CHAPTER 5

VOICE COMMUNICATION CONCEPTS AND TECHNOLOGY

Concepts Reinforced

Top-down model
OSI Model
Protocols and interoperability

Concepts Introduced

Voice digitization
Voice compression
PBX functionality and architecture
Voice transmission alternatives
Data/voice integration
Voice network concepts
Computer telephony integration
Voice over the Internet

OBJECTIVES

After mastering the material in this chapter you should be able to:

• Understand the underlying technical concepts for voice transmission, voice
digitization, voice compression, and data/voice integration.

• Understand currently available voice-related technology including PBXs,
voice digitizers, and voice/data multiplexers and modems.

• Understand the functionality, standards, business impact, and technology
involved with computer telephony integration.

• Understand the functionality, concepts, standards, business impact, and
technology involved with voice network services, voice transmission alter-
 natives, and voice/data integration.

INTRODUCTION

Network analysts must be qualified to design networks that are capable of carrying
voice as well as data. Before designing such networks, it is essential for the network
analyst to understand the nature of voice signals, as well as how voice signals can be
processed and integrated into a cohesive network with data transmissions.
Once based exclusively on analog transmission, voice communication is rapidly becoming dependent on digital transmission technology. Once the voice signal has been digitized, a wide variety of transmission services can potentially be employed to complete the transmission of the voice signal to its designated destination. As is the case with any type of communications system involving the interoperability of multiple pieces of hardware and software technology, standards play an essential role in assuring end-to-end interoperability.

Although traditional telephony continues to be the bearer of most voice calls, alternatives such as voice over IP (VOIP) promise equal quality at significantly lower cost. Technologies such as H.323 and SIP, combined with broadband Internet technologies, have the potential to replace the modern telephone network architecture and revolutionize how calls are placed and billed.

One of the fastest-growing areas of telecommunications is wireless telephony. With the development of higher capacity, better-sounding digital networks cellular phone use has grown exponentially in the United States and worldwide. In addition to voice communication, these networks are capable of carrying digital data to a spate of new handheld devices. As these wireless digital networks continue to evolve, new applications that take advantage of their better interoperability and higher data speeds will continue to be developed.

A private branch exchange (PBX) acts as a local phone switch for an organization. The PBX allows for calls to be made inside the organization without using the PSTN, while allowing for calls to be made between the internal extensions and the PSTN. Newer PBX features allow for the integration of VOIP and wireless technologies into the PBX.

### VOICE TRANSMISSION BASIC CONCEPTS

The modern telephone system is commonly known as the public switched telephone network, or PSTN. A voice conversation consists of sound waves that are of varying frequency and amplitude and are represented as a continuously varying analog waveform. The POTS (plain old telephone service) network employed analog transmission methodologies to transmit the voice signals from source to destination.

But how does this analog waveform get from a person's mouth, the human transmitter, onto the PSTN and subsequently into the ear, the human receiver, of the person who was called? Figure 5-1 illustrates the mechanics of a typical phone handset, which consists of both transmitter and receiver components.

The telephone handset, consisting of both a transmitter and receiver, is really a fairly simple device that works largely based on the properties of electromagnetism. The transmitter, or mouthpiece, contains a movable diaphragm that is sensitive to changes in voice frequency and amplitude. The diaphragm contains carbon granules that have the ability to conduct electricity. Because the human voice spoken into the transmitter varies, the amount of carbon granules striking the electrical contacts in the mouthpiece varies, sending a varying analog, electrical signal out onto the voice network.

This constantly varying analog electrical wave is transmitted over the voice network to the phone of the receiving person. The receiver or earpiece portion of the handset basically works in the opposite fashion of the mouthpiece. The varying electrical waves produced by the transmitter are received at the receiver at an electromagnet. Varying levels of electricity produce varying levels of magnetism that, in
Voice Transmission Basic Concepts

The moving diaphragm produces varying sound waves that correspond to the sound waves that were input at the transmitter. The electromagnetically reproduced sound produced at the receiver resembles the actual sound waves input at the transmitter closely enough to allow for voice recognition by the receiving party.

**Voice Bandwidth**

Although the approximate range of hearing of the human ear is between 15,000 to 20,000 Hz, significantly less bandwidth is used to transmit the electromagnetic representations of analog voice signals over the analog PSTN. POTS uses a bandwidth of 4,000 Hz including two guardbands to prevent interference from adjacent frequencies from interfering with the voice signal. As a result, the usable bandwidth on the local loop circuit connecting an individual’s home or business to the phone company’s central office for dial-up analog voice transmission is 3,000 Hz, from 300 to 3,300 Hz.

*Figure 5-1* Getting Voice onto and off of the Network

- **Electromagnet**
- **Speaker diaphragm (movable)**
- **Receiver (earpiece)**
- **Permanent magnet**
- **Variable magnetic field**
- **Electrical contacts**
- **Diaphragm (movable)**
- **Sound Waves**
- **Transmitter (mouthpiece)**
- **Granulated carbon**
- **Handset**
- **RJ-22 connector**
- **RJ-11 connectors**
- **2 wires**
Figure 5-2 illustrates the comparative bandwidths of human speech and the analog phone network. This limited bandwidth is why people sound less lifelike on the telephone than in person.

**VOICE NETWORK CONCEPTS**

Telephone calls are connected from source via circuit switching. Circuit switching is an analog telecommunications term that originally meant that a physical electrical circuit was created from the source telephone handset to the destination telephone handset. In the early days of the telephone system a telephone system operator manually connected these connections at a switchboard. Later the rotary telephone and automatic switching was introduced.

A better definition of a switched connection in the modern phone system is a reserved bandwidth connection between two telephone handsets. Although there is no longer a physical circuit in place between the handsets, the capacity on the telephone network required to deliver the call is reserved for the exclusive use of the call: The same amount of telephone system capacity is used by two people who are being perfectly quiet as is used by two people who are talking at the same time. The capacity is dedicated to the call as soon as it is placed.

**Basic Telecommunications Infrastructure**

Figure 5-3 illustrates the major components of the PSTN. The circuits between a residence or business and the local central office or CO are known as local loops. A central office is a facility belonging to the local phone company in which calls are switched to their proper destination. As covered in chapter 1, the LATA is an area
within which the local carrier completed all of the calls during the time period between the break-up of AT&T and the Telecommunications Act of 1996.

Telephone calls are established by a device located at the local telephone companies CO known as a telephone switch. The telephone switch is directly connected to the customer’s telephone handset via the local loop. The telephone switch routes calls to the destination telephone handset. Requested destinations for phone calls are indicated to the telephone switch by dialing a series of numbers. These numbers tell the telephone switch whether the call will be local, intra-LATA, or inter-LATA, and subsequently, which circuits must be accessed and combined to complete the call as requested.

All voice traffic destined for locations outside of the local LATA, and some traffic within the LATA, must be handed off to the long distance or inter-exchange carrier (IXC) of the customer’s choice. Competing long-distance carriers wishing to do business in a given LATA maintain a switching office in that LATA known as a POP, or point of presence. This POP handles billing information and routes the call over the long-distance carrier’s switched network to its POP in the destination’s LATA. The circuit between the POPs may be via satellite, microwave, fiber-optic cable, traditional wiring, or some combination of these media. Depending on traffic levels on the long-distance carrier’s network, calls may be routed through any combination of switches before reaching their final destination.

In the basic infrastructure illustrated in Figure 5-3, the only analog links in the PSTN are the local loops running from the end points to the central offices. Once the voice signal hits the central office it is converted into a digital signal for transmission across the PSTN to the destination central office, where it is converted back into an analog signal for transmission across the local loop to the destination telephone. The processes used to convert the phone call from analog to digital signals are covered later in this chapter.

Because these local loops are the sole remaining analog links in the modern PSTN and cover relatively short distances, they are commonly referred to as the last mile in the telephone system. Throughout the next few chapters you will see how the analog local loops pose a serious limitation in terms of the rate of data transmission across the PSTN.
**PSTN Network Hierarchy**

As can be seen in Figure 5-4, a residential or business call is first processed in the local central office, also known as an end office or local office. In terms of the network hierarchy, an end office is known as a **Class 5 office**. This local central office contains a switch that processes incoming calls, determines the best path to the call destination, and establishes the circuit connection.

Local calls come into the local central office via a local loop and travel to their local destination via a local loop. If a call is destined to another telephone on the

![Figure 5-4 Representative Voice Network Hierarchy](image-url)
same telephone switch, the call is switched at the CO to the destination local loop. Calls that are not local but are still within the same LATA are known as intra-LATA calls and are handled by the caller’s selected intra-LATA carrier, most often an RBOC. Technically, these are long-distance calls, and a local CO may not have a direct trunk to the destination CO. In this case, the call is routed through a tandem office that establishes the intra-LATA circuit and also handles billing procedures for the long-distance call.

If the call is bound for a destination in another LATA, it must be turned over from the local carrier to a long-distance carrier such as AT&T, MCI, or Sprint. In most cases, the Inter-Exchange Carrier (IXC) will have been chosen by individual residential and business subscribers. The local CO still receives such inter-LATA calls from subscribers. However, rather than routing the call itself, the CO merely forwards the call to the local Point of Presence of the long-distance carrier of choice.

Such a long-distance switching office is also known as a POP, or a Class 4 toll center. The term toll center implies that long-distance billing calculation as well as switching activities are performed at these locations. A given local CO might have trunks to more than one toll center. As will be seen, circuit redundancy offering multiple alternative paths for call routing is a central premise of the voice network hierarchy. If the local toll center can find adequate space on a trunk headed to the destination CO, then the connection between source and destination COs is completed. If no paths to the destination are directly available to the Local toll center, then the call is escalated up the network hierarchy to the next level of switching office. The overall desire is to keep the call as low on the hierarchy as possible. This provides both quicker call completion for the subscriber as well as maximization of the cost-effective use of the lowest and least-expensive switching offices possible.

Higher levels on the network hierarchy imply greater switching and transmission capacity, as well as greater expense. When calls cannot be completed directly, Class 4 toll centers turn to Class 3 Primary Centers that subsequently turn to Class 2 Sectional Centers that turn finally to Class 1 Regional Centers. These categories of switching and transmission centers were originally AT&T’s, and not all inter-LATA or long distance carriers have such a five-level network hierarchy. However, the five-level hierarchy has become industry standard terminology.

### Telephone Number Plans

Telephone numbers are a hierarchical address method. U.S. telephone numbers can be broken into three basic parts: a three-digit area code, a three-digit exchange, and a four-digit subscriber number. To make a telephone call, at a minimum the exchange plus the subscriber number must be dialed. If the call is to a destination phone outside of the source phone’s area code, the destination area code must be dialed as well.

