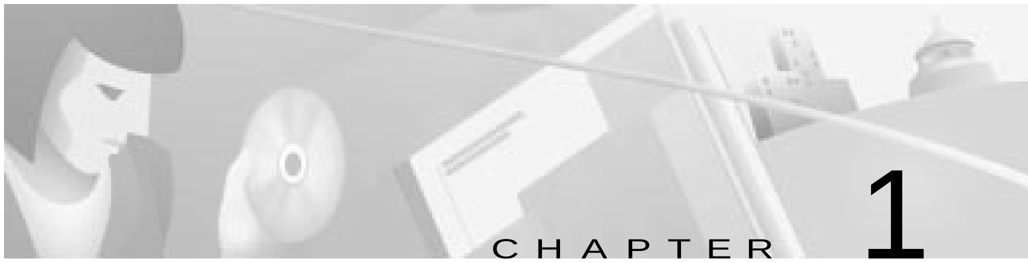


CHAPTER 1

Introduction

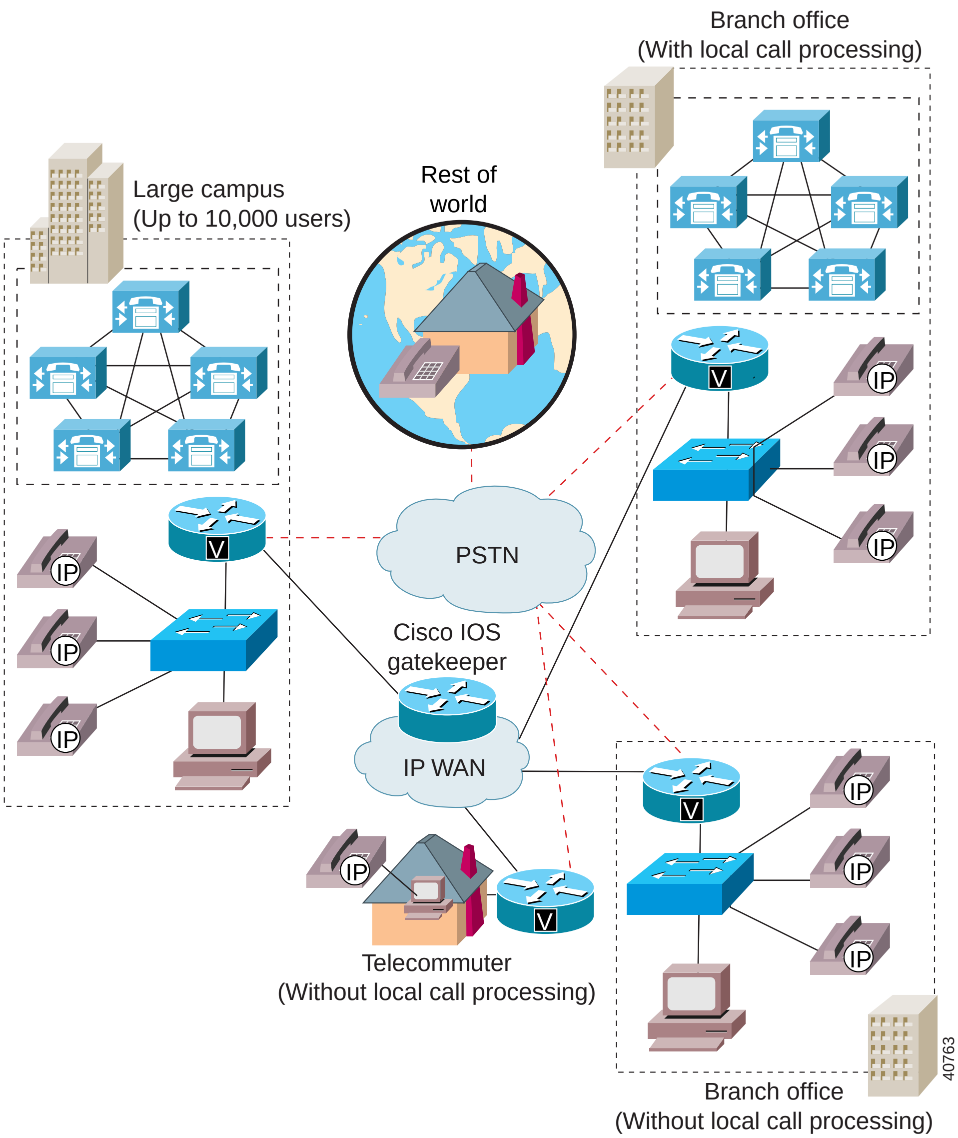
This chapter presents a high-level overview of several basic models that you can use in designing your IP telephony network. This overview provides some guidance with respect to when and why a particular design should be selected. Subsequent chapters delve into each network model in greater detail, beginning with the simplest model and building to increasingly complexity models.

