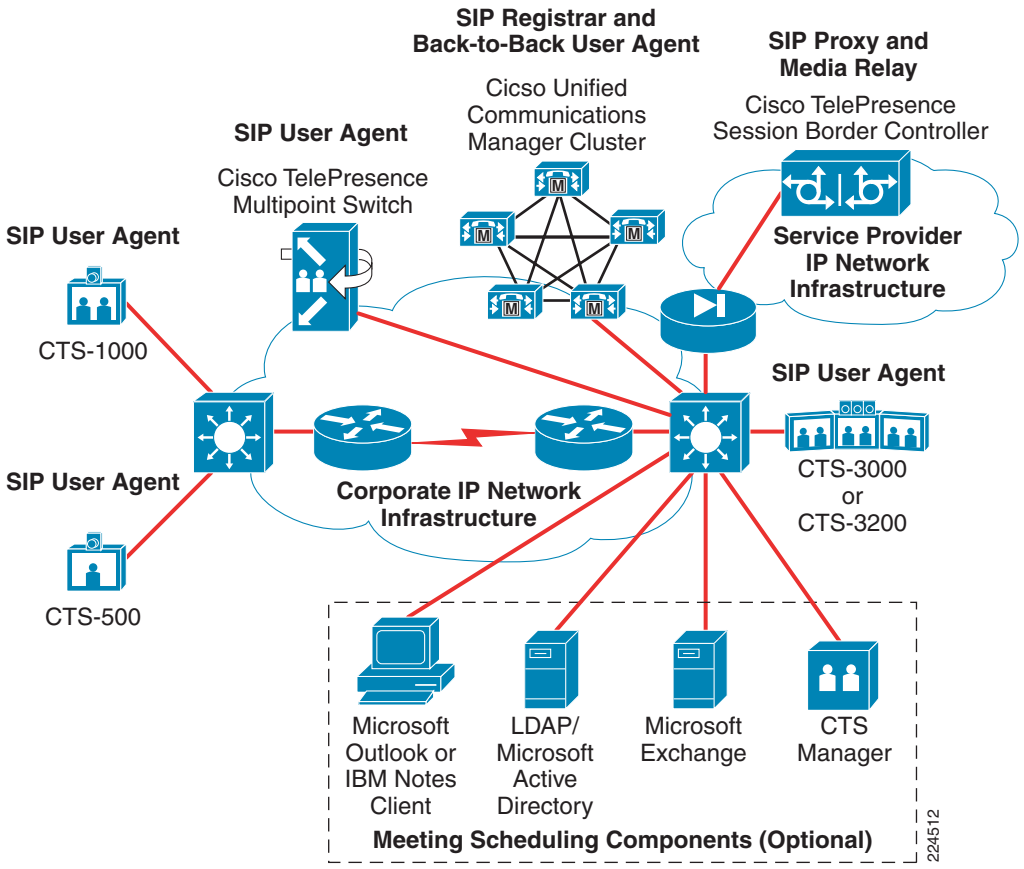
This chapter discusses the Session Initiation Protocol (SIP) and call processing design for Cisco TelePresence, including:

- How the Cisco TelePresence suite of virtual meeting solutions integrates with Cisco Unified Communications Manager (CUCM)
- CUCM software version requirements
- Current CUCM cluster design recommendations
- How the Cisco TelePresence Codecs use Session Initiation Protocol (SIP) and how they register using a shared line appearance with the Cisco Unified 7975G IP phone
- How Cisco TelePresence multipoint resources, such as the Cisco TelePresence Multipoint Switch (CTMS), are configured as a SIP trunk to CUCM and how multipoint calls are routed

Call Processing Components

Figure 7-1 shows the components involved in point-to-point and multipoint TelePresence meetings.