Originally assigned to geographic areas, area codes are the top level of the hierarchy. When the U.S. telephone system was originally designed, all area codes had either a zero or a one as the center digit. Conversely, exchanges were not allowed to have a zero or a one as the center digit. This technique allowed the telephone switches at the central office to easily distinguish between area codes and exchanges. Because of this differentiation, long-distance calls placed within an area code could simply be dialed by adding a one to the beginning of the local telephone number.

This system worked well until the number of area codes increased to the point that all of the area codes with ones and zeros in the center were in use. At that point,
there was no choice but to add an area code with a different number as the center digit. The first area code placed in service that did not have a one or zero as the center digit was the 770 area code that serves the area outside of Atlanta, Georgia.

The implementation of these nonstandard area codes originally caused some problems. Telephone switches and PBXs had to be reprogrammed to support the new number scheme. When the 770 area code first went into use, several areas could not place a call to it. This new area code number scheme created another problem. Because the telephone switch cannot tell the difference between an exchange and an area code based on number structure, it is now usually necessary to dial all 10 digits when placing a long-distance call, even if the call is within the same area code.

Although the area code + exchange + subscriber number system is used in the United States, a broader ranging system is used internationally. To place an international call you must dial 011 + country code + city code + number. The numbering systems for country codes, city codes, and local numbers can vary between countries. It is important to carefully research the dialing pattern before placing an international call.

TELEPHONE NUMBER SHORTAGES AND NEW AREA CODES

The need for new area codes is driven from the increasing need for new telephone numbers. The proliferation of fax machines, cellular phones, pagers, and second lines for use to connect to the Internet has caused an exponential increase in the number of telephone numbers currently in use. Each exchange supports 10,000 telephone numbers. Each area code supports 1,000 exchanges, or 10 million separate telephone numbers. When the required amount of telephone numbers exceeds the available capacity, there is no choice but to add another area code to the geographic area served by the original area code.

There are two basic approaches to adding area codes. The geographic area served by the original area code can be broken into two smaller sections. One section would retain the original area code while the new section would be given a new area code. Although this approach is true to the original concept of an area code, the area code for half of the telephones in the old area code is changed. This change affects the calling patterns of every subscriber in the area code and of everyone who needs to place a long distance call to them. Each business that experiences an area code change must replace all of its business cards, letterhead, and any other items that contain its telephone number; often at a significant expense.

To resolve these issues, an alternate concept known as overlaying is becoming common. In an overlay solution, a new area code is added to the original area code’s geographic area. New telephone numbers are simply assigned to this new area code. In this scenario, no existing customers are forced to change their area code. However, it is possible that your neighbor might be in a different area code than you. In this case you would simply dial the full 10-digit telephone number to place a local call to your neighbor.

System Signaling

In addition to carrying the actual voice signals, the telephone system must carry information about the call itself. This information is commonly referred to as system signaling, or inter-office signaling. At a bare minimum, system signaling
needs to provide a means of accomplishing call set-up and call termination. Advanced functions are also available, including call waiting, caller ID, and three-way calling. Each of these functions requires the source telephone set to send data to the local phone switch or for the local phone switch to send data to the destination phone switch and telephone set in addition to the basic voice data transmission. There are two basic approaches to sending system signaling data across the PSTN: in band and out of band.

**In-Band Signaling**  In an in-band signaling system the signals are sent on the same channel as the voice data itself. This is the method used to send signals across the analog local loop for most home telephones. When you pick up the phone you listen for a **dial tone** to make sure the telephone switch at the CO is ready to serve you. At that point you dial (sending the phone number across in the voice bandwidth) and listen for the phone to ring. If the called party answers the phone, the remote telephone switch comes off the hook and the connection is established.

The destination telephone number can be communicated to the telephone switch in two ways. Older style rotary phones, like the one which was taken apart in order to draw Figure 4-1, have a round dial that causes a certain number of pulses of electricity to be generated, depending on the number dialed. Dialing a “1” produces one electrical pulse, dialing a “2” produces two electrical pulses, and so on. These pulses were used to physically operate relays in the first automatic phone switches. Modern phone switches no longer use mechanical relays and therefore do not require electrical pulses to indicate the destination telephone number.

Many of today’s phones no longer have rotary dials on them. Instead, they contain twelve buttons that correspond to the ten numbers on the rotary dial plus two characters, the star (*) and the octothorpe (#—also known as the pound key in the US). A switch is often included that can be set to have the telephone set issue a series of **pulses** to emulate the dialing process of the older style of phone for areas where central office switches have not yet been upgraded to understand **touch-tone** dialing.

**TOUCH-TONE DIALING**

Touch-tone dialing is technically called **DTMF**, or **dual tone multi-frequency**, because the tone associated with each number dialed is really a combination of two tones selected from a matrix of multiple possible frequencies. Figure 5-5 illustrates the numbers and symbols found on a typical telephone touch panel and their associated dual tone frequencies. The two keys on either side of the 0 were officially named star (*) and octothorpe (#) by Bell Labs, although they have different commonly used names in different languages.

The tones generated by DTMF phones can be used for much more than merely dialing destination telephone numbers. As will be seen later in the chapter, these same tones can be used to enable specialized services from PBXs, carriers, banks, information services, and retail establishments.

When you decide to terminate the call, you hang up or change the status of your local loop to on-hook. At that time the local telephone switch changes from off-hook to on-hook and the telephone switch knows you are off the phone and are ready to receive a telephone call. Call waiting, three-way calling, and caller ID all use similar in-band means to communicate information between the telephone handset and the local telephone switch.
Out-of-Band Signaling

While in-band signaling works well for communication between telephone handsets and the local telephone switch across the analog local loop, the inter-switch connections on the digital PSTN make use of a separate channel to carry system signaling data. This **out-of-band signaling** approach provides a means to manage the network itself by handling the routing of calls and circuit establishment as well as the monitoring of circuit status and notification and re-routing in the case of alarms or circuit problems. By moving the call setup and management data to a separate network it becomes easier to support transparent operation between different digital encoding mechanisms. The management data is readily available to each piece of telephone network equipment regardless of the encoding mechanism used.

The worldwide, CCITT approved standard for out-of-band signaling is known as **Signaling System 7 (SS7)**. SS7 controls the structure and transmission of both circuit-related and non-circuit related information via out-of-band signaling between central office switches. SS7 delivers the out-of-band signaling via a packet switched network physically separate from the circuit switched network that carries the actual voice traffic. Each node on the PSTN must connect to both the voice network and the SS7 network.

The SS7 network is really nothing more than a packet-switched network not unlike other suites of protocols that will be examined in chapters 5 to 7. Like most protocol suites, SS7 can be modeled in comparison to the OSI 7 Layer Reference Model. Figure 5-6 summarizes the major characteristics of the Signaling System 7 Protocols as well as comparing the SS7 Protocol Suite to the OSI Model.
Signaling System 7 and the intelligent services that it enables are often described as part of an all-encompassing interface between users and the PSTN known as the AIN or advanced intelligent network. The AIN is sometimes simply referred to as the intelligent network (IN).

The AIN provides a great deal of flexibility to customers. The use of AIN services provides a means of achieving many different applications:

- **Alternate billing service (ABS)**—This service allows a long-distance call to be billed to a calling card, to a third party, or to the receiver (collect call).
- **Custom local area signaling service (CLASS)**—A group of services that allows many services local access to the customer’s telephone. Examples include call waiting, call forwarding, call blocking, caller ID blocking, busy number redial, and automatic redial of missed calls.
- **Enhanced 800 service**—This service allows 800-number portability. Originally, 800 numbers were tied to a specific area code and long-distance provider. This service resolves those limitations.
- **Intelligent call processing (ICP)**—Using this service customers are able to reroute incoming 800 calls among multiple customer service centers in a matter of seconds. This rerouting is done completely transparent to the calling customer. ICP allows multiple call centers geographically dispersed throughout the country to function as one logical call center, with the overall number of incoming calls distributed in a balanced manner across all centers.

User-oriented network services such as the AIN are being offered in response to user demands for in-house control over a key element of their business: their

---

**Figure 5-6  Signaling System 7 Protocols and the OSI Model**

<table>
<thead>
<tr>
<th>OSI Model</th>
<th>Signaling System 7</th>
</tr>
</thead>
<tbody>
<tr>
<td>Application</td>
<td>O&amp;MAP</td>
</tr>
<tr>
<td>Presentation</td>
<td></td>
</tr>
<tr>
<td>Session</td>
<td></td>
</tr>
<tr>
<td>Transport</td>
<td>SSCP</td>
</tr>
<tr>
<td>Network</td>
<td></td>
</tr>
<tr>
<td>Datalink</td>
<td></td>
</tr>
<tr>
<td>Physical</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Protocol Name</th>
<th>Description/Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>Operations Maintenance Application Part (O&amp;MAP)</td>
<td>O&amp;MAP provides standards for routing and management of messages related to network operations and maintenance</td>
</tr>
<tr>
<td>Transaction Capabilities Application Part (TCAP)</td>
<td>TCAP provides standards for routing and management of noncircuit related information for transaction processing applications requiring out-of-band signaling</td>
</tr>
<tr>
<td>Signaling Connection Control Part (SCCP)</td>
<td>SCCP provides standards for routing and management of signaling messages. Not related to call set-up between switches. A connection-oriented service providing reliable message delivery</td>
</tr>
<tr>
<td>Message Transfer Part (MTP)</td>
<td>MTP provides standards for routing of signaling messages between switches. A connectionless, datagram service</td>
</tr>
<tr>
<td>Network Service Part (NSP)</td>
<td>Another term for the combination of the SCCP and the MTP3</td>
</tr>
</tbody>
</table>
telecommunications systems and the links from those systems to the wide area PSTN. Catalog sales organizations are literally out of business without their phones and must have contingency plans in place in order to deal with and perhaps avoid possible catastrophes.

**Voice Digitization**

The analog POTS system has largely been supplanted in the modern telephone system by a combination of analog and digital transmission technologies. While analog signaling is effective, it is limited in terms of quality, distance, and capacity. The longer the signal has to travel, the poorer the quality. There are also significant capacity issues associated with analog transmission. In general, only one voice conversation can be carried on a single set of wires using analog transmission. Although it is possible to partially overcome this limitation through the use of multiplexing, as illustrated in detail in chapter 2, digital transmission offers better quality and higher capacity than analog transmission over a given media. The modern voice network is almost entirely digital in nature.

