



CHAPTER 9

Call Processing Deployment Models

Overview

For the current release of the Cisco TelePresence Solution and the Intra-Enterprise Deployment Model, a single Cisco Unified Communication Manager (CUCM) cluster is recommended to support all TelePresence devices within the enterprise. TelePresence meetings currently can only be scheduled across a single cluster by the Cisco TelePresence Manager (CTS-MAN) scheduling server because CTS-MAN only supports a single CUCM cluster. Although devices can register across multiple CUCM clusters, and ad hoc and permanent meetings can be established between clusters, this design is not currently recommended for customers deploying CTS-MAN. For customers not deploying CTS-MAN, this restriction is not applicable. Furthermore, a future release of CTS-MAN is planned to support multiple CUCM clusters, at which point this restriction will be removed.

In addition, in environments where TelePresence is deployed along with other generic Videoconferencing/Video Telephony devices on the same cluster, CUCM cannot instruct Videoconferencing/Video Telephony to use the recommended AF41 QoS marking and TelePresence to use the recommended CS4 QoS marking. The marking of audio and video traffic by CUCM is handled at the cluster level and not at the device level, because the marking of audio and video traffic is a cluster-wide (i.e., global) parameter and CUCM offers only a single parameter for video, which by default is set to AF41. For this reason it is recommended that TelePresence be placed on a separate cluster from all other Videoconferencing / Video Telephony applications. Finally, Cisco TelePresence requires CUCM release 5.1.1 or higher, with version 5.1.2 recommended to support the Auto Collaborate endpoint feature of TelePresence. Therefore, to summarize the guidance based upon the above three criteria, if a customer has a single existing cluster running version 5.1.1 or higher deployed for IP telephony and has no other Videoconferencing/Video Telephony devices, it is acceptable to integrate TelePresence devices onto that cluster. However, since the vast majority of deployments are not expected to meet these criteria, it is recommended that a separate CUCM cluster be deployed to support TelePresence and the guidance contained in this document is based upon that approach.

Dial-Plan Recommendations

For the current release of TelePresence, it is recommended that the Cisco Unified 7975G IP phones that serve as the user interface to the Cisco TelePresence system endpoints be marked to indicate that they should not be used for emergency services calls. A separate IP Phone registered to the production IP Telephony CUCM cluster should be deployed in the same room to provide access to emergency services.

To support functionality such as the ability to bridge audio participants into the TelePresence meeting via the audio add-in feature of the TelePresence System, the CUCM cluster which supports the TelePresence deployment may require additional components: either one or more voice gateways

connecting the TelePresence CUCM cluster to the customer's PBX or to the PSTN, and/or one or more Inter-Cluster Trunks (either H.323 or SIP) between the TelePresence CUCM cluster and the existing IP Telephony CUCM cluster(s).

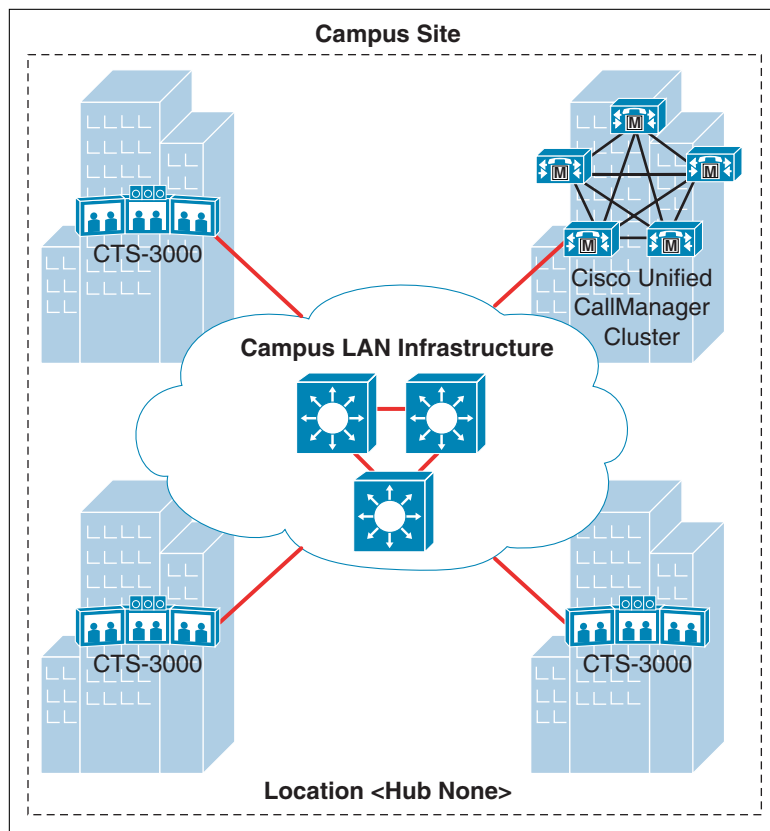
In either scenario, the TelePresence dial plan must be selected carefully and call routing set up appropriately to allow the TelePresence systems to reach and to be reached by other phones, audio conferencing bridges, and the PSTN. Therefore, the dial plan, Directory Numbers, Partitions, and Calling Search Spaces allocated to the TelePresence systems should be consistent with the rest of the enterprise to provide full support for current and future capabilities.