Figure 7-1 Cisco TelePresence Solution Components



These components consist of:

- Two or more Cisco TelePresence systems (any combination of CTS-300s, CTS-3200s, CTS-1000s,or CTS-500s), each with a Cisco Unified 7975G IP phone (not shown in Figure 7-1) which functions as the user interface for launching, controlling, and concluding the meeting
- One CUCM Cluster

  TelePresence requires CUCM version 5.1.1 or higher, with version 5.1.2 recommended for support of the Auto Collaborate feature.
- One or more Cisco TelePresence Multipoint Switches (required for multipoint TelePresence meetings)
- IP network infrastructure over which the signaling, video, and audio media are transported
- A Cisco TelePresence Session Border Controller (SBC) typically used for inter-Enterprise TelePresence calls
- Meeting scheduling components (optional):
  - Microsoft Exchange 2003 server
  - Microsoft Active Directory 2000 or 2003 server
  - Microsoft Outlook client
  - Cisco TelePresence Manager (CTS-MAN)

  These components are only required for scheduled TelePresence meetings. Ad hoc and permanent TelePresence meetings do not require them.

TelePresence Endpoint Interface to CUCM (Line-Side SIP)

CUCM is the core call processing software for the Cisco TelePresence solution as well as all other Cisco IP telephony devices. CUCM functions as both a SIP registrar and Back to Back User Agent (B2BUA). TelePresence Codecs and 7975G IP phones use SIP for call signaling and control, functioning as SIP user agents which register with a CUCM cluster. Cisco TelePresence Systems use TCP for their SIP signaling to/from CUCM. It should be noted that TelePresence devices are currently not supported by the Survivable Remote Site Telephony (SRST) feature of Cisco router platforms, which is often used to provide resiliency in CUCM deployments with remote sites.

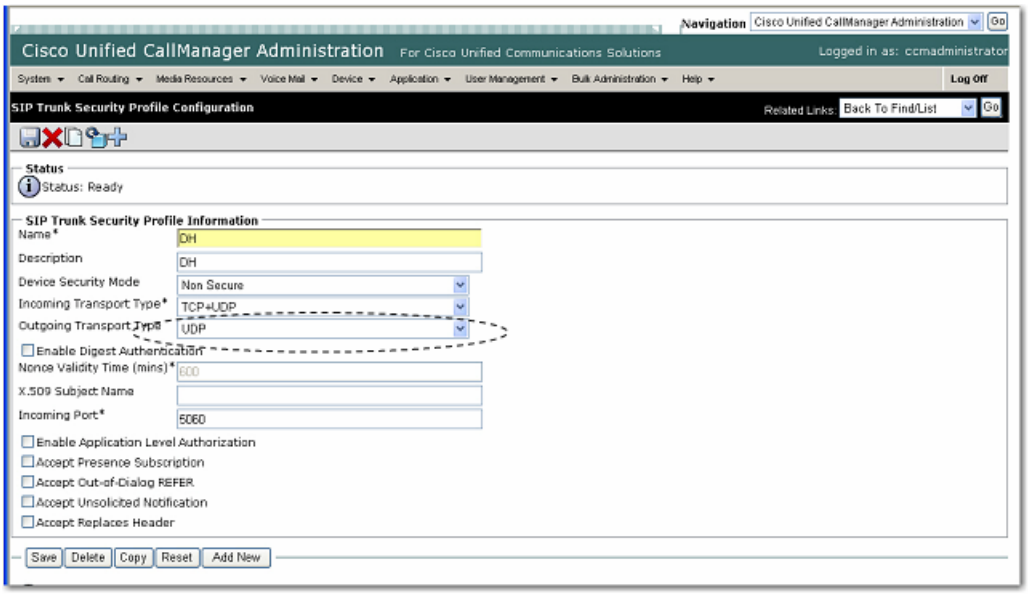
The following sections provide an overview of how TelePresence components register with CUCM, initiate a meeting, and then conclude a meeting.

TelePresence Multipoint Switch Interface to CUCM (Trunk-Side SIP)

The Cisco TelePresence Multipoint Switch (CTMS) multipoint solution connects to CUCM by way of a SIP Trunk. SIP trunks do not use the SIP REGISTER method, and thus for trunks CUCM functions solely as a Back-to-Back User Agent (B2BUA). Route Pattern(s) are configured to route multipoint calls to the SIP trunk(s) of the multipoint switch(es). Prior to software version 1.1, CTMS supported only UDP for SIP signaling to/from CUCM. As of software version 1.1, CTMS also supports TCP.

Therefore the outgoing transport type on the CUCM SIP Trunk Security Profile Configuration must be set for UDP for CTMS. This is shown in Figure 7-2.