Although the local loop between the local central office and a residence or place of business might be an analog circuit, it is highly unlikely that the continuously varying analog signal representing a person’s voice will stay in analog form all the way to the destination location’s phone receiver. Rather, it is very likely that high-capacity digital circuits will be employed to transport that call, especially between COs or carriers. The fact that carriers might be converting a voice conversation to digital format and converting it back to analog form before it reaches its destination is completely transparent to phone network users.

The basic technique for **voice digitization** is relatively simple. The constantly varying analog voice conversation must be sampled frequently enough so that when the digitized version of the voice is converted back to an analog signal, the resultant conversation resembles the voice of the call initiator. Most voice digitization techniques employ a sampling rate of 8,000 samples per second.

Recalling that a digital signal is just a discrete electrical voltage, there are only a limited number of ways in which the electrical pulses can be varied to represent varying characteristics of an analog voice signal:

- **Pulse amplitude modulation**, or PAM, varies the amplitude or voltage of the electrical pulses in relation to the varying characteristics of the voice signal. PAM was the voice digitization technique used in some earlier PBXs.
- **Pulse duration modulation** (PDM), otherwise known as **Pulse Width Modulation** (PWM), varies the duration of each electrical pulse in relation the variances in the analog signal.
- **Pulse position modulation** (PPM), varies the duration between pulses in relation to variances in the analog signal. By varying the spaces in between the discrete electrical pulses on the digital circuit, PPM focuses on the relative position of the pulses to one another as a means of representing the continuously varying analog signal. Figure 5-7 illustrates these three voice digitization techniques.

**Pulse Code Modulation (PCM)**  Although any of the above methods may be used for voice digitization, the most common voice digitization technique in use today is known as **Pulse Code Modulation** or PCM. Figure 5-8 illustrates the basics of PCM.
As can be seen from Figure 5-8, 8 bits or 1 byte are required in order to transmit the sampled amplitude of an analog signal. Since an 8-bit code allows $2^8$ or 256 different possible values, each time the actual analog wave is sampled, it is assigned a value from 0 to 255, dependent on its location or amplitude at the instant it is sampled. Some simple mathematics will reveal the bandwidth required to transmit digitized voice using PCM. This computed required bandwidth will, by no coincidence, correspond exactly to a very common digital circuit bandwidth.

The device that samples the analog POTS transmission coming in from the local loop and transforms it into a stream of binary digits using PCM is known as a coder/decoder or codec. As mentioned in the following section, each codec outputs a digital signal at a data rate of 64 Kbps. This data rate, known as a DS-0, is the basic unit of voice data transmission in a PCM based telephone system. All higher speed
voice connections will operate at some multiple of this DS-0 speed. Codecs are usu-
ally deployed as part of a channel bank. A channel bank is a hybrid device consisting
of 24 codecs and the circuitry required to place the digitized PCM voice signals onto
a T-1 circuit. Codecs and channel banks may be integrated into telephone switches or
purchased separately.

**VOICE DIGITIZATION BANDWIDTH REQUIREMENTS**

Since 8,000 samples per second are required to assure quality transmission of digi-
tized voice and each sample requires 8 bits to represent that sampled bandwidth in
binary (ones and zeroes) notation, the following equation reveals that 64,000 bits per
sec is the required bandwidth for transmission of voice digitized via PCM. A DS-0
circuit has a transmission capacity of exactly 64 Kbps. Twenty-four DS-0s are com-
bined to form a T-1, yielding the fact that a T-1 can carry 24 simultaneous voice con-
versations digitized via PCM. Following is the mathematical proof:
8,000 samples/sec × 8 bits/sample = 64,000 bits/sec (bps)

64,000 bits/sec = 64 Kbps = DS-0 Circuit

24 DS-0s = 24 × 64 Kbps = 1536 Kbps = 1.536 Mbps

Plus: 1 framing bit/sample × 8,000 samples/sec. = 8,000 framing bits/sec

8 Kbps + 1536 Kbps = 1544 Kbps = 1.544 Mbps = Transmission capacity of T-1 circuit

It is important to note that the maximum data-carrying capacity of a T-1 circuit is only 1.536 Mbps because the framing bits cannot be used to carry data.

**ADPCM** A variation of this digitization technique known as *adaptive differential pulse code modulation*, or ADPCM, is most commonly used in Europe. ADPCM is a CCITT (ITU) standard that takes a slightly different approach to coding sampled amplitudes in order to use transmission bandwidth more efficiently; ADPCM requires roughly half the bandwidth for each digitized conversation as compared to PCM. By transmitting only the approximate difference or change in amplitude of consecutive amplitude samples, rather than the absolute amplitude, only 32 Kbps of bandwidth is required for each conversation digitized via ADPCM.

Using a specialized circuit known as an adaptive predictor, ADPCM calculates the difference between the predicted and actual incoming signals and specifies that difference as one of sixteen different levels using 4 bits \(2^4 = 16\). Since each voice channel can be represented by just 4 bits, ADPCM can support 48 simultaneous voice conversations over a T-1 circuit.

The ITU standard for 32 Kbps ADPCM is known as G.721 and is generally used as a reference point for the quality of voice transmission known as *toll quality*. G.721 has been superseded by other ADPCM standards that use less than 32Kbps per voice channel. For example, G.723 defines ADPCM for 24 Kbps and 40 Kbps while G.726 defines ADPCM for 40, 32, 24, and 16 Kbps.

Fortunately voice signals can be readily converted between any of these digital formats enabling transparent telephone conversations regardless of the voice digitization technique used at either end of the call.

**Voice Compression**

ADPCM is also known as a *voice compression* technique because of its ability to transmit twenty-four digitized voice conversations in half of the bandwidth required by PCM. Other more advanced techniques employ specially programmed microprocessors known as *digital signal processors* that take the digitized PCM code and further manipulate and compress it. In doing so, DSPs are able to transmit and reconstruct digitized voice conversations in as little as 4,800 bps per conversation, an increase in transmission efficiency of more than thirteen times over PCM!

Numerous voice compression technological approaches exist. Voice compression can be performed by standalone units or by integral modules within multiplexers. The particular method by which the voice is compressed may be according to an open standard or by a proprietary methodology. Proprietary methods require that a given vendor’s equipment must be present on both ends of the voice circuit in question. Each voice compression technique seeks to reduce the amount of transmitted voice information in one way or another.
Some voice compression techniques attempt to synthesize the human voice, other techniques attempt to predict the actual voice transmission patterns, while still others attempt to transmit only changes in voice patterns. Regardless of the voice compression technique employed, one thing is certain. The quality of compressed voice transmissions does not match the quality of an analog voice transmission over an analog dial-up line or a PCM-digitized voice transmission using a full 64 Kbps of digital bandwidth. The transmission-quality degradation will vary from one instance to another. However, only the end users of the compressed voice system can determine whether the reduced voice quality is worth the bandwidth and related cost savings.

## VOICE TRANSMISSION ALTERNATIVES

Although the PSTN has traditionally been seen as the cheapest and most effective way to transmit voice, alternative methods for voice transmission do exist. Several such methods are briefly explored in terms of configuration requirements, advantages, and disadvantages.

### Voice over IP (VOIP)

Although this alternative voice transmission methodology is also commonly referred to as **voice over the Internet**, it is actually the underlying transport protocols of the Internet that deliver the voice conversations. Voice over IP refers to any technology used to transmit voice over any network running the IP protocol. The important point about VOIP technologies is that they are not exclusively confined to use on the Internet. They can be used just as effectively in any of the following topologies:

- Modem-based point-to-point connections
- Local area networks
- Private networks, also known as **intranets**

VOIP can be successfully deployed in any of the previously mentioned topologies provided that required technology is properly implemented. Figure 5-9 illustrates both the technology required to implement IP-based voice transmission as well as the alternative topologies possible.

Required client hardware and software technology for VOIP transmission includes the following:

- VOIP client software
- PC workstation with sufficiently fast CPU to digitize and compress the analog voice signal
- Sound card for local playback of received voice transmission
- Microphone for local input of transmitted voice signals
- Speakers for local output of received voice signals

Figure 5-10 summarizes many of the key features and functionality of VOIP client software.
VoIP Alternatives to PCs: Wired Handsets and Wireless Handsets

In addition to personal computers network manufactures are also selling wired and wireless handsets that support natively support VOIP technologies. These handsets look like a traditional telephone handset, but instead of having a local loop connection they have an Ethernet or a wireless LAN connection. Because they contain microprocessors, these IP handsets offer additional features not found on traditional phones, such as customizable ring...
tones and specialized phone applications that can be written to connect to data residing on the network. Currently the costs of these new handsets are more expensive than a traditional telephone handset, but prices are steadily coming down. With the potential savings on long-distance toll calls associated with VOIP technologies, the additional cost might be easy to justify in many environments.

In a sign of the impending convergence in the telecommunication marketplace, some vendors are developing wireless handsets that support VOIP over wireless LANS along with digital cellular technologies such as those detailed later in this

<table>
<thead>
<tr>
<th>Feature</th>
<th>Importance/Implication</th>
</tr>
</thead>
<tbody>
<tr>
<td>Client platform support</td>
<td>Most IP-based voice transmission software supports Windows, with fewer packages supporting UNIX and Macintosh operating systems.</td>
</tr>
<tr>
<td>Interoperability</td>
<td>The ITU H.323 standard for interoperability among client software for low bandwidth audio (voice) and video conferencing is supported by some, but not all, client IP-based voice transmission software.</td>
</tr>
<tr>
<td>Transmission quality</td>
<td>Although transmission quality has improved thanks to improved voice compression algorithms, the fact remains that shared IP networks were designed to carry data that could tolerate delays. Voice networks are designed with dedicated circuits offering guaranteed bandwidth and delivery times to voice transmissions.</td>
</tr>
<tr>
<td>Multipoint audioconferences</td>
<td>Some packages may employ proprietary methods while others may support the T.120 conferencing standard.</td>
</tr>
<tr>
<td>Addressing for call creation</td>
<td>IP-based software packages employ a variety of different addressing techniques to create calls. In some cases, a directory server must be established listing all potential voice call destinations. In other cases, e-mail addresses or IP addresses may be used to initiate and IP-based voice call. Third-party directory services may also be supported.</td>
</tr>
<tr>
<td>Bandwidth reservation on networks</td>
<td>In order to more closely emulate the dedicated bandwidth circuits of the PSTN, an IP-based protocol known as RSVP (Resource Reservation Protocol) enables routing software to reserve a portion of network bandwidth known as a virtual circuit. This dedicated, guaranteed bandwidth is assigned to a particular IP-based voice transmission session, thereby minimizing transmission delay and increasing the quality of the transmission. Other quality of service (QoS) options are also available. Different hardware devices implement QoS in different ways, so you will need to consult the manufacturer for the best options. Some of the most popular options are Priority Queuing, Custom Queuing, VLANs, and Weighted Random Early Discard.</td>
</tr>
<tr>
<td>Voice compression</td>
<td>Depending on the particular codec algorithm used, voice compression can cause a major difference in required bandwidth. Among the more popular codec standards are high bandwidth GSM (Global Systems Mobile Communication) that uses 9600–to 11,000 bps and low-bandwidth RT24, 2400 bps.</td>
</tr>
<tr>
<td>Auxiliary features</td>
<td>Many IP-based voice packages support a variety of other functions that may be important to some organizations. Examples of such functions include: answering machine/recorded message capabilities, online rolodex with photographs of called parties, text-chat when sufficient voice quality cannot be maintained, electronic whiteboard for long distance brainstorming, file transfer and application sharing, incorporation of voice transmission into HTML documents for Web pages, API to integrate voice transmission in customized applications.</td>
</tr>
</tbody>
</table>

*Figure 5-10* Features and Functionality of VOIP Software
chapter. These phones will connect via the wireless LAN when in a building, then automatically switch to the digital cellular carrier when outside the wireless LAN coverage zone. In this manner, calls would be routed across the most cost-effective network available at any given time.