All current TelePresence deployments use either a single-site call processing model or a multi-site WAN with centralized call processing model. In both of these models, the CUCM cluster which supports the TelePresence devices resides at one location, such as a main campus. All communications with devices at remote locations takes place over the IP network infrastructure.

Single-Site Call Processing Model

The single-site call processing model applies to Cisco TelePresence deployments within a single campus and to deployments across MANs with LAN speed (i.e., Gigabit Ethernet) connectivity between sites. [Figure 9-1](#) shows an example of this deployment model.

Figure 9-1 Cisco TelePresence Single-Site Deployment



220218

Call Admission Control

In a single-site design, it is assumed that a high-speed LAN provides connectivity between all devices. CAC is typically not an issue, since the LAN can easily be scaled to provide sufficient bandwidth to simultaneously support all possible TelePresence meetings. TelePresence devices can be left within the default Hub_None location within the CUCM configuration, which provides no bandwidth restrictions on the total amount of video and audio traffic.

The region settings within the CUCM configuration are used to control the audio codec and the amount of video bandwidth used per call within a region and between regions. Since there are no other video devices in a standalone TelePresence deployment, all TelePresence devices can be placed in a single region. The region should be configured for AAC/Wideband audio (which as of release 5.1.1 or higher of CUCM permits up to 256 Kbps of audio per call) and a video bandwidth of at least 12500 Kbps (12.5 Mbps). As of release 5.1.1 or higher of CUCM, the maximum video bandwidth permitted is 32,256 Kbps. These settings are illustrated in Figure 9-2.

Figure 9-2 Recommended CUCM Region Settings for TelePresence

Application User Management Bulk Administration Help Log Off

Related Links: Back To Find/List Go

Audio Codec	Video Call Bandwidth
Wideband	32256
Use System Default	Use System Default