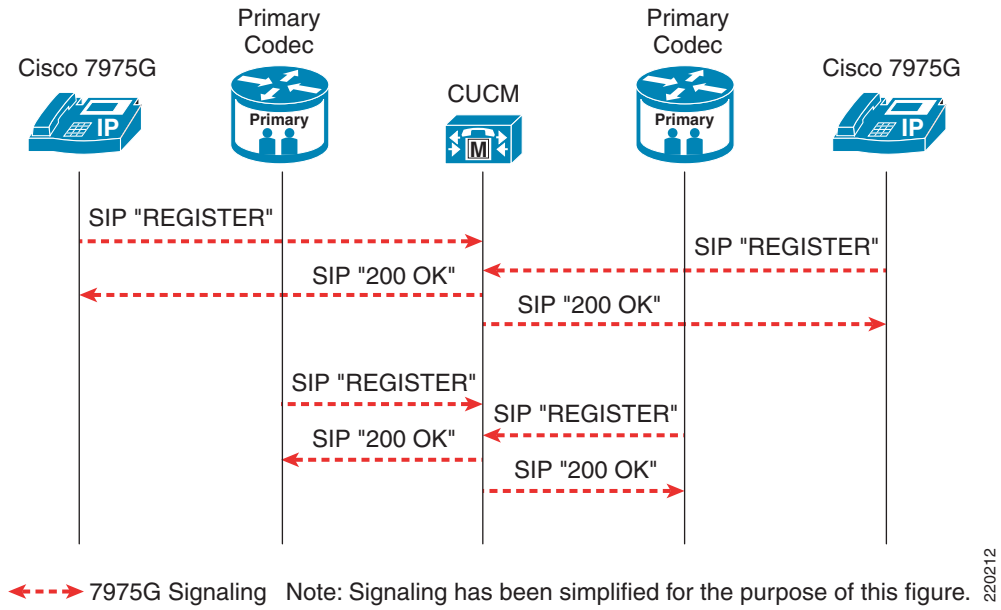
Figure 7-2 CUCM SIP Trunk Security Profile Configuration



TelePresence Endpoint Device Registration

For the initial release of the TelePresence solution, it is recommended that all TelePresence devices in a deployment register to a single CUCM cluster. Although TelePresence devices can be registered across multiple CUCM clusters, Cisco TelePresence Manager (CTS-MAN), which performs meeting scheduling, can only support a single CUCM cluster in the current release. The 7975G IP phones which function as the user interface for the TelePresence solution also register with CUCM, sharing the same dial extension as the TelePresence Codecs. Figure 7-3 shows an example of the high-level data flows in the registration process.