SIP: A Replacement for H.323 Signaling  
Session Initiation Protocol (SIP) is a signaling protocol for Internet conferencing, telephony, presence, events notification and instant messaging developed by the IETF to be the standardized mechanism to send multimedia over the Internet. Although H.323 was the original protocol used for VOIP, SIP is gaining momentum as the communication protocol of choice for VOIP implementations. Many vendors such as Cisco and Microsoft, as well as carriers like MCI, have embraced SIP as the communication protocol of choice for VOIP applications.

Another emerging application for SIP is in third-generation cellular networks. The 3GPP (3rd Generation Partnership Project) has chosen SIP as the signaling protocol of choice for these emerging networks. For more information on 3G wireless, see the Digital Cellular section later on in this chapter.

Voice Gateways: More than just a PBX  
As VOIP has matured, so have the devices that support it. VOIP has evolved from being just a “cool application” used by two or more computers on the Internet into a full-fledged alternative to traditional telephony. However, most organizations are not going to be able to simply replace all of their existing telephone technology overnight with a VOIP solution. Instead, mechanisms must be developed to integrate newer VOIP technologies into existing telecommunications infrastructures. For organizations that have an existing PBX, the preferred solution is to install a VOIP card that enables them to support VOIP software and handsets. Such an “IP enabled” PBX is also referred to as a voice gateway.

A voice gateway provides a means for the VOIP-enabled devices to access the traditional telephone network. The gateway can be set up to assign telephone numbers to IP devices so they can make traditional phone calls over the existing telephone network, have access to voicemail services, and other traditional phone features. By adding IP to the PBX, vendors can extend the use of the PBX into the new VOIP paradigm. For those organizations that do not have a PBX vendors such as Cisco Systems have introduced voice gateways that provide similar functionality. These devices have connections to both the data network and to the PSTN to route outgoing and incoming calls.

COST/BENEFIT ANALYSIS FOR IP-BASED VOICE TRANSMISSION  
Figure 5-11 identifies many of the potential costs associated with implementing an IP-based voice transmission network. All cost categories listed will not apply in all situations. Benefits will be most significant for those organizations with large domestic or international long distance calling expenses. However, it should be noted that such organizations often already have large-volume discounted rate contracts with their phone service providers that minimize or negate any potential savings that might be achieved by shifting to an IP-based voice transmission network.

A more subjective criteria that must be considered is the minimum acceptable transmitted voice quality. Higher transmission quality demands higher amounts of dedicated bandwidth. Lower amounts of shared bandwidth can cause transmission delays that will be manifested as voice drop-outs or clipped words.
Typical delays on voice transmission networks such as PSTN are in the 50–70 milliseconds range, while IP-based voice transmission networks can exhibit delays of 500 milliseconds to 1.5 seconds. Many corporations may conclude that the current generation of IP-based voice transmission technology is sufficient for internal corporate communication but is unacceptable for external communication with clients and customers.

Finally, should large amounts of revenue begin to bypass phone carriers as a result of a massive use of IP-based voice transmission over the Internet, it is likely that the Federal Communications Commission might take steps to assure that Internet Service Providers (ISP) do not have an unfair competitive advantage.

**Voice over Frame Relay**

IP-based voice transmission via the Internet is not the only alternative to traditional voice transmission over wide areas. Frame relay is another wide area transmission
services that was primarily or initially deployed for data transmission but is now capable of delivering voice transmissions as well. Although Frame Relay will be discussed further in chapter 6, the implications of transmitting voice over this service will be detailed here.

In order to be able to dynamically adapt to transmit data as efficiently as possible, frame relay encapsulates segments of a data transfer session into variable length frames. For longer data transfers, longer frames with larger data payloads are used, and for short messages, shorter frames are used. These variable-length frames introduce varying amounts of delay due to processing by intermediate switches on the frame relay network. This variable-length delay introduced by the variable-length frames works very well for data but is unacceptable to voice payloads that are very sensitive to delay.

The FRAD or frame relay access device is able to accommodate both voice and data traffic by employing any or all of the following techniques:

- **Voice prioritization**—FRADs are able to distinguish between voice and data traffic and prioritize voice traffic over data traffic.
- **Data frame size limitation**—Long data frames must be segmented into multiple smaller frames so that pending voice traffic can have priority access. However, data must not be delayed to unacceptable levels.
- **Separate voice and data queues**—In order to more effectively manage pending data and voice messages, separate queues for data and voice messages can be maintained within the FRAD.

Voice conversations transmitted over Frame Relay networks require 4 to 16 Kbps of bandwidth each. This dedicated bandwidth is reserved as an end-to-end connection through the frame relay network known as a PVC, or permanent virtual circuit. In order for prioritization schemes established by FRADs to be maintained throughout a voice conversation’s end-to-end journey, intermediate frame relay switches within the frame relay network must support the same prioritization schemes. At this point, voice conversations can only take place between locations connected directly to a frame relay network. There are currently no interoperability standards or network-to-network interface standards defined between frame relay networks and the voice-based PSTN. Figure 5-12 illustrates voice transmission over a frame relay network.

**Voice over ATM**

Whereas frame relay is a switch-based WAN service using variable length frames, ATM (asynchronous transfer mode) is a switch-based WAN service using fixed-length frames, more properly referred to as cells. Fixed-length cells assure fixed-length processing time by ATM switches, thereby enabling predictable, rather than variable, delay and delivery time. Voice is currently transmitted across ATM networks using a bandwidth reservation scheme known as CBR, or constant bit rate, which is analogous to a Frame Relay virtual circuit. However, constant bit rate does not make optimal use of available bandwidth because, during the course of a given voice conversation, moments of silence intermingle with periods of conversation. The most common method for currently transmitting voice over an ATM network is
to reserve a CBR of 64 Kbps for one voice conversation digitized via PCM (pulse code modulation).

**OPTIMIZING VOICE OVER ATM**

More efficient use of ATM network capacity for voice transmission can be achieved in one of the following ways:

- **Voice compression**—The ITU standardized voice compression algorithms via the G series of standards. Algorithms vary in the amount of bandwidth required to transmit toll-quality voice (G.726: 48, 32, 24, or 16 Kbps; G.728: 16 Kbps; G.729: 8 Kbps). An important point to remember with voice compression is that the greater the compression ratio achieved, the more complicated and processing-intensive the compression process. In such cases, the greatest delay is introduced by the voice compression algorithm with the highest compression ratio, requiring the least bandwidth.

- **Silence suppression**—All cells are examined as to contents. Any voice cell that contains silence is not allowed to enter the ATM network. At the destination end, the nontransmitted silence is replace with synthesized background noise. Silence suppression can reduce the amount of cells transmitted for a given voice conversation by 50 percent.

- **Use of VBR (variable bit rate) rather than CBR**—By combining the positive attributes of voice compression and silence suppression, ATM-based voice conversations are able to be transmitted using variable-bit rate bandwidth management. By only using bandwidth when someone is talking, remaining bandwidth is available for data transmission or other voice conversations. Use of VBR is controlled via two parameters:
1. Peak voice bit rate controls the maximum amount of bandwidth a voice conversation can be given when there is little or no contention for bandwidth.

2. Guaranteed voice bit rate controls the minimum amount of bandwidth that must be available to a voice conversation regardless of how much contention exists for bandwidth.

Standards for voice transmission over ATM (VTOA) networks are being developed by the ATM Forum. Among the standards available or under development are the following:

- Circuit Emulation Standard (CES)—Defines voice transport over ATM networks using CBR (constant bit rate). Equivalent to PVCs over frame relay nets.
- VTOA–ATM—For use on private or public ATM networks, defines voice transmission using the following:
  - ISDN (integrated services digital network) as a voice source network
  - Transport of compressed voice over ATM
  - Virtual tunnel groups that are able to handle multiple calls simultaneously between two locations
- VTOA to the Desktop—Defines interoperability between ATM and non-ATM networks.

Figure 5-13 illustrates the transmission of voice conversations over an ATM network.
Voice/Data Multiplexers

As opposed to using a switch-based frame relay or ATM network for wide area transmission of voice and data, organizations have traditionally chosen to link combined voice and data transmission over long distances via leased digital transmission services such as T-1. From a business perspective, a key difference between switched services such as Frame Relay or ATM and leased services such as T-1 is that switched services are usually tariffed according to usage and leased services are usually tariffed according to a flat monthly rate. As a result, leased services are being paid for 24 hrs/day, 7 days/week, whether they are being used or not.

Many corporations that once maintained a private network of voice/data multiplexers linked via T-1 or other high-speed digital services, have found that the usage-based pricing of frame relay networks can save them significant expense. A voice/data multiplexer is able to simultaneously transmit digitized voice and data over a single digital transmission service by assigning the voice and data transmissions to separate channels.

ISDN

Integrated services digital network is a switched digital, rather than analog, service that is also capable of transmitting voice and data simultaneously. Rather than using modems, ISDN requires devices that are officially known as terminal adapters but that are frequently marketed as ISDN data/voice modems. ISDN BRI (Basic Rate Interface) service offers two 64 Kbps channels. One of these channels is used for data while the other is used to simultaneously transmit voice. Analog phones or fax machines can be interfaced to the ISDN data/voice modem in order to allow these analog devices to access ISDN’s digital transmission service. Point-to-point ISDN connections require both ends of the transmission to be able to access ISDN services via ISDN data/voice modems.

ISDN is not nearly as available as switched analog voice phone service. In addition, pricing policies for ISDN can include both a monthly flat fee as well as an additional usage-based tariff Figure 5-14 illustrates the differences between simultaneous voice and data transmission using ISDN.