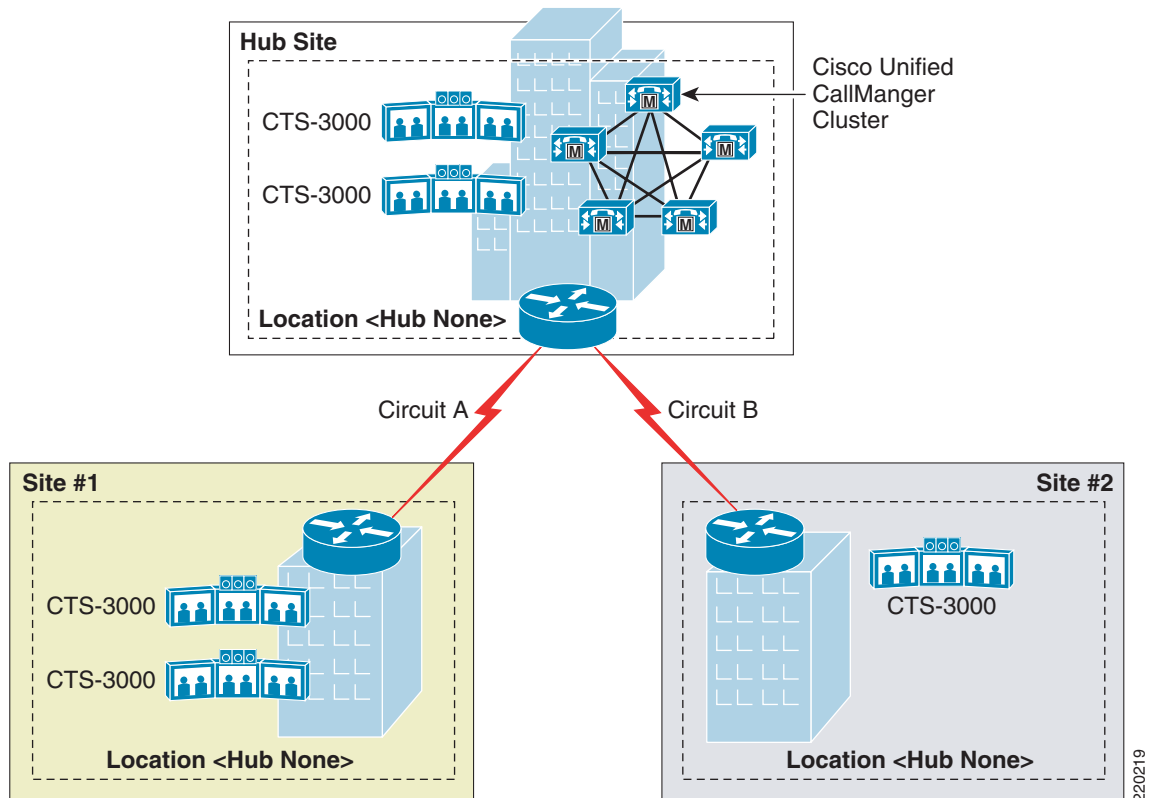
Audio Codec	Video Call Bandwidth
Wideband	<input type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input checked="" type="radio"/> None <input type="text" value="32256"/> kbps

221699

Multi-Site WAN with Centralized Call Processing Model

In a multi-site WAN with centralized call processing model, a single CUCM cluster is deployed at a central site. This acts as the call processing agent for TelePresence devices both at the local and remote sites. Figure 9-3 shows an example of this deployment model over a hub-and-spoke network topology.

