Chapter 2. ENTERPRISE VOICE SYSTEMS

Let's start with an overview of a typical voice system. What are its components and what functions should you expect from such a system?

PBX systems – based on time-division multiplexing (TDM) – were traditionally deployed by large enterprises, until the arrival of professional- grade IP telephony systems in the late 1990s.

Before we drill down into the details, let's review the PBX in general terms. These systems typically include:

Telephone handsets
Cables connecting the telephones
Line interfaces to the phone cables
Switching and call processing to make calls
Trunk interfaces to communicate with the outside world
Management console and ability to track and account for calls
Applications and enhanced services

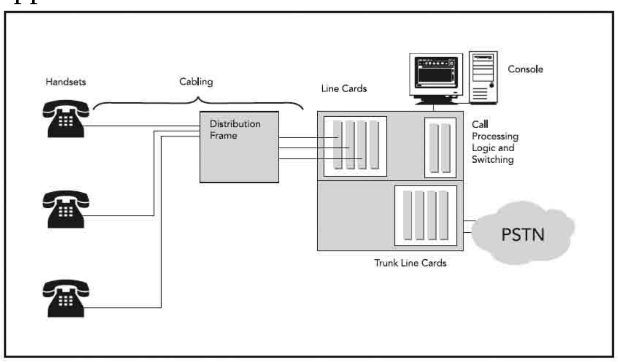

Figure 1: Components of the Legacy PBX

For these components to fulfill their tasks, software and signaling capabilities are also required. The next section explores the functionality of each element and discusses its contribution to the overall solution.

2.1 CALL SWITCHING, PROCESSING AND SIGNALING

To understand what PBX switches do, it helps to travel back in time, to before switches existed. At that time, a switchboard operator was required to set up a call between two phones. A call could take place only after a continuous connection of wire had been established from the calling party to the called party, to form a circuit. The switchboard consisted of a wooden panel with cables and jacks, and an operator connected a cable to the plug of each party in order to set up a call. Things could get fairly involved when setting up a long distance or international call. Operators had to talk to each other as they established a continuous circuit across many of these switchboards. Just like the switchboard operators of the past, today's PBX switches must remember what everyone is doing at each moment in time, and connect telephone calls between the appropriate places. The switch effectively establishes a circuit between the called parties, and the act of establishing this circuit (i.e., setting up and terminating calls) is referred to as call processing.

Call processing is accomplished using specific signaling protocols between the PBX and attached handsets, adjacent PBXs, and Public Switched Telephone Network (PSTN). In some cases, these protocols tend to be vendor-specific and proprietary, while in other cases, the protocols are based on national or international standards. The list shows the protocols used to communicate between various devices:

• PBX and Analog Handsets – Standard signaling protocols
• PBX and Digital Handsets – Proprietary, vendor-specific signaling protocols
• PBX and Central Office (CO) Exchange – Standard signaling protocols
• PBX to PBX – Both proprietary and standard signaling (feature loss with standard signaling)

In general, customers moved to non-standard signaling to take advantage of enhanced (though often unused) functionality. This strategy worked well enough when a customer was using products from a single vendor. However, the downside was that the customer was locked into a permanent relationship with that vendor, losing interoperability with products that relied on existing industry standards.

2.2 LINE INTERFACES
As mentioned earlier, there are two types of line interfaces for legacy PBX systems. These are trunk-line interfaces that connect the PBX to the CO exchange, and terminal-line interfaces that connect the PBX to telephone handsets.

2.2.1 TERMINAL LINE CARDS
Every telephone handset connects directly to at least one corresponding port on a line card, although multi-line handsets and attendant consoles (Direct Station Select/Busy Lamp Field or DSS/BLF) may use up additional line card ports. Terminal line cards fall into two categories – analog and digital – and each supports only the corresponding analog or proprietary digital handsets.

The type of telephone handsets provided typically depends on the user’s role and status within the organization. A manager might expect a full-featured phone. Department secretaries or administrative assistants often require specialized multi-line sets and a broader set of telephony features. In such cases, the telephony team is faced with increased cost and administrative issues.

2.2.2 TRUNK LINE INTERFACES
Trunk interfaces connect the PBX to the PSTN, enabling communication to the outside world. Trunks were the first part of the telephone network to adopt digital technology–deployments began in the early 1960s. Prior to this, telephone connections were entirely analog. Many of us use analog telephones at home or even at work, yet the phone systems to which we connect are almost exclusively digital, so it makes sense at this point to explain why we shifted to digital.

If you throw a pebble into a calm lake, it generates waves. These waves emanate out from the place where the pebble hit the water. As they travel further, the waves begin to flatten and attenuate. In a similar action, when we talk into a telephone handset, the microphone converts the sound waves generated by our vocal chords into electrical waves, which are transmitted down the line. Electrical waves behave like waves on the lake –as they move further away from the source, they flatten and attenuate, eventually becoming impossible to decipher. Early phone systems boosted (amplified) the signal, but this caused minor pops and crackles due to interference from other power sources, which were also amplified. In the days of analog telephony, long distance calls suffered from these hissing, crackling and popping sounds, which often made call quality very poor.

Enter digital telephony. When the electrical signal reaches the telephony exchange, it is sampled very quickly. Each sample is converted to a numerical value representing the frequency of the sound at the moment the sample was taken. This number is sent as a pattern of ones and zeros all the way through the network. If the signal becomes weak, then the ones and zeros simply regenerate along the way – without the hissing and crackling, of course.

