IP Telephony Migration Options

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 CallManager 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 19-1.
Parallel Cutover

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 Cisco Unified CallManager because Cisco Unified CallManager can be configured as the “network” side of the connection.

Cisco Unified CallManager, 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 Multiple-Site Enterprises, page 19-3.)

With either PSTN-type PRI or QSIG, the process of phased migration is similar: move subscribers from the PBX to Cisco Unified CallManager 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 Cisco Unified CallManager. During the weekend, new extensions were created in Cisco Unified CallManager 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...
The need for QSIG in multiple-site 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 Cisco Unified CallManager Release 3.3 to enable Cisco Unified CallManager to be part of a large enterprise network. (See Figure 19-2.)
Figure 19-2  QSIG Used Between Cisco Unified CallManager and a PBX

QSIG as implemented in Cisco Unified CallManager 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, Cisco Unified CallManager 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

Although both methods of 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 Cisco Unified CallManager to become part of the enterprise network.

Cisco has a lab facility dedicated to testing interoperability between Cisco Unified CallManager 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.