Figure 5-14  Simultaneous Voice/Data Transmission with ISDN
Wireless Voice Transmission

Modern wireless telephones are based on a cellular model. As shown in Figure 5-15, a wireless telephone system consists of a series of cells that surround a central base station, or tower. Cells are arranged so that no two adjoining cells use the same frequency. In this manner, there is a clear point of demarcation between cells. The term *cellular phone* or its abbreviation *cell phone* comes from the cellular nature of all wireless networks.

When turned on, a wireless phone is constantly communicating with the closest cell tower in the background. The point of this background communication is to let the cellular system know where the phone is so that incoming calls can be routed to the correct tower for transmission to the phone. When a phone makes or receives a call it initiates a connection between itself and the nearest tower. If over the course of the call the phone handset moves away from the current tower into a new cell the background communication link will be used to “hand off” the call from the tower in the old cell to the tower in the new cell. The handoff between the towers must be seamless to the end user and must carry forward the call information, such as air time, user ID, and so on, for proper billing. The connection of multiple cell sites, together with handoffs, allows a carrier to build a nationwide network in which calls can be made coast to coast.

**Analog Cellular**

The traditional circuit-switched analog cellular network is more properly known by the transmission standard to which it adheres: the *advanced mobile phone service*
AMPS), which operates in the 800 MHz frequency range. Although AMPS networks currently have the broadest coverage of any network in the United States, they have significant limitations. Because the connection is analog in nature, AMPS calls offer relatively poor signal quality; static and interference are inherent with the system. Another key issue with analog AMPS networks is that they can handle relatively few concurrent calls per cell.

**Digital Cellular**

To overcome these limitations, carriers have steadily moved to digital cellular systems. In a digital cellular system, the call is digitized at the telephone handset and sent in a digital format to the tower. Because this signal is digital in nature, quality is greatly improved. As long as you can get a signal, the call will sound perfect. However, as soon as the signal is lost, the call is dropped.

Another key feature from the perspective of wireless carriers is that the digital network allows more calls to share the common bandwidth in a cell concurrently, increasing capacity and billable minutes compared to analog cellular. This digital link is also better equipped to support wireless data transmission, paving the way for technologies such as Internet enabled phone handsets.

Digital cellular seeks to evolve an all-digital network architecture capable of delivering a variety of telecommunications services transparently to users at any time, regardless of their geographic location. Digital cellular is not a totally new “from the bottom up” telecommunications architecture. In fact, it is the integration of a number of existing telecommunications environments. Digital cellular seeks to combine the capabilities of the PSTN, otherwise known as the landline telephone network, with a new, all-digital cellular network, along with paging networks, and satellite communications networks.

The need for seamless delivery of a combination of all of the above services is easily illustrated by the plight of today’s mobile professional. A single person has a phone number for his or her home phone, a voice and fax number for the office, a cellular phone number for the automobile, a pager phone number for a pager, and perhaps even another phone number for a satellite service phone for use outside of cellular phone areas. The premise of digital cellular is rather straightforward: one person, one phone number.

This personal phone number or PPN would become the user’s interface to digital cellular and the vast array of transparently available telecommunications services. This personal phone number is a key concept to digital cellular. It changes the entire focus of the interface to the telecommunications environment from the current orientation of a number being associated with a particular location regardless of the individual using the facility to a number being associated with particular individual regardless of the location, even globally, of the accessed facility. Figure 5-16 illustrates the basic elements of digital cellular.

**Digital Cellular Standards**

Given the limited bandwidth (only about 140 MHz from 1.85 GHz to 1.99 GHz, referred to as the 2 GHz band) allocated to digital cellular and the potentially large number of subscribers needing to share that limited bandwidth, a key challenge for digital cellular is the ability to maximize the number of simultaneous conversations over a finite amount of bandwidth. Just as multiplexing was originally introduced in the study of wide area networks as a means of maximizing the use of wire-based circuits, two variations of multiplexing are being field tested as a means of maximizing the use of the allocated bandwidth of these air-based circuits.
TDMA (Time Division Multiple Access) and CDMA (Code Division Multiple Access) are the two access methodologies used in digital cellular systems. Both offer significant capacity increases compared to AMPS analog cellular systems. A TDMA-based digital cellular can support three times (some tests indicate six or seven times) the transmission capacity of analog cellular, while CDMA can offer as much as a tenfold increase. Note that the names of each of these techniques end in the words *multiple access* rather than *multiplexing*. The *multiple access* refers to multiple phone conversations having access to the same bandwidth and yet not interfering with each other. Figure 5-17 illustrates TDMA and CDMA.

TDMA achieves more than one conversation per frequency by assigning time slots to individual conversations. Ten timeslots per frequency are often assigned, with a given cellular device transmitting its digitized voice only during its assigned time slot. Receiving devices must be in synch with the time slots of the sending device in order to receive the digitized voice packets and reassemble them into a natural-sounding analog signal. TDMA should be able to transmit data at 9.6 Kbps. TDMA digital standards to handle call set-up, maintenance, and termination have been defined by the Telecommunications Industry Association (TIA) as follows:

- IS-130: TDMA Radio Interface and Radio Link Protocol 1
- IS-135: TDMA Services, Async Data, and FAX
Global System for Mobile Communication (GSM) is a new service layer that overlies TDMA. GSM provides a standardized billing interface and a means of offering enhanced data services. The standardized billing interface is particularly important to carriers because it simplifies the way they bill the consumer and provides a framework for seamless roaming between the GSM networks of different companies. Another key feature of GSM is the use of a SIM card to store the user’s information.
including their phone number, contacts, and so on. If users want to change phones, they need only move the SIM card from one device to another. Since the SIM card contains the entire user’s information, no programming of the new phone is needed, thus reducing the service cost to the carrier. Most TDMA-based carriers are migrating to GSM.

CDMA attempts to maximize the number of calls transmitted within a limited bandwidth by using a spread spectrum transmission technique. Rather than allocating specific frequency channels within the allocated bandwidth to specific conversations as is done with TDMA, CDMA transmits digitized voice packets from numerous calls at different frequencies spread all over the entire allocated bandwidth spectrum.

To keep track of which voice packet belongs with which call, each is marked with a code. This technique is not unlike the datagram connectionless service used by packet switched networks to send packetized data over numerous switched virtual circuits within the packet switched network. By identifying the source and sequence of each packet, the original message integrity is maintained while maximizing the overall performance of the network. The CDMA standards defined by the TIA are IS-95a and IS-99: Data Services Option for Wideband Spread Spectrum Digital Cellular Systems, commonly known as CDMAone.

TDMA and CDMA networks are both deployed in the United States. In Europe and much of the rest of the world, GSM (based on TDM) is deployed, while Personal Handyphone System (PHS) is the digital cellular standard implemented in Japan. Currently, these various digital cellular transmission standards are not interoperable, thereby precluding the possibility of transparent global access to digital cellular services.

**TDMA VS. CDMA: A TALE OF CAPABILITIES VS. PATENT ROYALTIES**

An analysis of the technical details of TDMA and CDMA shows that CDMA is the better technology in almost all aspects. However, there is a business limitation associated with CDMA that has limited its deployment. CDMA is patented by Qualcomm, based in San Diego, California. As the patent holder for the CDMA technologies, Qualcomm currently requires an 8 percent patent royalty on all CDMA devices sold. Many manufacturers are not willing to reduce their profit margin to pay Qualcomm, so they have instead chosen to use the more technically limited TDMA technology. Other vendors are actively working to develop an alternative CDMA-like technology implementation that is different enough that it will skirt Qualcomm’s patent rights.

The question a carrier has to ask itself is if it is worth the hassle and expense of licensing CDMA technology to realize the benefits of higher cell density and the faster data transmission speeds offered by CDMA-based 1xRTT and EV-DO, as described in the next section. While Verizon Wireless and Sprint have chosen to go with CDMA, other major carriers such as AT&T Wireless, Cingular, and T-Mobile have chosen to stick with TDMA and avoid Qualcomm’s patents.

An early GSM draft standard called for GSM to operate over CDMA rather than TDMA networks. However, many vendors and carriers were not willing to standardize on a patented technology that required royalty payments to a single company. The lesson to be learned here is that the marketplace likes open standards and is willing to forgo potential technological benefits to eliminate the cost and limitations associated with licensing patented technology.
MARKET ISSUES FACING WIRELESS TELEPHONE SYSTEMS

A key issue in the digital cellular market space is number portability. If a user decides to change from one cellular company to another, the person would be required to change their phone number. With the low cost of cellular services, some people have replaced their land line phone with cell phone. For these users, the pain of change numbers to change carriers held them “hostage” to one carrier. In mid-2003, the FCC won judicial approval for a ruling requiring that cellular companies allow a person to keep their phone number whenever they change carriers, effective November 2003. Although sure to have its deployment pains, cellular number portability ushers in a new level of consumer choice and carrier responsibility. No longer will customers be stuck with a carrier because they have advertised their cellular number.

Perhaps the most significant hurdles to the future of digital cellular services are the individual, conflicting, business missions of the various companies that must somehow produce a comprehensive, seamless, global, transparent digital cellular service for subscribers. Each of these firms must look out for their best interests while trying to work together for the betterment of the consumer.

Cellular telephony is a big business, with carriers investing $7.7 billion for auctioned spectrum in 1995 to 1996. It is estimated that somewhere between an additional $10 billion and $50 billion must be spent on digital cellular infrastructure before nationwide services can be achieved. The dilemma is that cellular vendors must price their services attractively enough to gain market share while maintaining adequate cash flow to service a tremendous amount of debt.

Ultimately the consumer will determine the future of cellular technologies. If recent trends continue to hold true, that future will be bright. The number of cellular phones continues to grow worldwide. In developing nations that never achieved a solid wired telecommunications infrastructure, carriers are skipping that step and rolling out cellular services instead. In countries that have a solid wired infrastructure, many consumers are choosing to forgo a traditional wired telephone and are choosing to use their cellular service as their primary telephone.

WIRELESS DATA SERVICES

Although wireless LANs offer mobility to users across a local scope of coverage, a variety of wireless services are available for use across wider geographic spans. These wireless WAN services vary in many ways including availability, applications, transmission speed, and cost. Among the available wireless WAN services that will be explained further are the following:

- Private packet radio
- Enhanced paging and two-way messaging
- Circuit-switched analog cellular
- CDPD—cellular digital packet data
- GPRS—general packet radio service
- CDMA (a.k.a 1xRTT)—code division multiple access or single carrier (1x) radio transmission technology
Wireless Data Services

- EDGE—enhanced data for GSM evolution
- EV-DO—evolution data only

The key characteristics of these and other wireless WAN services are summarized in Figure 5-19.