This chapter includes the following major sections:

General Design Models

Figure 1-1 provides a composite scenario that illustrates the goals of the network design models discussed in this guide. This scenario represents what is possible with Cisco CallManager Release 3.0(5).

Figure 1-1 Composite Model



The overall goals of an IP telephony network are as follows:

- End-to-end IP telephony
- IP WAN as the primary voice path with the Public Switched Telephone Network (PSTN) as the secondary voice path between sites
- Lower total cost of ownership with greater flexibility
- Enabling of new applications

For IP telephony networks based on Cisco CallManager Release 3.0(5), there are four general design models that apply to the majority of implementations:

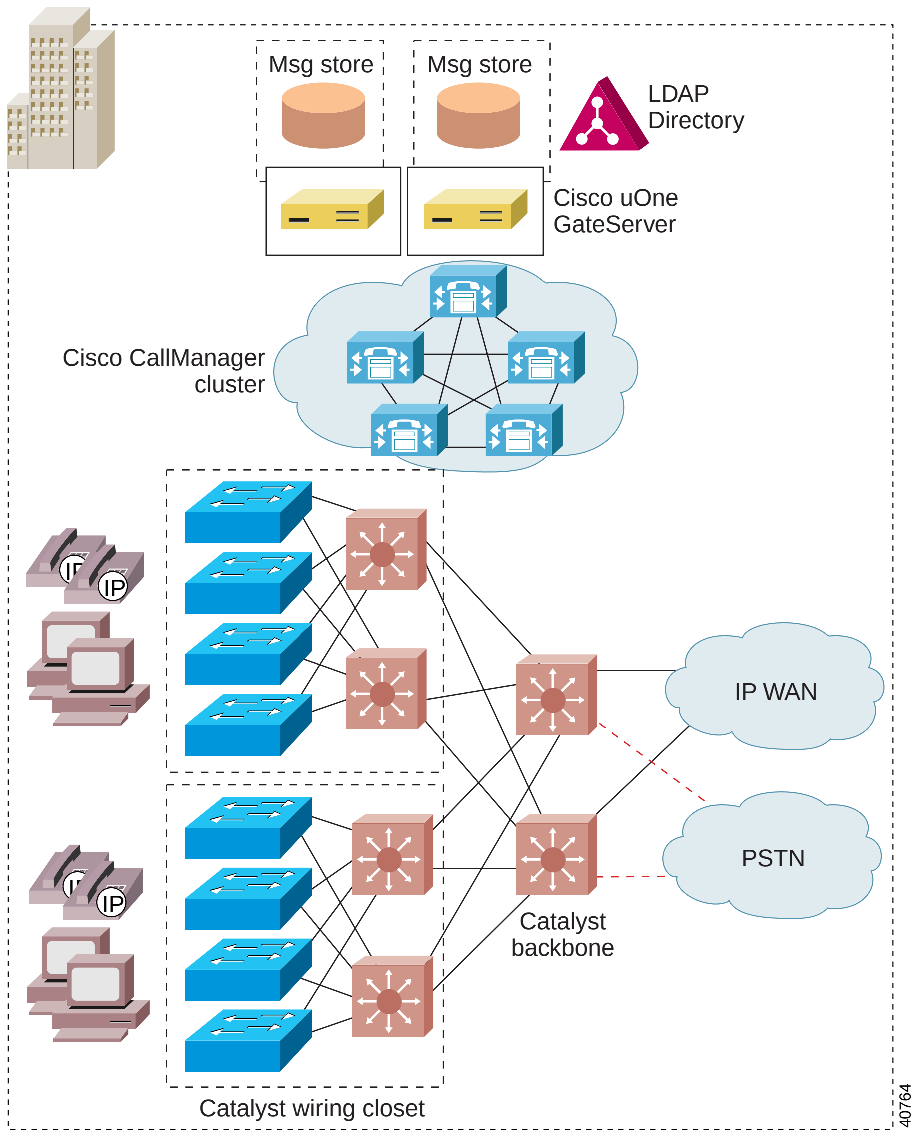
- Single-Site Model, page 1-3
- Multiple Sites with Independent Call Processing, page 1-5
- Multisite IP WAN with Distributed Call Processing, page 1-7
- Multisite IP WAN with Centralized Call Processing, page 1-10