Figure 7-3 Cisco TelePresence Device Registration

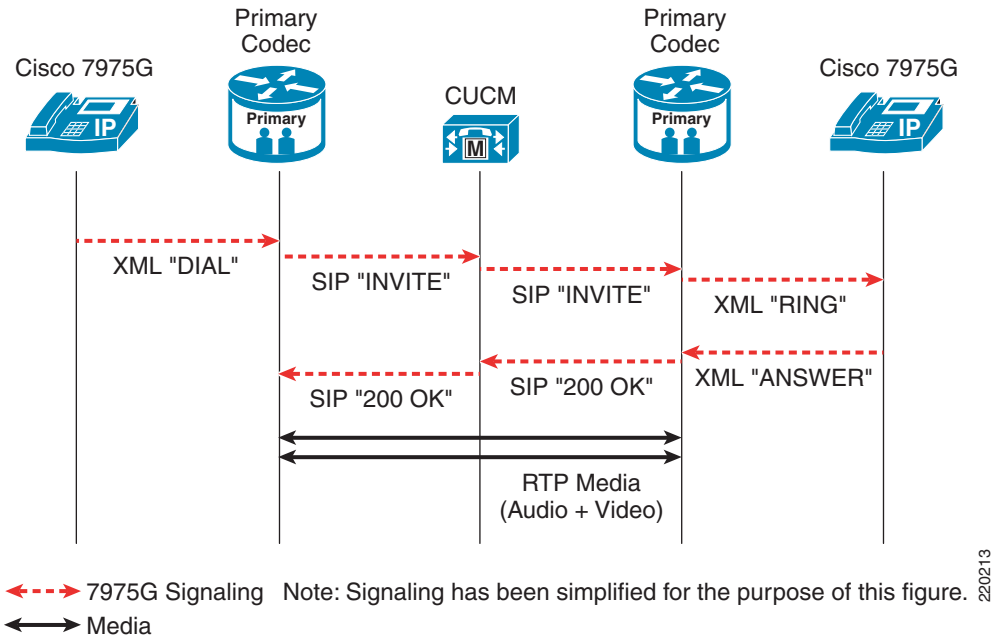


By default CUCM listens on TCP and UDP port 5060 for SIP-related signaling. Cisco TelePresence Systems and Cisco 7975G IP Phones use TCP and hence connect to CUCM on TCP port 5060. The contact header within the SIP REGISTER provides the IP address, transport protocol, port number, and the dial extension for CUCM to reach the TelePresence Codecs and 7975 IP phones.

Call Setup

Once registration is complete, meetings may be established between any two Cisco TelePresence systems or between any TelePresence System and a multipoint switch. Figure 7-4 shows a high-level overview of the call establishment signaling between TelePresence Codecs, their associated 7975G IP phones, and the CUCM cluster.

Figure 7-4 Point-to-Point Cisco TelePresence Call Setup



To make the SIP signaling easier to understand, it has been greatly simplified in Figure 7-4. SIP SUBSCRIBE and NOTIFY messages have been removed from the call flow. These messages are used primarily to update the 7975G IP phones and TelePresence Codecs regarding the status of the call. Finally, HTTP messages between TelePresence Codecs and the Cisco TelePresence Manager have also been removed. These messages inform the Cisco TelePresence Manager of the beginning and ending of a TelePresence meeting.

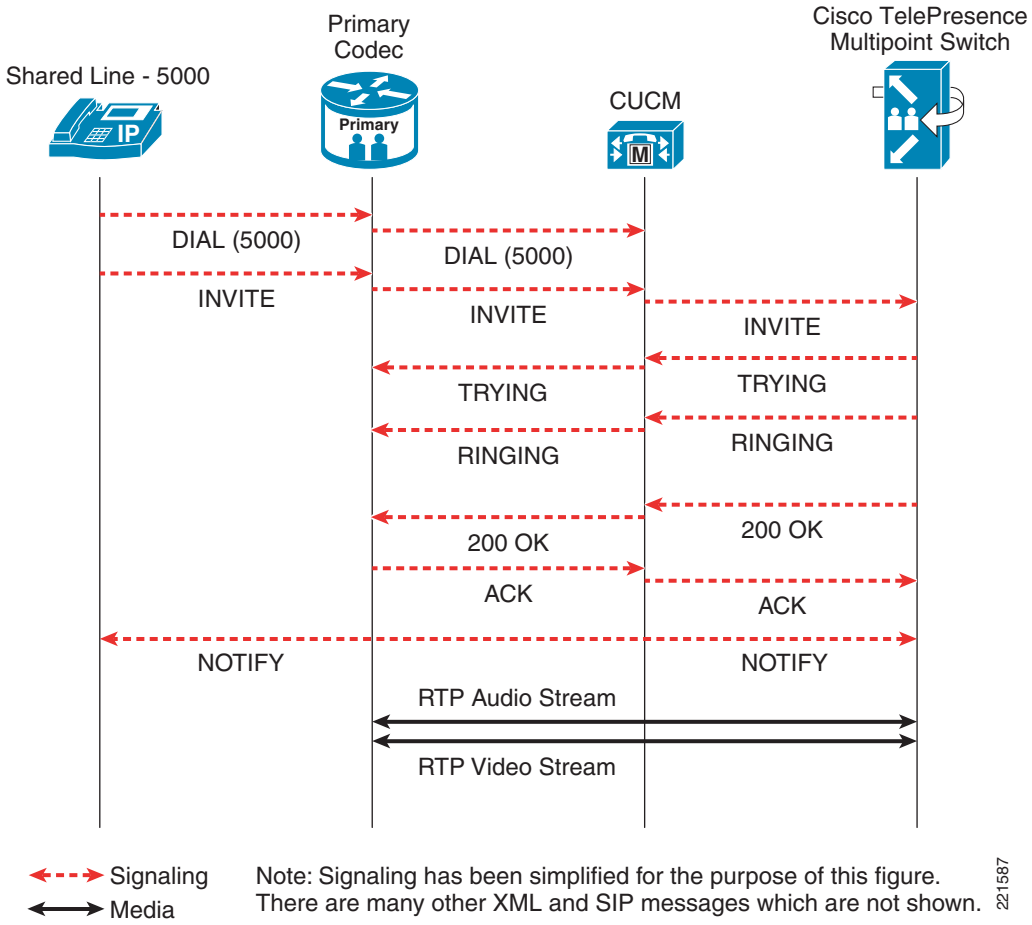
Call setup is initiated when the end user enters or selects, via the touch-screen user interface of the 7975G IP phone, the remote TelePresence location to which he or she wishes to establish a meeting. This causes the 7975G IP phone to generate an XML message to the TelePresence Codec. The XML message instructs the TelePresence Codec to generate a SIP INVITE, which is sent to the CUCM cluster. Within the initial SIP INVITE, the TelePresence Codec uses the Session Description Protocol (SDP). SDP, discussed in IETF RFC 2327, allows two endpoints which are configured for different audio or video modes to negotiate a common set of media parameters for the call. This is accomplished primarily through the use of the media (m=…), attribute (a=…), and bandwidth (b=…) lines. The quality parameter within the TelePresence device configuration in CUCM determines what media capabilities are offered in the initial SDP.

