2.3 CABLING

The cables pulled between telephone devices represent a significant portion of the investment in the phone system. It is important to ensure that the cabling is appropriate for that location and is installed correctly. Today, category (CAT) 5e twisted pair cable is the most popular cabling system. It carries both voice and data traffic at gigabit-per-second speeds.

The jack linking a cable to the desktop varies, depending on whether a telephone or a network device (such as a PC Network Interface Card (NIC) is connected. The Ethernet NIC uses an RJ 45 plug, but a standard analog telephone utilizes an RJ11 plug. When the cabling system is installed, the vendor tests each line for integrity. It is important to ensure that this testing is performed and that test reports are provided on each line.

The other end of the cable terminates close to the PBX, normally at a distribution frame or punch down block. The distribution frame is a rack-like structure where cables are threaded from an entry point to the appropriate exit point. The telephone engineer establishes the connection using a special-purpose tool that pushes the copper wire into a receiving contact. A dedicated corporate telephone network—where phones are connected directly to the PBX through a structured cabling system—increases reliability, but decreases flexibility. Moves, adds and changes (MACs) in the legacy PBX environment often require reconfiguring the wiring infrastructure. According to many enterprise telecom managers, a typical mid-size enterprise experiences MACs that involve approximately 12 percent of its users every year, with an average cost of $150 per user. Therefore, MACs in a traditional PBX environment are a significant, yet hidden, cost of ownership.

In contrast, data network cabling terminates desktop wires on a patch panel, so that an Ethernet drop cable can link the desktop device to its corresponding Ethernet port. This same scheme is increasingly being used for voice cabling, because it significantly reduces the costs of handling MACs.

2.4 BASIC FEATURES AND FUNCTIONS

A telephony system is expected to deliver basic features and functions, and we expect these features to behave in a predictable and familiar manner. Following is a list of the typical features available to users and administrators:

- Speaker button
- Mute call button
- Call forward
- Call transfer
- Blind transfer
- Call park
- Conference
- Hunt groups
- LCD displaying calling information
- Support for DTMF codes
- Programmable keys
- Redials

- Music on hold
- Last number redial
- Call pickup
- Shared line ringing
- Line hold (Hold)
- Speed dial

Value-added features are often embedded in telephone handsets to encourage customers to upgrade in order to gain access to these functions (handsets represent a large portion of the overall cost of owning a PBX). The feature lists associated with these handsets are fairly similar from one vendor to the next. Unfortunately, adding features in new handsets requires significant engineering in the central PBX every time a new feature is added. And even more problematic, the list of required features is exploding. The good news is that as the market continues to move to a all-pervasive IP environment, adding new features is similar to loading a new plug-in for a Web browser. This ability to increase the functionality of voice communications is a critical driver for the adoption of next-generation telephony.

2.5 ENHANCED FEATURES AND APPLICATIONS

Beyond the basic feature list, PBX vendors are scrambling to develop additional application components that can be added to the system, in order to significantly increase the types of services provided by the phone system. The final section of this guide provides an in-depth view of some of the more strategic next-generation applications, such as unified messaging, voice recognition and CRM. These adjunct systems are frequently listed by PBX vendors in their solution offerings:

- Voice mail
- Automated attendant
- CTI connectivity
- Conference bridge

Often, these systems are not fully integrated within the PBX itself, but are part of an increasing number of system adjuncts that reside outside the chassis and are linked via several line interfaces. The cost of such applications is beyond the basic PBX purchase and significantly increases the price of the overall system.

The model of adding value to the system using third-party devices is made easier when the voice system and applications are designed from the ground up to share a common IP infrastructure. The advantages of different architectures are covered in Section 3.

2.6 CALL FLOWS AND DIAL PLANS

When installing a voice communication system, one of the most important decisions that must be made is how calls are routed, even when the person is not available to take the call. Will calls be transferred to the auto-attendant, operator, assistant, off-site number, pager or cellular phone?

In evaluating how to determine call routing policies, it is imperative to seek input from system users, particularly high volume users and groups. For service centers and customer reps, “hunt groups" and workgroups often must be defined. The term "hunt group" describes the way a call might be handled by the phone system. For example, if a call is not answered by a customer agent after a few rings, it is forwarded to the next available phone in the agent group until it is picked up. If the call reaches the end of the available extensions without being picked up, it may be passed on to the group's voicemail. Understanding and configuring such functionality is critical to building a successful system.

The call handling process also must be carefully planned for outbound calls in such a way that, for any number dialed, a corresponding route is available for it. For very large multi-site systems with local hop off, the dial plan information can become quite complex. Here are a few examples of call handling policies:

• Your New York office is linked to the Dallas office. You would like to save money by routing long-distance calls over your company network. For example, if someone places a call from the Dallas office to an external number in New York, the call transits between internal PBX systems in those two offices. The external call only has to make a local hop to the destination number, saving long distance charges.

• You make a deal with a long distance carrier for calls made to London, so that all international calls to the country code 44 are prefixed with the alternate carrier's prefix number.

• When a staff member calls an internal extension using the full external number prefixed by 9, the system automatically strips off all but the extension number and routes the call internally. Whether an external service organization is involved or not, it's clear that defining dial plans requires careful analysis and thought.

2.7 AUTOMATED ATTENDANT

The auto-attendant provides a customizable way for incoming calls to be quickly routed to their destinations. This application uses in-band signaling called Dual Tone Multi-Frequency (DTMF) codes. DTMF assigns a certain sound frequency to each telephone key, so when the dial pad keys are pressed, the auto-attendant "hears" these frequencies, interprets the information contained in these frequencies and acts on the information.

For small businesses, the immediate advantage of an auto-attendant is the cost savings of not hiring an operator. However, keep in mind that this feature can frustrate potential customers if the menu levels get too deep. When comparing features of an auto-attendant application, consider these questions:

1. How many menu levels can the system provide?
2. Can I design different menus depending on time of day and year? How many?
3. Can these menus be programmed to automatically update themselves on a particular time/date?
4. How does the system handle incorrect user input?
5. Can the seasoned user go straight to a destination?
6. Can the user bypass a long prompt or are they forced to wait?
7. Does the system provide directory search with name lookup?

8. Can the system forward calls to workgroups or call center agents?

2.8 CALL DETAIL RECORDS AND BILLING

It is important to manage the overall cost of running the telephone system. PBX systems typically generate detailed logs of calls on the system. These call logs can be outputted from the management console and saved to file for processing and analysis.

Because PBXs are isolated from the IT infrastructure, the generated call detail information is fed into a report engine that produces more structured reports by department, group or usage cost. This information can be used to answer questions like:

- What calls are being made outside office hours and where are calls being placed?
- Which extensions are costing the organization the most money?
- What are the phone usage costs by department?

Caller Line Identification (CLI) can be used to determine the duration of calls from specific customers. This information can be useful for basic customer billing or service level review. More detailed statistics require a call center-type system. The process of generating such reports can be outsourced to third parties that take the basic PBX data and convert it into useful reports. Service organizations like legal, advertising, etc. that bill by the hour use such call detail records as input into customer billing systems.

2.9 NEXT STEPS

The information presented to this point provides a basic understanding of business telephony – at least, the way it used to be. But the reality is that the world of voice communication is changing, and as a result, next-generation IP technologies are replacing outdated TDM technologies in the enterprise. This section includes an introduction to data communication technologies and explains how IP voice communication is delivered on top of this infrastructure. Data networking professionals may want to skip ahead to Chapter 4.