The following sections summarize the design goals and implementation guidelines for each of these models.

Single-Site Model

Figure 1-2 illustrates the model for an IP telephony network within a single campus or site.

Figure 1-2 Single-Site Model



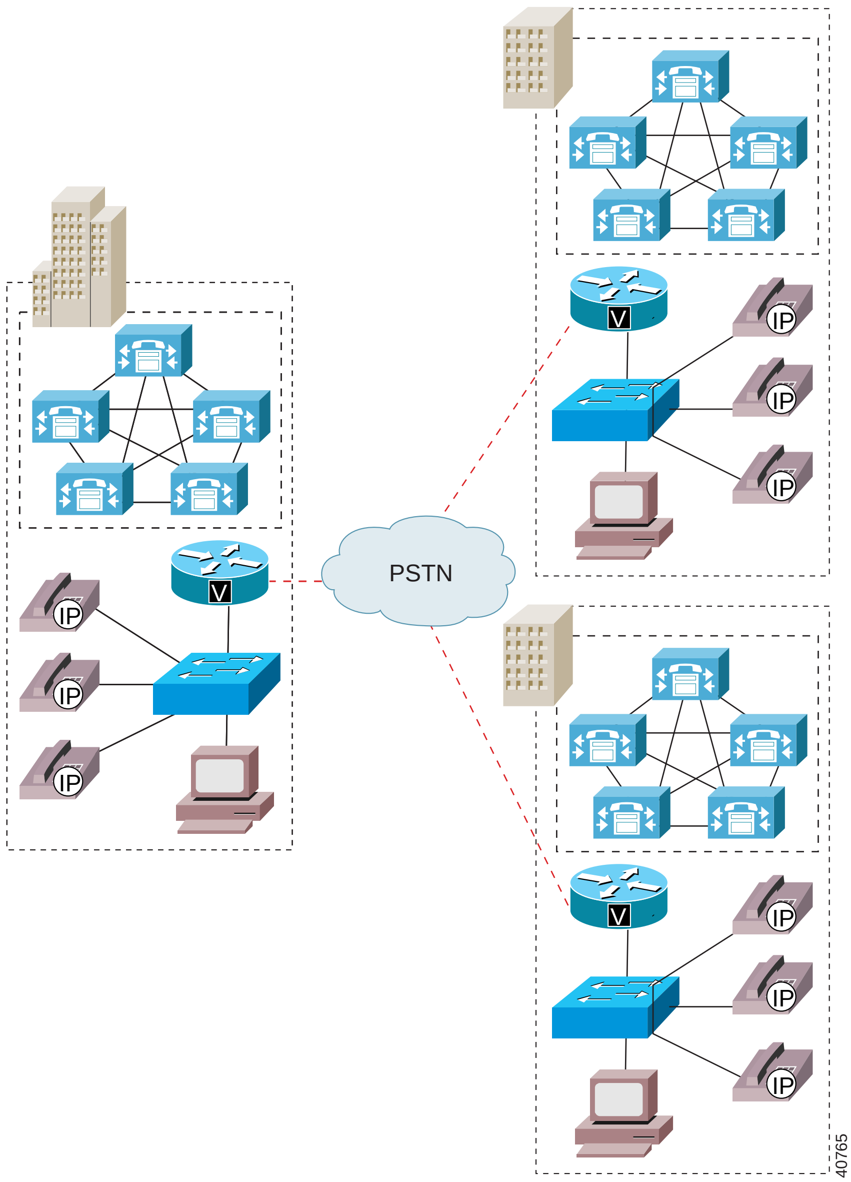
The single-site model has the following design characteristics:

- Single Cisco CallManager or Cisco CallManager cluster.
- Maximum of 10,000 users per cluster.
- Maximum of eight servers in a Cisco CallManager cluster (four servers for primary call processing, two for backup call processing, one database publisher, and one TFTP server).
- Maximum of 2,500 users registered with a Cisco CallManager at any time.
- PSTN only for all external calls.
- Digital signal processor (DSP) resources for conferencing.
- Voice mail and unified messaging components.
- G.711 codec for all IP phone calls (80 kbps of IP bandwidth per call, uncompressed).
- To guarantee voice quality, use Cisco LAN switches with a minimum of two queues. See Chapter 2, “Campus Infrastructure Considerations,” for more details.

Multiple Sites with Independent Call Processing

Figure 1-3 illustrates the model for multiple, isolated sites that are not connected by an IP WAN. In this model, each site has its own Cisco CallManager or Cisco CallManager cluster to handle call processing for that site.

Figure 1-3 Multiple Independent Sites



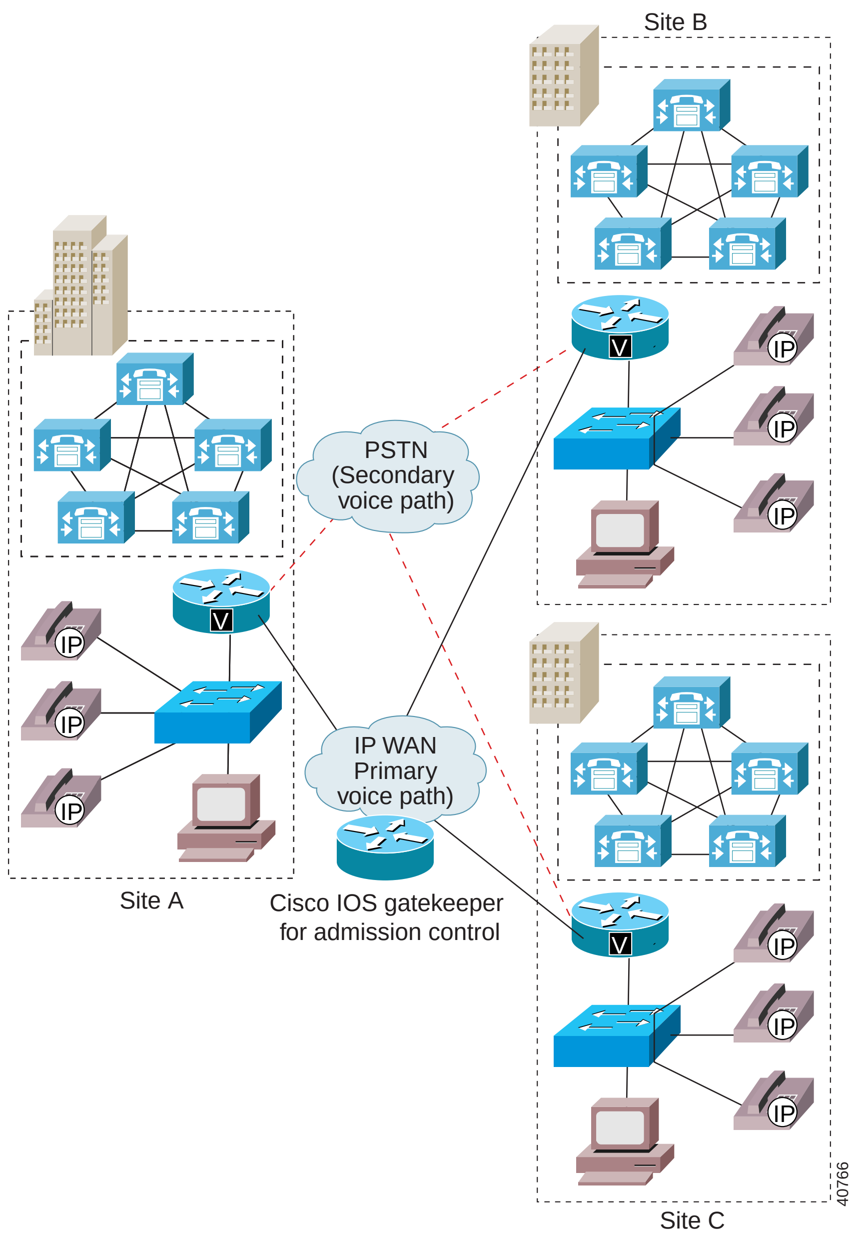
The model for independent multiple sites has the following design characteristics:

- Cisco CallManager or Cisco CallManager cluster at each site to provide scalable call control.
- Maximum of 10,000 IP phones per cluster.
- No limit to number of clusters.
- Use of PSTN for networking multiple sites and for all external calls.
- DSP resources for conferencing at each site.
- Voice message or unified messaging components at each site.
- Voice compression not required.

Multisite IP WAN with Distributed Call Processing

Figure 1-4 illustrates the model for multiple sites with distributed call processing.

Figure 1-4 Multisite Model with Distributed Call Processing



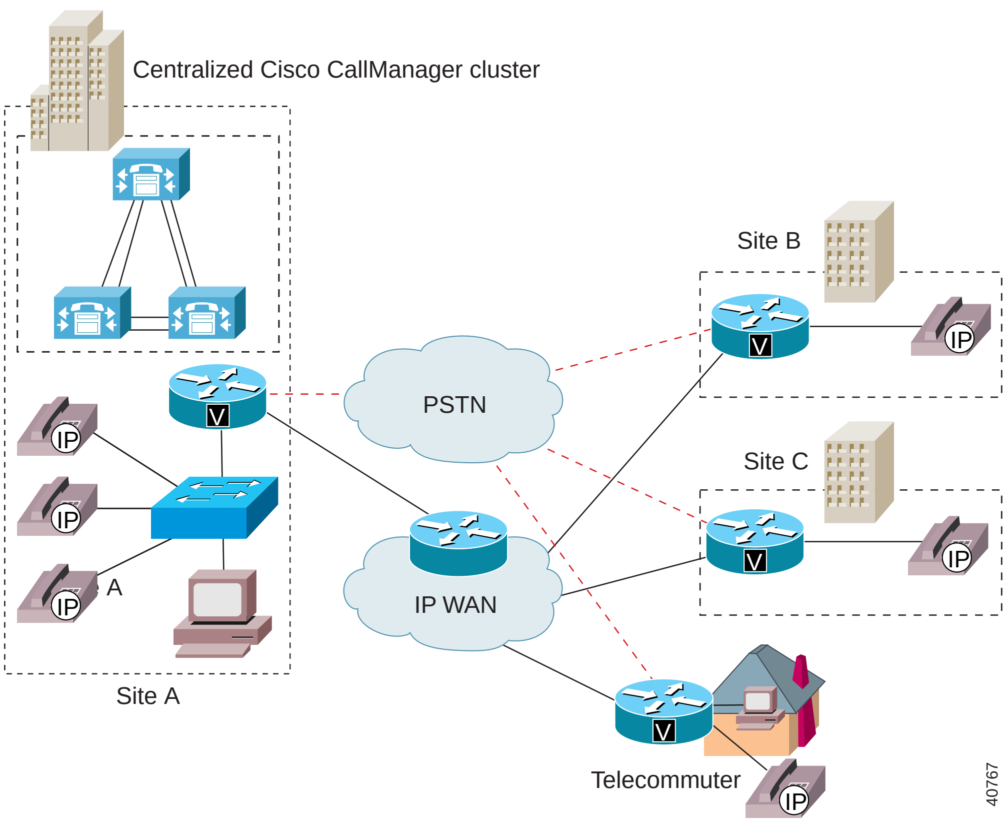
The multisite IP WAN with distributed call processing has the following design characteristics:

- Cisco CallManager or Cisco CallManager cluster at each location (10,000 users maximum per site).
- Cisco CallManager clusters are confined to a single campus and may *not* span the WAN.
- IP WAN as the primary voice path between sites, with the PSTN as the secondary voice path.
- Transparent use of the PSTN if the IP WAN is unavailable.
- Cisco IOS gatekeeper for E.164 address resolution.
- Cisco IOS gatekeeper for admission control to the IP WAN.
- Maximum of 100 sites interconnected across the IP WAN using hub and spoke topologies.
- Compressed voice calls supported across the IP WAN.
- Single WAN codec supported.
- DSP resources for conferencing and WAN transcoding at each site.
- Voice mail and unified messaging components at each site.
- Minimum bandwidth requirement for voice and data traffic is 56 kbps. For voice, interactive video, and data, the minimum requirement is 768 kbps. In each case, the bandwidth allocated to voice, video, and data should not exceed 75% of the total capacity.
- Remote sites can use Cisco IOS as well as gateways based on the Skinny Gateway Protocol.

Multisite IP WAN with Centralized Call Processing

Figure 1-5 illustrates the model for multiple sites with centralized call processing.

Figure 1-5 Multisite Model with Centralized Call Processing



The multisite IP WAN with centralized call processing has the following design characteristics:

- Central site supports only one active Cisco CallManager. A cluster can contain a secondary and tertiary Cisco CallManager as long as *all* IP phones served by the cluster are registered to the same Cisco CallManager at any given time. This is called a *centralized call processing cluster.*
- Each centralized call processing cluster supports a maximum of 2500 users (no limit on number of remote sites). Multiple centralized call processing clusters of 2500 users at a central site can be interconnected using H.323.
- IP phones at remote sites do not have a local Cisco CallManager.
- The call admission control mechanism is based on bandwidth by location. See the "Call Admission Control" section on page 7-3.
- Compressed voice calls across the IP WAN are supported.
- Manual use of the PSTN is available if the IP WAN is fully subscribed for voice traffic (PSTN access code must be dialed after a busy signal).
- Dial backup is required for IP phone service across the WAN in case the IP WAN goes down.
- Voice mail, unified messaging, and DSP resource components are available at the central site only.
- Minimum bandwidth requirement for voice and data traffic is 56 kbps. For voice, interactive video, and data, the minimum requirement is 768 kbps. In each case, the bandwidth allocated to voice, video, and data should not exceed 75% of the total capacity.
- Remote sites can use Cisco IOS as well as gateways based on the Skinny Station Protocol.
- If using voice mail, each site must have unique internal dial plan number ranges. You cannot overlap internal dial plans among remote sites if voice mail is required. (For example, no two sites can share 1XXX.)