Upon receiving an INVITE from one TelePresence System and determining the destination endpoint (based on the number dialed), CUCM generates a new SIP INVITE to the remote TelePresence Codec. Upon receipt of the SIP INVITE, the TelePresence Codec informs the 7975G IP phone of the incoming call via an XML message. The end user at the remote location accepts the incoming call via the touch-screen user interface of the 7975G phone. This causes a final XML message to be sent to the remote TelePresence Codec, informing it to answer the call. After that, the audio and video media streams begin. Optionally, the TelePresence codec may be configured (in CUCM) to automatically answer all incoming calls, in which case the XML message sequence to/from the phone is skipped and the call is answered immediately. Incidentally, it should be noted that since the same dial extension is shared between the remote TelePresence Codec and the remote 7975G IP phone which functions as its user interface, CUCM generates the new SIP INVITE message to both remote devices. This allows the user to answer the call using the **handset** of the IP Phone (in which case the call is established as an audio-only call). Under normal conditions though, the TelePresence Codec is the one to answer the call and the SIP INVITE to the 7975G IP Phone is canceled.

CUCM acts as a back-to-back user agent (B2BUA), processing requests as a user agent server (UAS) and generating requests as a user agent client (UAC). Unlike a proxy server, CUCM maintains dialog state and participates in all requests sent on the dialogs it establishes. Since CUCM functions as a B2BUA, it sees the SDP information regarding the media capabilities of both sides of the TelePresence call. It determines what audio and video parameters are used for the meeting based on the parameters that are common to both TelePresence devices and what is allowed via the configuration within CUCM. The configuration parameters for the allowed audio and video rates are based on two things: the Quality Setting for each TelePresence System (e.g. 1080p-Best, 1080p-Better, 1080p-Good, 720p-Best, 720-Better and 720p-Good) and the region settings of the device pool to which the TelePresence devices belong. This allows CUCM to set up a call between two TelePresence devices which are configured for different video modes. For example, if one TelePresence device is configured for 1080p-Best while another is configured for 720p-Good, CUCM specifies 720p in the outgoing SIP message to the 1080p system, thereby negotiating the call down to 720p in both directions.

Multipoint calls are no different than point-to-point calls in that each TelePresence System dials the number of the multipoint switch in a point-to-point fashion. In other words, a multipoint call is nothing more than several point-to-point calls all landing on the same destination device (the multipoint switch). The differences are that instead of matching the dialed number to a Directory Number assigned to a registered endpoint, CUCM matches the dialed number to a Route Pattern assigned to a SIP trunk. The signaling and media negotiation sequences are otherwise the same.