### Wireless Data Service Generations

Often wireless data services are referred to by their generation: 1\textsuperscript{st} Generation (1G), 2\textsuperscript{nd} Generation (2G), Advanced 2\textsuperscript{nd} Generation (2.5G) and 3\textsuperscript{rd} Generation

<table>
<thead>
<tr>
<th>Wireless WAN Service</th>
<th>Geographic Scope</th>
<th>Directionality</th>
<th>Data Characteristics</th>
<th>Billing</th>
<th>Access Device</th>
<th>Standards</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enhanced Paging</td>
<td>National</td>
<td>One- or two-way relatively short messages</td>
<td>100 characters or less</td>
<td>Flat monthly charges increasing with coverage area</td>
<td>Pagers</td>
<td>Mobetex</td>
</tr>
<tr>
<td>Private Packet Radio</td>
<td>Nearly national, more cities than CDPD but less than circuit-switched cellular</td>
<td>Full duplex Packet-switched digital data</td>
<td>4.8 Kbps</td>
<td>Per character</td>
<td>Proprietary modem compatible with particular private packet radio service</td>
<td>Proprietary</td>
</tr>
<tr>
<td>Circuit Switched Analog Cellular</td>
<td>National</td>
<td>Full-duplex Circuit switched</td>
<td>14.4 Kbps max</td>
<td>Call duration</td>
<td>Modems with specialized error correction for cellular circuits</td>
<td>Analog cellular</td>
</tr>
<tr>
<td>CDPD</td>
<td>Limited to large metropolitan areas</td>
<td>Full duplex Packet-switched digital data</td>
<td>19.2 Kbps max</td>
<td>flat monthly charge plus usage charge per kilopacket</td>
<td>CDPD modem</td>
<td>CDPD</td>
</tr>
<tr>
<td>GPRS</td>
<td>National in large metro areas</td>
<td>Full duplex Packet-switched digital data</td>
<td>30 Kbps to 40 Kbps</td>
<td>Monthly flat rate for business users; metered based on usage for consumer</td>
<td>Tethered cell phones and PCMCIA/CF cards</td>
<td>TDMA</td>
</tr>
<tr>
<td>IxRTT</td>
<td>National in large metro areas</td>
<td>Full duplex Packet-switched digital data</td>
<td>40 Kbps to 56 Kbps</td>
<td>Monthly flat rate for business users; metered based on usage for consumer</td>
<td>Tethered cell phones and PCMCIA/CF cards</td>
<td>CDMA</td>
</tr>
<tr>
<td>EDGE</td>
<td>Limited metro areas</td>
<td>Full duplex Packet-switched digital data</td>
<td>Up to 384 Kbps</td>
<td>Monthly flat rate for business users; metered based on usage for consumer</td>
<td>Tethered cell phones and PCMCIA/CF cards</td>
<td>TDMA</td>
</tr>
<tr>
<td>EV-DO</td>
<td>Limited metro areas</td>
<td>Full duplex Packet-switched digital data</td>
<td>600 Kbps to 2 Mbps</td>
<td>Monthly flat rate for business users; metered based on usage for consumer</td>
<td>PCMCIA cards</td>
<td>CDMA</td>
</tr>
</tbody>
</table>

*Figure 5-19* Wireless WAN Services Technology Analysis
(3G). Each of these generations can be defined by the services and data speeds they offer:

- 1G networks are defined as analog networks that carried just voice traffic such as AMPS.
- 2G networks are defined by technologies that use digital transmission between the handset and the tower, such as TDMA and CDMA. Most 2G systems provide 9.6-14.4 Kbps circuit-switched data service.
- 2.5G networks are defined as digital networks that provide between 56 Kbps and 115 Kbps of data capacity, such as GPRS and 1xRTT.
- 3G is an ITU specification for the third generation of mobile communications technology. The primary benefit of 3G is increased bandwidth ranging from 128 Kbps in a moving car to 2 Mbps for fixed applications. Examples of 3G network are EDGE and EV-DO; however, many other 3G network technologies are currently being developed.

**Circuit-Switched Analog Cellular**

Transmitting data over AMPS analog cellular networks requires modems that support specialized cellular transmission protocols on both ends of the cellular transmission in order to maximize throughput. Examples of such protocols include **MNP-10** **Adverse Channel Enhancements** and **Enhanced Throughput Cellular (ETC)**. In some cases, cellular service providers have deployed modem pools of cellular enhanced modems at the **mobile telephone switching office (MTSO)** where all cellular traffic is converted for transmission over the wire line public switched telephone network (PSTN). Figure 5-19 illustrates data transmission over the circuit switched analog cellular network.

Using analog cellular, a call must be placed from the handset to a modem. From the perspective of the network this data call is just like a voice call and is billed based on call duration. Speeds for analog cellular are limited to a maximum of 14.4 Kbps with speed dependent on the quality of the connection.

![Figure 5-19](image-url)  
**Figure 5-19**  
Data Transmission over the Circuit-Switched Analog Cellular Network
**CDPD**

Once one of the most widely deployed technologies, **CDPD** is rapidly be phased out. AT&T Wireless, one of the largest CDPD network providers, has determined that the CDPD is not in its future. With the wireless carrier strapped for cash after buying spectrum for newer digital technologies, it has determined it is not financial viable to maintain two separate networks. AT&T has stated that existing CDPD users should migrate to EDGE, and AT&T Wireless will discontinue the CDPD service at the end of 2004. Other carriers are following suit, making CDPD a poor choice for new installations.

**GPRS and 1xRTT**

Unlike analog cellular transmission, transmitting digital data from a notebook computer over digital cellular networks does not require modulation. As a result, computers can interface directly to digital TDMA and CDMA-based digital cellular phones via serial ports. A second advantage to digital cellular networks is that connections are practically instantaneous because there is no carrier to establish and call set-up is greatly simplified. Figure 5-20 illustrates data transmission over a digital cellular network.

**GPRS** and **1xRTT** are the two currently competing digital cellular data standards in the United States. Both networks were deployed in 2002/2003 across major metropolitan areas. GPRS is based on the TDMA model, while 1xRTT is based on the CDMA model. This means that mobile devices will only work on one type of network AT&T Wireless, Cingular, and T-Mobile support the GPRS standard, while Sprint and Verizon Wireless support the 1xRTT standard.

Side-by-side tests between GPRS and 1xRTT have shown that 1xRTT is currently faster. 1xRTT networks have shown data rates as high as 100 Kbps, while GPRS testing has shown a maximum of around 56 Kbps. However, a multitude of factors can affect throughput, including cell tower load, Internet traffic conditions, and signal strength. As the build-out of these services continues, overall speeds are expected to increase. Both technologies have the potential to offer data service at speeds upward of 200 Kbps.

![Figure 5-20  Data Transmission over a Digital Cellular Network](image)
The build-out of GPRS and 1xRTT networks was not driven entirely by the need for wireless data. By deploying these technologies, wireless carriers were able to further increase the number of calls each cellular tower could handle while offering add-on features like downloadable ring tones, games, instant message, and multimedia messaging. These new features represent a new income stream for the carriers. Since the majority of a wireless carrier’s revenue is based on consumer voice service, they initially marketed and priced the technology for consumers. Currently, business data users only make up a small percentage of a carrier’s revenue stream, so they weren’t targeted during the initial roll-out. The next upgrade to the technologies will increase the data throughput and be targeted more at business users.

**EDGE and EV-DO**

EDGE and EV-DO are third generation (3G) wireless data transmission technologies for existing TDMA/CDMA network infrastructures. EDGE is the upgrade path for TDMA-based GPRS, while EV-DO is the upgrade path for CDMA-based 1xRTT solutions. As discussed in the GPRS and 1xRTT section, devices manufactured to work on a TDMA/EDGE network will not work on a CDMA/EV-DO network, and visa versa. However, EDGE device will be backward compatible with GPRS and TDMA, while EV-DO devices will be backward compatible with 1xRTT and CDMA.

EDGE networks have theoretical data speeds up to 384 Kbps, although early deployments have only provided around 100 Kbps. EV-DO networks have theoretical data speeds of up to 2 Mbps. However, just as in EDGE, early deployments have shown practical speeds much lower, in the 384 Kbps range.

These technology upgrades are focused on delivering high speeds for wireless data. Unlike the GPRS and 1xRTT build-out, the upgrades are focused more the business user than the consumer. The good news for carriers is that this upgrade is easier than the migration from TDMA/CDMA to GPRS/1xRTT. This upgrade generally includes software changes in the MTSO equipment and upgrades in the devices used to connect the network. This upgrade does not require all existing devices to be replaced. Users that need the extra data speeds can upgrade their equipment to the new standard, while users with older technology will be able to continue to work, albeit at their existing data speeds.

**Next Generation Wireless Services**

The next generation of wireless will focus on specialized high-bandwidth wireless services. The standards for this next generation of wireless transmission services are most often grouped under the name of 3G (Third Generation) Mobile Telephony, otherwise known as UWC (Universal Wireless Communications)-136. The ITU’s overall initiative for 3G Wireless Telephony is known as IMT 2000 (International Mobile Telecommunications) and consists of three separate initiatives: UMTS (ETSI), UWC-136 (TIA), and CDMA2000 (TIA). These initiatives are summarized in Figure 5-21.

In addition to new 3G standards, there is ongoing research into what has been termed 4G. NTT DoCoMo in Japan has begun testing the next evolution of their W-CDMA network. The hope is for this new 4G network to work up to 100 Mbps. DoCoMo expects to have a commercial 4G network available by 2010.
A TOP-DOWN APPROACH TO WIRELESS WAN SERVICES ANALYSIS

Due to the many variable factors concerning these wireless WAN services, it is important to take a top-down approach when considering their incorporation into an organization’s information systems solution. Questions and issues to be considered on each layer of the top down model for wireless WAN services are summarized in Figure 5-22.

As a practical example of how to use the top down model for wireless WAN services analysis, start with the business situation that requires wireless support and examine the applications and data characteristics that support the business activity in question. For example, which of the following best describe the data to be transmitted by wireless means?

