CHAPTER 18

IP Telephony Migration Options

Last revised on: December 15, 2008

This chapter describes several methods for migrating from separate standalone voice, video, and collaboration systems to an integrated Cisco Unified Communications System. The major topics discussed in this chapter include:

What's New in This Chapter

Table 18-1 lists the topics that are new in this chapter or that have changed significantly from previous releases of this document.

Table 18-1 New or Changed Information Since the Previous Release of This Document

New or Revised Topic	Described in:
Migration of collaboration systems	Migration of Voice and Desktop Collaboration Systems, page 18-6
Migration of video systems	Video Migration, page 18-5

IP Telephony Migration

There are two main methods for migrating to an IP Telephony system (or any other phone system, for that matter):

- Phased Migration, page 18-2
- Parallel Cutover, page 18-3

Neither method is right or wrong, and both depend upon individual customer circumstances and preferences to determine which option is most suitable.

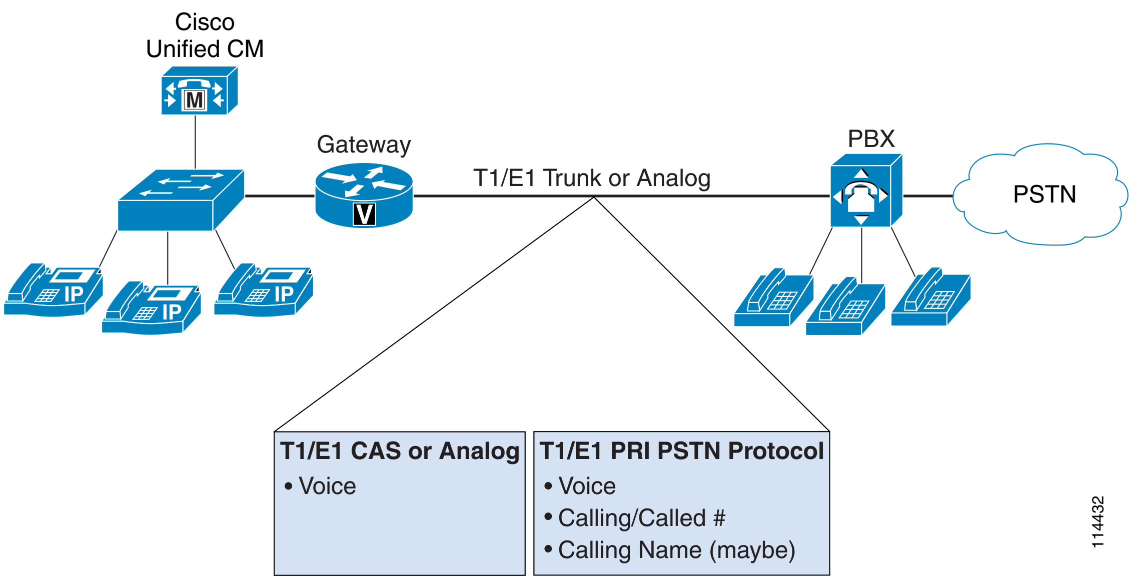
IP Telephony migration also involves The Need for QSIG in Multisite Enterprises, page 18-3.

Phased Migration

This approach typically entails a small initial IP Telephony deployment that is connected to the main corporate PBX. The choice of which signaling protocol to use is determined by the required features and functionality as well as by the cost of implementation. Cisco Unified Communications Manager (Unified CM) can support either regular PSTN-type PRI or QSIG PRI as well as H.323 and SIP. Of these options, T1/E1 QSIG provides the highest level of feature transparency between the two systems.

PSTN-type PRI provides for basic call connectivity as well as Automatic Number Identification (ANI). In some instances, the protocol also supports calling name information, as illustrated in Figure 18-1.

Figure 18-1 Features Supported by Signaling Protocols



This level of connectivity is available to *all* PBXs; that is, if the PBX can connect to the public network via PRI, then it can connect to Unified CM because Unified CM can be configured as the "network" side of the connection.

Cisco Unified CM, Release 3.3 and later, incorporates the International Standards Organization (ISO) variant of QSIG. The QSIG protocol allows for additional feature transparency between PBXs from different vendors, over and above the features that can be obtained from PSTN-type PRI, and it is therefore more appropriate for large enterprises that are already operating complex networks. (Refer to The Need for QSIG in Multisite Enterprises, page 18-3.)

With either PSTN-type PRI or QSIG, the process of phased migration is similar: move subscribers from the PBX to Unified CM in groups, one group at a time, until the migration is complete.