Figure 7-5 Multipoint Cisco TelePresence Call Setup

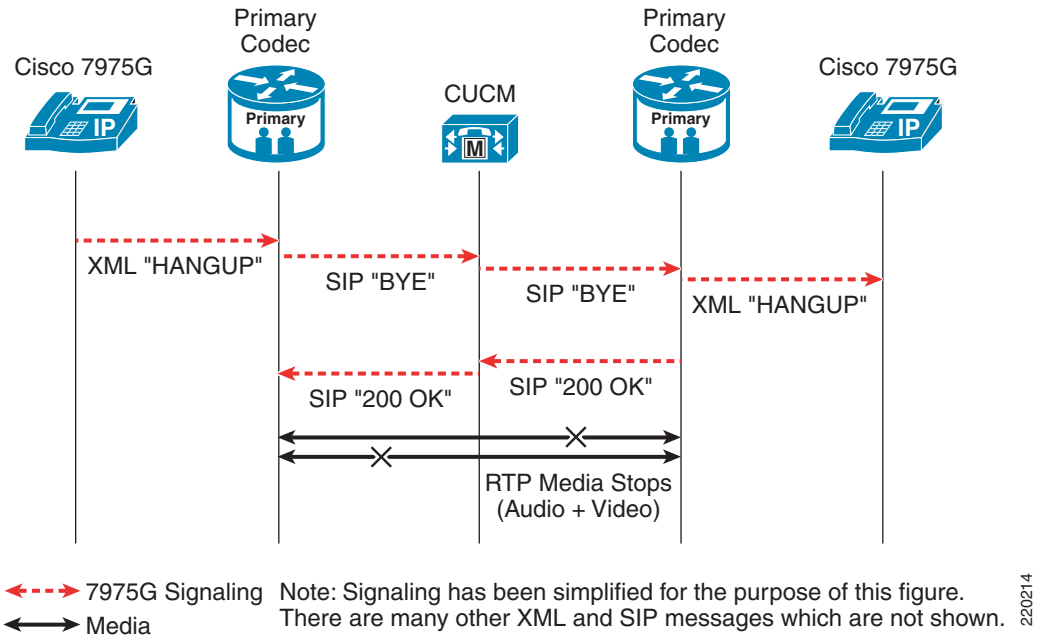


TelePresence utilizes a single AAC-LD over RTP audio stream and a single H.264 over RTP video stream in each direction, for a total of four RTP media streams per bi-directional point-to-point TelePresence meeting. This holds regardless of the model of Cisco TelePresence system device. With CTS-3000 devices, the video streams from the multiple cameras are multiplexed into a single RTP stream. Likewise, the audio streams are multiplexed into a single audio stream. The auxiliary video and audio streams are also multiplexed into these streams.

Call Teardown

Figure 7-6 shows a high-level overview of the call termination signaling between TelePresence Codecs, the 7975G IP phones which function as their user interfaces, and the CUCM cluster.

Figure 7-6 Cisco TelePresence Call Termination



To make the SIP signaling easier to understand, it has again been greatly simplified in Figure 7-6. Call termination begins when the end user at one end of a TelePresence meeting uses the touch-screen user interface of the 7975G IP phone to end the meeting. This causes the 7975G IP phone to send an XML message to the TelePresence Codec, instructing it to hang up the call by generating a SIP BYE message. The SIP BYE message is sent to CUCM, which then generates a new SIP BYE message to the remote TelePresence Codec. The remote TelePresence Codec informs the 7975G phone at the remote site that the call is terminating. Upon receipt of the SIP 200 OK messages from the TelePresence Codecs, the audio and video media streams stop.

Since CUCM functions as a B2BUA which maintains state of all SIP calls initiated and terminated through it, it can capture call detail records of when TelePresence meetings start and stop. This may be necessary for management systems and for billing charges for TelePresence meetings back to individual departments.

Firewall and NAT Considerations

TelePresence embeds the audio and video media endpoint addresses within the SIP call signaling messages. This has implications for firewalls and network address translation. For a firewall to determine the IP addresses and ports to dynamically open to allow the audio and video media through, the firewall may need to monitor the SIP signaling flow. Also, any IP address translation within the network may may require special handling, since the addressing received by the remote TelePresence device may not represent a routable IP address to the routers and Layer 3 switches at the remote site. Firewalling within an inta-enterprise TelePresence deployment is discussed in Chapter 13, “Internal Firewall Deployments with Cisco TelePresence.”