<table>
<thead>
<tr>
<th>3G Transmission Technology</th>
<th>Stationary Maximum Transmission Rate</th>
<th>Moving Upstream Transmission Rate</th>
<th>Moving Downstream Transmission Rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>W-CDMA—Wideband Code Division Multiple Access</td>
<td>2 Mbps</td>
<td>100 Kbps</td>
<td>384 Kbps</td>
</tr>
<tr>
<td>UMTS—Universal Mobile Telecommunications System</td>
<td>2 Mbps</td>
<td>100 Kbps</td>
<td>384 Kbps</td>
</tr>
<tr>
<td>CDMA 2000—Code Division Multiple Access 2000</td>
<td>2 Mbps</td>
<td>100 Kbps</td>
<td>384 Kbps</td>
</tr>
</tbody>
</table>

*Figure 5-21 3G Mobile Telephony Standards*

---

**Applied Problem Solving**

**A TOP-DOWN APPROACH TO WIRELESS WAN SERVICES ANALYSIS**

Due to the many variable factors concerning these wireless WAN services, it is important to take a top-down approach when considering their incorporation into an organization’s information systems solution. Questions and issues to be considered on each layer of the top down model for wireless WAN services are summarized in Figure 5-22.

As a practical example of how to use the top down model for wireless WAN services analysis, start with the business situation that requires wireless support and examine the applications and data characteristics that support the business activity in question. For example, which of the following best describe the data to be transmitted by wireless means?

<table>
<thead>
<tr>
<th>Top-Down Layer</th>
<th>Issues/Implications</th>
</tr>
</thead>
</table>
| Business       | • What is the business activity that requires wireless transmission?  
                 • How will payback be calculated? Has the value of this business activity been substantiated?  
                 • What are the anticipated expenses for the 6-month, 1-year, and 2-year horizons?  
                 • What is the geographic scope of this business activity? Localized? National? International? |
| Application    | • Have applications been developed especially for wireless transmission?  
                 • Have existing applications been modified to account for wireless transmission characteristics?  
                 • Have training and help-desk support systems been developed? |
| Data           | • What is the nature of the data to be delivered via the wireless WAN service? short bursty transactions, large two-way messages, faxes, file transfers?  
                 • Is the data time-sensitive or could transmissions be batched during off-peak hours for discounted rates?  
                 • What is the geographic scope of coverage required for wireless data delivery? |
| Network        | • Must the WAN service provide error correction?  
                 • Do you wish the WAN service to also provide and maintain the access devices? |
| Technology     | • Which wireless WAN service should be employed?  
                 • What type of access device must be employed with the chosen WAN service?  
                 • Are access devices proprietary or standards-based? |

*Figure 5-22 Top Down Analysis for Wireless WAN Services*
The nature of the content, geographic scope, and amount and urgency of the data to be transmitted will have a direct bearing on the particular wireless WAN service employed. Unfortunately, no single wireless WAN service fits all application and data needs. Once a wireless WAN service is chosen, compatibility with existing local area network architectures and technology must be assured.

**PBX AND CTI SERVICES**

In order to provide flexible voice communications capability among people within a business organization as well as with the outside world, a switching device known as a **PBX**, or **Private Branch Exchange** is often employed. Sales of PBXs in the United States represent approximately a $4 billion annual market with service on these PBXs accounting for an additional $2 billion in revenue. The PBX market is currently dominated by Nortel (Northern Telecom) with its line of Meridian PBXs and by Lucent Technologies (formerly AT&T Global Business Communications Systems) with its line of Definity PBXs. Each of these two major players control about 25 percent market share. The major players in the PBX market and their approximate market shares are displayed in Figure 5-23.

**PBX Functionality and Architecture**

As illustrated in the I-P-O (Input-Processing-Output) diagram of Figure 5-24, a PBX provides an interface between users and the shared private or public network connections available for carrying users’ voice and data traffic. The additional intelligent services offered by a PBX allow users to use their phones more efficiently and effectively.

A PBX is really just a privately owned, smaller version of the switch in telephone company central offices that control circuit switching for the general public.

<table>
<thead>
<tr>
<th>Approximate PBX Market Share</th>
<th>PBX Vendors</th>
</tr>
</thead>
<tbody>
<tr>
<td>25–30%</td>
<td>Nortel, Lucent Technologies</td>
</tr>
<tr>
<td>10–20%</td>
<td>NEC and Mitel</td>
</tr>
<tr>
<td>5–10%</td>
<td>Siemens Rolm</td>
</tr>
<tr>
<td>2–5%</td>
<td>Fujitsu, Intecom, Ericsson, Hitachi</td>
</tr>
<tr>
<td>Less than 2%</td>
<td>Toshiba, Executone, Tadrian, SRX, Harris Digital</td>
</tr>
</tbody>
</table>

*Figure 5-23  PBX Vendors and Market Share*
PBX and CTI Services

Depending on the requested destination, switched circuits are established, maintained, and terminated on a per-call basis by the PBX switching matrix.

Beyond the switching capabilities of a PBX, programmable features offer advanced functionality to users. These features and the overall performance of the PBX are controlled by software programs running on specialized computers within the PBX in an area sometimes referred to as the PBX CPU, stored program control or common control area.

Telephone sets in user offices are connected to the PBX via slide-in modules or cards known as **line cards**, **port cards**, or **station cards**. Connection to an outside network (usually the PSTN) is accomplished via **trunk cards**. Trunk cards vary in design from easily scalable cards to specialized cards for a particular type of network line.

Some PBXs allow any chassis slot to be used for any type of card or module, while other PBXs specify certain slots for line cards and others for trunk cards. Starting with an open chassis or cabinet with power supply and backbone or backplane, modules or cards are added to increase PBX capacity for either user extensions or connections to the outside network. Additional cabinets can often be cascaded to offer PBX expandability. Figure 5-25 illustrates the physical attributes of a representative PBX.

PBX Technology Analysis

Before reviewing PBX technology, it is essential to have performed a thorough top-down analysis beginning with the business needs and functionality that must be met by the chosen PBX technology. Having identified those PBX features that are most important to a given business, the following information could be used as a representative sample of typical PBX features and services.

PBX features and services tend to fall into three broad categories:

- Features and services that provide users with flexible usage of PBX resources
- Features and services that provide for data/voice integration
- Features and services that control and monitor the use of those PBX resources
In the flexible usage category, features such as conference calling, call forwarding, call transfer, speed dialing, redialing and call hold are commonplace and shouldn’t require further explanation. Other voice-based PBX features and services that support flexible usage of PBX resources are summarized in Figure 5-26.

<table>
<thead>
<tr>
<th>Feature/Service</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Least Cost Routing (LCR)</td>
<td>Using routing and pricing information supplied by the user, the PBX chooses the most economical path for any given call. This feature was especially important when WATS (Wide Area Telecommunications Service) lines were more prevalent. These days, thanks to competition and discount programs among long-distance carriers, PBXs can access any outgoing trunk rather than trying to get certain calls onto certain trunks.</td>
</tr>
<tr>
<td>Automatic Call Distribution (ACD)</td>
<td>Incoming calls are routed directly to certain extensions without going through a central switchboard. Calls can be routed according to the incoming trunk or phone number. Often used in customer service organizations in which calls may be distributed to the first available agent.</td>
</tr>
<tr>
<td>Call Pickup</td>
<td>Allows a user to pickup or answer another user’s phone without the need to actually forward calls.</td>
</tr>
<tr>
<td>Paging</td>
<td>Ability to use paging speakers in a building. May be limited to specific paging zones.</td>
</tr>
<tr>
<td>Direct Inward Dialing (DID)</td>
<td>Allows calls to bypass the central switchboard and go directly to a particular user’s phone.</td>
</tr>
<tr>
<td>Hunting</td>
<td>Hunt groups are established to allow incoming calls to get through on alternate trunks when a primary trunk is busy. For example, most businesses publish only one phone number even though they may have multiple incoming trunks. If the primary trunk is busy, the PBX hunts for an open trunk transparent to the user.</td>
</tr>
<tr>
<td>Prioritization</td>
<td>Individual extensions can be given priority access to certain trunks or groups of trunks. In most cases, PBXs are equipped with fewer outgoing trunks than internal extensions or station lines. If certain users must have access to outside lines, prioritization features are important.</td>
</tr>
<tr>
<td>Night Mode</td>
<td>Many companies close their switchboard at night but still have employees working who must be able to receive and make phone calls.</td>
</tr>
</tbody>
</table>

*Figure 5-25  PBX Physical Architecture*

*Figure 5-26  Voice Based PBX Features and Services*
Data/Voice Integration Features and Services  

Data/Voice integration by PBXs is increasingly common although PBXs can vary significantly in the extent of support for data transmission. Differences in data interfaces and whether or not those interfaces and associated software represent an upgrade at additional cost should be investigated thoroughly before any PBX purchase. In some cases, data is transmitted through the PBX via a dedicated connection and in other cases a specialized hybrid voice/data phone is used to transmit both voice and data simultaneously over a single connection to the PBX. Data and data/voice integration related features and issues are summarized in Figure 5-27.

Control and Monitoring Features and Services  

Control and monitoring features range from the simple, such as limiting access to outside lines from certain extensions, to the complex, such as entire stand-alone call accounting systems. Call accounting systems are often run on separate system that interfaces directly to the PBX and execute specially written software. Accounting reports or bills sorted by department or extension can be run on a scheduled basis or on demand. Exception reports can be generated to spot possible abuses for calls over a certain length or cost, or calls made to a particular area code. Incoming as well as outgoing calls can be tracked. Call accounting systems can pay for themselves in a short amount of time by spotting and curtailing abuse as well as by allocating phone usage charges on a departmental basis.

The information on which such a call accounting system depends is generated by the PBX. In a process known as SMDR or station message detail recording, an individual detail record is generated for each call. This data record can then be transferred from the PBX to the call accounting system computer, usually from an RS-232 DB-25 port on the PBX to the serial port on the PC. Data records can be stored and summarized on the call accounting system computer, dependent on available disk space. Figure 5-28 illustrates the set-up of a call accounting system.

Auxiliary Voice-Related Services  

Just as call accounting systems are most often an add-on device for PBXs, other auxiliary systems exist to enhance PBX capability. The auxiliary nature of these systems implies that they are often not included as standard

<table>
<thead>
<tr>
<th>Feature/Service</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ISDN Support</td>
<td>Are WAN interfaces for ISDN (Integrated Services Digital Network) supplied or available as upgrades? ISDN BRI (Basic rate interface) service is two 64 Kbps channels. ISDN PRI service is twenty-three 64 Kbps channels.</td>
</tr>
<tr>
<td>T-1 Support</td>
<td>Are T-1 (1.544 Mbps) interfaces supported? Outside of North America, are E-1 (2.048 Mbps) supported? Are codecs included? Are channel banks included?</td>
</tr>
<tr>
<td>Data Interfaces</td>
<td>Are computer data interfaces included on the PBX? Are hybrid voice/data phones available? Are LAN interfaces such as ethernet and token ring as well as serial interfaces such as RS-232 supported? How many of the following services are supported: fax transmission, modem pooling, printer sharing, file sharing, video conferencing?</td>
</tr>
<tr>
<td>PBX-to-Host Interfaces</td>
<td>Prior to the advent of open systems computer telephony integration APIs such as TAPI and TSAPI, each PBX vendor had their own PBX-Host interface specification. How many of the following vendor-specific PBX-Host interfaces are supported? Nortel: Meridian Link; Rolm: CallBridge; IBM: CallPath; AT&amp;T: Passageway; Siemens: Applications Connectivity Link (ACL); Mitel: NeVaDa (Networked Voice and Data).</td>
</tr>
</tbody>
</table>

*Figure 5-27* Data and Data/Voice Integration PBX Features and Services
features on PBXs but may be purchased separately from either the PBX vendor or third-party manufacturers. Sometimes these services are available as a combination of specialized PC boards and associated TAPI or TSAPI compliant software. Figure 5-29 lists and describes a few of the more popular auxiliary PBX systems.