Traditional digital trunks are often sold in terms of multiple channels, each with a capacity of 64 kilobits per second (kbps). These channels form the basis of the global telephony network– so where did the number 64 come from? The answer relates to the way we convert analog sound waves into the ones and zeros carried over the digital network. The frequency range for the human voice has a size of 4000 hertz (Hz). To render this into numerical values that can be converted back into something representative of the original sound waves, we need to sample the wave at twice the highest frequency value, i.e. 8,000 samples per second. Eight bits represents the frequency values numerically, so each time the sound wave is sampled, we use eight bits. To sample 8,000 times per second times eight bits per sample (8 x 8000), equals 64,000 bits per second: 64 kbps. Today's codecs can improve on these numbers, but 64 kbps is still found throughout the telephony network.

It is important to understand the needs of your organization and associated costs when selecting and ordering a trunk connection from your local telecommunication provider. Technology in this area moves very rapidly, so care should be taken not to sign up for long-term contracts that may lock you into outdated technology. Finally, IP trunks are now widely available commercially. They compete with traditional digital or analog connections, keeping the call on-net to the carrier, which then uses its own gateways to break out to the PSTN.

Depending on your business requirements, it may make sense to establish service level agreements as part of your service provider contract. These can include: time to respond, time to fix and latency over a data or VoIP link. Third-party applications and appliances can be used to independently gather statistics concerning availability and service quality.

Another factor is that signaling can be in-band (robbed bit) or out of band with the use of a separate, dedicated channel. In Table 1, the letter D stands for a dedicated signaling channel. The channels used to carry voice calls are known as bearer or B channels. So the formula 2B+D describes an integrated services digital network (ISDN) Basic Rate Interface (BRI) providing 2 x 64kbps channels.

BRI interfaces are still widely used outside North America, although DSL is increasingly being used to carry on-net VoIP using open protocols like the session initiation protocol (SIP). One thing to keep in mind about BRI is that for historical reasons, two interfaces are available. North America utilizes the U interface, which connects directly to the local exchange. In Europe, the

S/T interface connects the ISDN device behind a small network terminal owned that is operated by the service provider.

Trunk Type	Channels
Analog (FXO)	1
ISDN BRI	2 + D
T1	24
T1-PRI	23 + D
E1	32
E1 PRI	30 + D
DS3	672
SIP trunking	Bandwidth dependent

Table 1: Trunk Options

Trunks can be analog (like the foreign exchange office or FXO) or digital: T-1 with ISDN PRI; or E-1, which is used in Europe. The various options for each of these trunk types offer tradeoffs in terms of cost, capacity and features. Most vendors support the full range of trunk options available; so tradeoffs are based on cost and which features are required by the customer. These are discussed in the next section. The technical details of how PBX systems and central office exchanges initiate calls and present audio streams over trunks are beyond the scope of this guide.

2.2.3 TRUNK FEATURES

A common feature deployed by nearly all businesses is Caller Identification (Caller ID). This allows the called party to see the calling party’s name and telephone number before picking up the phone (unless the calling party has specifically blocked this feature). There are two Caller ID formats for delivering this information—Single Data Message Format (SDMF) and Multiple Data Message Format (MDMF). SDMF provides the calling number, while MDMF provides any combination of calling name and number. Note: If you are leveraging a complex call center application, be sure to work closely with your vendor to determine which other trunk features may be necessary.

Two additional mechanisms deliver caller ID:
1. Automatic Number Identification (ANI), similar to Caller Line Identification (CLI)
2. Dialed Number Identification Service (DNIS), an enhancement of 800-number services that enable the use of CLI intelligence for sophisticated routing of calls into the organization.

Another feature delivered by your telecommunication provider (telco) is used for inbound call routing. In North America, it is called direct inward dial (DID); in the U.K., it is DDI. This feature enables external callers to contact a user directly at his or her unique phone number, without intervention by an automated attendant or operator.

DID trunks are ordered in blocks consisting of 20 or more 10- digit telephone numbers. These numbers are assigned by the telco to each customer, and are routed to DID trunks connected to the PBX. When a call is made to a DID number, the telephone company sends the last three or four digits of the 10-digit number via the DID trunks at call set-up time. The PBX monitors for the digits and routes the calling party to the called party’s extension. “Wink start” is a mechanism for initiating an inbound call and passing the extension number to the PBX using a specific signal. Analog DID trunks are inbound only and cannot be configured as two-way trunks. Connecting PBX systems across the WAN or within the same office location can be accomplished using either T- 1 or analog interfaces. These interfaces were designed to interact with the telco's CO switches; therefore, one of the PBX systems must simulate CO signaling to enable the two PBXs to communicate effectively. Similar schemes are often used when configuring a gateway or IP telephony system to connect to a legacy PBX.

2.2.4 TRAFFIC CALCULATIONS

To decide the exact number of telephone lines and trunks your company requires, first determine the number of telephone users, calling traffic and acceptable percentage of call blocking (failure of calls completed due to an insufficient number of available trunks). A sample traffic calculator for determining the number of telephone lines and trunks can be found at www.erlang.com. If no data is available for determining your telephone line and trunk requirements, you can follow the recommendations given in Table 2.

In general, smaller installations require more trunks per telephones (typical configuration), whereas larger installations do not need as many trunks per telephones (light configuration).

Telephone Traffic	Trunks per Telephones	Trunk Factor
Heavy	3 trunks per 6 telephone users	3/6
Typical	2 trunks per 6 telephone users	2/6
Light	1 trunk per 6 telephone users	1/6

Table 2: Trunk Ratios

Note: These numbers are not applicable to call center implementations, which are much more intensive users of trunk capacity. In call centers, calls are often held in queue prior to passing them through to agents. Please consult your vendor for suggested ratios.