The Cisco San Jose campus, consisting of some 23,000 subscribers housed in approximately 60 buildings, was migrated to IP Telephony in this manner and took just over one year from start to finish. We converted one building per weekend. All subscribers in the selected building were identified, and their extensions were deleted from the PBX on a Friday evening. At the same time, additions were made to the PBX routing tables so that anyone dialing those extension numbers would then be routed over the correct PRI trunk for delivery to Unified CM. During the weekend, new extensions were created in Unified CM for the subscribers, and new IP phones were delivered to their appropriate locations, ready for use by Monday morning. This process was simply repeated for each building until all subscribers had been migrated.

Parallel Cutover

This approach begins with implementation of a complete IP Telephony infrastructure that is redundant, highly available, QoS-enabled, and equipped with Ethernet ports that are powered inline. Once the infrastructure is complete, the IP Telephony application can then be deployed. All IP phones and gateways can be fully configured and deployed so that subscribers have two phones on their desk simultaneously, an IP phone as well as a PBX phone. This approach provides the opportunity to test the system as well as allowing subscribers time to familiarize themselves with their new IP phones. Outbound-only trunks can also be connected to the IP Telephony system, giving subscribers the opportunity to use their new IP phones to place external calls as well as internal calls.

Once the IP Telephony system is fully deployed, you can select a time slot for bringing the new system into full service by transferring the inbound PSTN trunks from the PBX to the IP Telephony gateways. You can also leave the PBX in place until such a time as you are confident with the operation of the IP Telephony system, at which point the PBX can be decommissioned.

A parallel cutover has the following advantages over a phased migration:

- If something unexpected occurs, the parallel cutover provides a back-out plan that allows you to revert to the PBX system by simply transferring the inbound PSTN trunks from the IP Telephony gateways back to the PBX.
- The parallel cutover allows for verification of the IP Telephony database configuration before the system carries live PSTN traffic. This scenario can be run for any length of time prior to the cutover of the inbound PSTN trunks from the PBX to the IP Telephony gateways, thereby ensuring correct configuration of all subscriber information, phones, gateways, the dial plan, and so forth.
- Training can be carried out at a more relaxed pace by allowing subscribers to explore and use the IP Telephony system at their own leisure before the cutover of the inbound PSTN trunks.
- The system administrator does not have to make special provisions for "communities of interest." With a phased approach, you have to consider maintaining the integrity of call pick-up groups, hunt groups, shared lines, and so forth. These associations can be easily accounted for when moving the complete site in a parallel cutover.

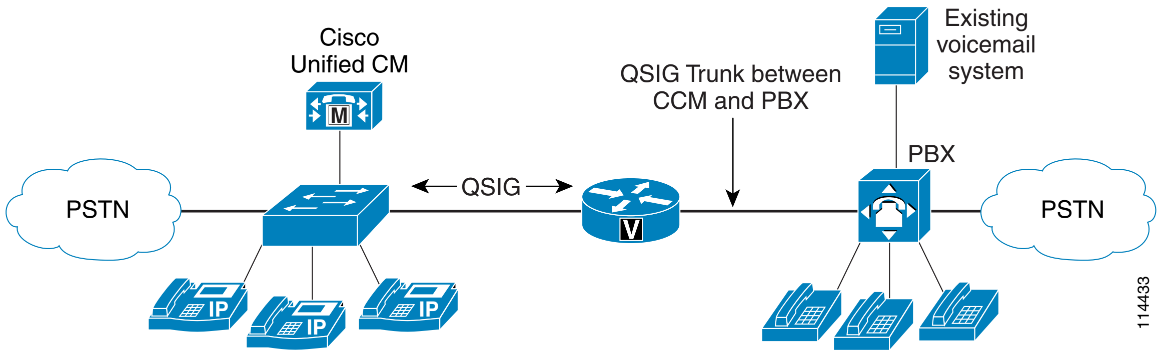
One disadvantage of the parallel cutover is that it requires the IP Telephony solution to be fully funded from the beginning because the entire system must be ready prior to bringing it into service. With a phased migration, on the other hand, you can purchase individual components of the system when they are needed, and this approach does not prevent you from starting with a small trial system prior to moving to full deployment.

The Need for QSIG in Multisite Enterprises

While some enterprises consist of only one location, others consist of many sites, some of which may potentially be spread over large distances. PBX networks for multisite enterprises are usually connected using T1 or E1 PRI trunks (depending on location) running a proprietary protocol such as Avaya DCS, Nortel MCDN, Siemens CorNet, NEC CCIS, Fujitsu FIPN, or Alcatel ABC, among others. These proprietary networking protocols enable the PBXs to deliver a high level of feature transparency between subscribers.

QSIG was developed to enable the interconnection of PBXs from different vendors, thereby allowing similar levels of feature transparency. Cisco first added QSIG to Unified CM Release 3.3 to enable Unified CM to be part of a large enterprise network. (See Figure 18-2.)