Figure 9-3 Cisco TelePresence Multi-Site Deployment

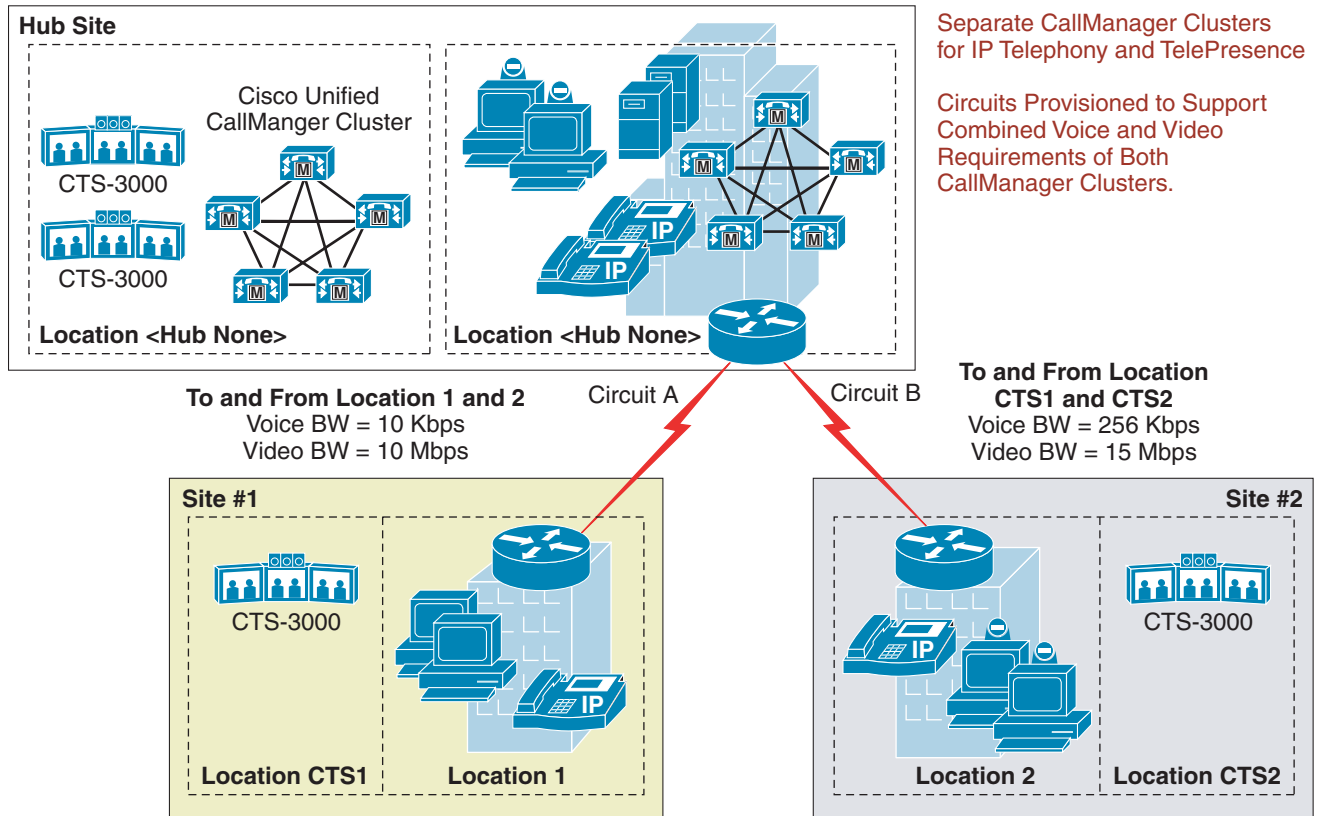


Call Admission Control

For current TelePresence deployments it is recommended that sufficient WAN bandwidth be provisioned to support all possible simultaneous meetings within the network. Refer to [Chapter 8, “Capacity Planning and Call Admission Control”](#) for details regarding the use of manual capacity planning to guarantee 100% call completion. For this design, all TelePresence devices can be left in the default `Hub_None` location which provides no bandwidth restrictions on the total amount of video and audio traffic (as shown above). Alternatively, TelePresence devices at each remote site can be assigned to a different location and the video and audio bandwidth between locations set to unlimited. It should be noted that when deploying multiple CTMS devices in a distributed multipoint TelePresence deployment, TelePresence devices need to be assigned to separate regions/locations in order for CTS-MAN to correctly choose the appropriate CTMS for the multipoint meeting. Further details are discussed in [Chapter 10, “Cisco TelePresence Multipoint Solution Essentials,”](#) [Chapter 11, “Cisco Multipoint Technology and Design Details,”](#) and [Chapter 12, “Cisco TelePresence Multipoint Solution Circuit and Platform Recommendations.”](#)

When implementing Cisco TelePresence alongside an existing CUCM deployment dedicated for IP telephony, the WAN circuits must be provisioned with sufficient bandwidth to take into account the CAC requirements of both CUCM clusters. An example of this is shown in Figure 9-4.

Figure 9-4 Separate Cisco Unified CUCM Design Example



As can be seen in Figure 9-4, separate CUCM clusters are deployed for TelePresence and for IP telephony (both dashed boxes). Each CUCM configuration has a different location configured for each remote site with a certain amount of bandwidth configured between each location for audio and video. In this scenario, the WAN circuits must be provisioned to accommodate the aggregate bandwidth pools configured in both CUCM clusters, since they operate independently of each other. Otherwise, the potential exists for oversubscribing the circuits and degrading the quality of voice, desktop video, and TelePresence meetings.

It should also be noted that the Survivable Remote Site Telephony (SRST) feature of Cisco router platforms do not currently support Cisco TelePresence system devices. Therefore in a multi-site WAN with a centralized call processing TelePresence design, SRST cannot be used to provide redundancy if the connection to the TelePresence CUCM cluster fails. However in the design shown in Figure 9-4, where a separate CUCM cluster is deployed for IP telephony devices, SRST works well for the IP phones and other devices which are supported.