Figure 18-2 QSIG Used Between Cisco Unified CM and a PBX



QSIG as implemented in Cisco Unified CM Release 4.1 supports the following features:

- Basic call
- Direct Inward Dialing (DID)
- Calling number
- Called number
- Connected name
- Transfer (by join)
- Message Waiting Indication (MWI)
- Divert (by forward-switching)
- Calling name restriction
- Calling number restriction
- Divert (by re-route)
- Divert (responding to "check restriction" request)
- Alerting name (on-ringing)
- Path Replacement
- Callback — Call Completion Busy Subscriber (CCBS) and Call Completion No Reply (CCNR)

By supporting QSIG, Unified CM can be introduced into a large enterprise network while also maintaining feature transparency between subscribers. PBX locations can then be converted to IP Telephony whenever convenient.

However, unless you already have QSIG enabled on your PBX or have a specific need for its additional features and functionality, the cost of upgrading the PBX might be hard to justify if it will be retired within a short period of time. For example, why spend $30,000 on enabling the PBX for QSIG if you plan to retire the PBX in two or three months?

Summary of IP Telephony Migration

Although both methods of IP Telephony migration work well and neither method is right or wrong, the parallel cutover method usually works best in most cases. In addition, large enterprises can improve upon either migration method by using QSIG to enable Unified CM to become part of the enterprise network.

Cisco has a lab facility dedicated to testing interoperability between Unified CM and PBX systems. The results of that testing are made available as Application Notes, which are posted at

http://www.cisco.com/go/interoperability

The Application Notes are updated frequently, and new documents are continuously added to this website. Check the website often to obtain the latest information.

Video Migration

Typically one of the following scenarios will exist in an enterprise network:

- A dedicated H.323 video network consisting of a Multipoint Control Unit (MCU) along with either desktop or room-based endpoints
- ISDN-enabled room-based endpoints

Before attempting to integrate any form of video with Unified Communications, you must consider the dial plan(s) between the two distinct networks. The dial plans must be compatible, with no overlapping numbers present.

Dedicated H.323 Video Network

Integration of video with Unified Communications is best achieved by the deployment of an H.323 gatekeeper because this method provides connectivity between the two networks. A gatekeeper provides for dial-plan resolution as well as call admission control.

A Cisco Unified Border Element can also be deployed to provide the following capabilities:

- Ability to handle differences in network addressing (for example, 10.x.x.x and 192.x.x.x)
- Number translations (adding or modifying calling/called numbers)
- Multiple layers of call admission control, such as RSVP and simultaneous call accounting as well as bandwidth accounting
- Resolution of interworking issues such as DTMF, SIP-to-H.323 conversions, and resolution of dial-plan numbers to IP addresses

Use of the Cisco Unified Border Element is best suited for enterprises that want to maintain separate networks for reasons of management and/or feature support.

The physical migration of H.323 endpoints can be achieved by simply moving them to the Unified Communications system, in which case they would continue to be supported as either H.323 endpoints or in some cases even SIP endpoints, assuming that option is supported. H.323 MCUs can also be migrated to the Unified Communications system and would be connected via an H.323 gatekeeper. This method is best suited for customers who wish to migrate fully to a Unified Communications solution and do not wish to support multiple networks.

ISDN Endpoints

Customers have the option of continuing to place/receive calls via the PSTN from/to a Unified Communications system, much as they would operate these endpoints prior to migration. PSTN calls to the Unified Communications system can be eliminated by implementing a toll-bypass configuration that connects the ISDN interface(s) to an IP gateway, thereby making use of the customer's IP network.

Additionally, some endpoints offer the capability to connect directly via IP (H.323 and/or SIP), and this method is preferred over a toll-bypass implementation.

Migration of Voice and Desktop Collaboration Systems

Typically these systems fall into one of two categories:

- Hosted via PSTN access
- On-premises

Hosted Systems

Customers may migrate from a hosted solution to an on-premises solution by first building out the on-premises system and then choosing an appropriate time to begin using this new system. This method is best suited for enterprises that are increasing in size and want to reduce ongoing subscription costs.

On-Premises System

These systems can be considered separately in terms of voice and desktop collaboration; in other words, a collaboration session can be carried out by using a separate solution for voice along with another solution for the desktop. Migration from these systems to Unified Communications can be accomplished by moving the voice and desktop solutions independently from each other.

This method is best suited for customers who wish to move to a collaboration solution that offers both consistency and simplicity across both voice collaboration and desktop collaboration. The Unified Communications solution provides the ability to record a conference that includes both voice and desktop collaboration along with the ability to dial out to conference attendees.