<table>
<thead>
<tr>
<th>Service/Device</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Automated Attendant</td>
<td>A recorded message works with a touch-tone phone that requests callers to press the number of the extension they wish to reach. Those wishing to speak to an operator are transferred.</td>
</tr>
<tr>
<td>Voice Mail</td>
<td>Voice mail systems can vary widely in cost and sophistication. After recording an initial message for someone, voice mail systems may allow the voice mail to be handled like a written phone message. It can be forwarded, copied, appended to, saved, recalled, or deleted.</td>
</tr>
<tr>
<td>Voice Response Units</td>
<td>Menu selections are offered to callers who use the touch pad to navigate through menu selections. Some answering machines include VRUs that allow the owner to check for messages remotely.</td>
</tr>
<tr>
<td>Interactive Voice Response</td>
<td>Banks allows customers to make account inquiries and transactions via VRUs and airlines provide arrival and departure information via VRUs.</td>
</tr>
<tr>
<td>Voice Processor</td>
<td>Performs same basic functions as VRU but may also provide additional services based on voice response, speech recognition or tone detection.</td>
</tr>
<tr>
<td>Voice Server</td>
<td>A LAN based server stores, processes and delivers digitized voice messages. Often used as the processing and storage component of a voice mail system.</td>
</tr>
<tr>
<td>Music/Ads on Hold</td>
<td>When customers are put on hold, music plays, or alternatively, a tape-recorded sales message interrupts periodically with messages such as, “Your call is important to us, please stay on the line.”</td>
</tr>
</tbody>
</table>

Figure 5-28 Call Accounting Systems Installation

Figure 5-29 Auxiliary Voice-Related Services/Devices
**Computer Telephony Integration (CTI)**

CTI or computer telephony integration seeks to integrate the two most common productivity devices, the computer and the telephone, to enable increased productivity not otherwise possible by using the two devices in a nonintegrated fashion. CTI is not a single application, but an ever-widening array of possibilities spawned by the integration of telephony and computing. Figure 5-30 briefly describes some of the sub-categories of CTI applications.

**CTI Architectures** Computer telephony integration is commonly implemented in one of the following three architectures:

- PBX-to-host interfaces
- Desktop CTI
- Client/server CTI

<table>
<thead>
<tr>
<th>CTI Application Category</th>
<th>Application Description</th>
</tr>
</thead>
</table>
| Call Control             | • Using computer-based applications, users are more easily able to use all of the features of their phone system or PBX, especially the more complicated but seldom used features.  
  • Includes use of features like on-line phone books, auto-dialing, click-and-point conference calls, on-line display and processing of voice mail messages. |
| Automated Attendant      | • This allows callers to direct calls to a desired individual at a given business without necessarily knowing their extension number. |
| Automated Call Distribution | • Used primarily in call centers staffed by large numbers of customer service agents, incoming calls are automatically distributed to the first available rep, or in some cases, the rep that serves a given geographic region as automatically determined by the computer based on the incoming phone number. |
| Audiotex                 | • These systems deliver audio information to callers based on responses on the touch-tone keypad to prerecorded questions. Primarily used for information hotlines. |
| Fax-On-Demand            | • By combining computer-based faxing with interactive voice response, users can dial in and request that specific information be faxed to their fax machine. |
| Interactive Voice Response | • Interactive Voice Response systems differ from audiotex systems in that IVR systems support on-line transaction processing rather than just information hotline applications. As an example, banks use IVR systems to allow users to transfer funds between accounts by using only a touch-tone phone. |
| Outbound Dialing         | • Also known as Predictive Dialing, this merger of computing and telephony uses a database of phone numbers, automatically dials those numbers, recognizes when calls are answered by people, and quickly passes those calls to available agents. |
| Unified Messaging         | • Perhaps the most interesting for the LAN-based user, unified messaging, also known as the Universal In-Box will allow voice mail, e-mail, faxes, and pager messages to all be displayed on a single graphical screen. Messages can then be forwarded, deleted, or replied to easily in point and click fashion. Waiting calls can also be displayed in the same Universal In-Box. |

*Figure 5-30  Computer Telephony Integration Functionality*
Traditionally, computer telephony integration was achieved by linking mainframes to PBX via proprietary PBX-to-host interfaces. Applications were required to be compatible with both the model of mainframe computer and PBX installed. In many cases, these systems actually linked to an ancillary device known as an ACD or automatic call distribution unit. These systems were very expensive and were usually only employed in large customer service call centers. In this CTI architecture, all phones are controlled by the CTI application running on the mainframe computer.

Desktop CTI, also known as first-party call control is a much less expensive and simpler alternative to the PBX-to-host interface architecture. In this CTI architecture, individual PCs are equipped with telephony boards and associated call control software. Each desktop CTI-equipped PC controls only the phone to which it is directly attached. There is no overall automatic call distribution across multiple agents and their phones, and there is no sharing of call-related data among the desktop CTI PCs.

Finally, client/server CTI offers the overall shared control of the PBX-to-host CTI architecture at a cost much closer to the desktop CTI architecture. In this CTI architecture, a CTI server computer interfaces to the PBX or ACD to provide overall system management while individual client-based CTI applications execute on multiple client PCs. The advantage to such an architecture is that multiple CTI applications on multiple client PCs can share the information supplied by the single CTI server. Figure 5-31 illustrates the various CTI architectures.

**SUMMARY**

Network analysts must be qualified to design networks that are capable of carrying voice as well as data. Before designing such networks, it is essential for the network analyst to understand the nature of voice signals, as well as how voice signals can be processed and integrated into a cohesive network with data transmissions.

Voice bandwidth on the analog Public Switched Telephone Network (PSTN) is limited to 3,100Hz. In order to combine voice with data over a single transmission link, voice signals must first be digitized. Pulse code modulation or one of its derivatives are the most popular voice digitization techniques. Depending on voice compression algorithms employed, digitized voice requires between 64 Kbps and 8 Kbps of bandwidth.

The voice network is comprised of a hierarchy of switching offices designed to offer fast, reliable service. Switches are able to communicate with each other via a common language known as Signaling System 7. End-users are able to now use SS7 to monitor and control their carrier-based transmission circuits through the Advanced Intelligent Network.

While traditional switched services still carry the majority of telephone calls today, newer data network based alternatives have the potential to replace traditional telephony solutions. Using CODECs to encode an analog voice conversation into data packets, these alternatives promise to greatly reduce the cost of long distance calls while increasing the ability to integrate telephone service into computing infrastructures.

While these data network based alternatives can operate directly over data services such as frame relay and ATM, the leading alternative is to simply encode the data into IP datagrams that can be sent across any underlying data network. Technologies used to implement this vision include H.323 and SIP.

One of the fastest growing areas of telephony is wireless telephony. Using a cellular approach where a user can be handed off between towers, wireless telephony systems provide a seamless means of placing calls from mobile phones.

Where the cellular system was predominately analog under the AMPS standard, it has now evolved into digital networks using TDMA, GSM, and CDMA. These digital networks offer improved voice call quality while adding the ability to offer additional services such as wireless data transmission.
Modern wireless data transmission technologies are most commonly based on GSM/GPRS or CDMA/1xRTT. These 2.5G networks offer data speeds up to 100 Kbps in an “always on” mode. Newer 3G networks have the potential of greatly increasing data rates.

The PBX is the voice server or switch that links users’ phones with their intended destinations. Today’s PBXs have evolved from proprietary monolithic architectures to open, standards-based architectures.

PBXs are being increasingly seen as a specialized server for distributed client/server information systems. Computer telephony integration seeks to optimize the use of the telephone and the desktop computer by being able to share the information and functionality offered by each. Standardized APIs and PBX-to-host interfaces are required if CTI is to ever reach its full potential.
KEY TERMS

1xRTT  3G  ACD
adaptive pulse code modulation  ADPCM
advanced mobile phone service  alternate billing service
AMPS  automatic call distribution
call accounting systems  CDMA
CDPD  central office
Class 1 regional carrier  Class 2 sectional center
Class 3 primary center  Class 4 toll center
Class 5 office  CO
code division multiple access  codec
codecs  computer telephony integration
CTI  custom local area signaling service
dial tone  digital cellular
digital signal processors  DSP
DTMF  dual tone multi-frequency
EDGE  enhanced 800 services
enhanced throughput cellular  ETC
EV-DO  first party call control
global system for mobile
communication  GPRS
GSM  H.323
in-band signaling  intelligent call processing
inter-exchange carrier  inter-office signaling
Intranet  landline telephone network
last mile  line cards
local loop  MNP-10 adverse channel
enhancements  mobile telephone switching office
MTSO  out-of-band signaling
PAM  PBX
PCM  PDM
personal handyphone system  personal phone number
PHS  plain old telephone system
point of presence  POP
port cards  POTS
PPM  PPN
private branch exchange  PSTN
public switched telephone
network  pulse amplitude modulation
pulse duration modulation
pulse position modulation  pulse width modulation
pulses  PWM
pulse code modulation  session initiation protocol
signaling System 7  SS7
SMDR (station message detail
recording)  SS7
station cards  switching matrix
system signaling  tandem office
TDMA  telephone switch
time division multiple access  toll quality
touch-tone  trunk cards
voice compression  voice digitization
voice gateway  voice over IP
voice/data multiplexers  VOIP
wireless WAN services

REVIEW QUESTIONS

1. Why is it important for network analysts to be qualified to design voice networks?
2. How do the sound waves of the human voice actually get transferred onto and off of the voice network?
3. What is DTMF and what are its potential uses beyond assisting in completing a call?
4. What are the differences between the various voice digitization techniques?
5. How does PCM differ from ADPCM in terms of bandwidth requirements?
6. How is voice compression accomplished? What technology is involved?
7. What is the business motivation for voice compression? What is the potential trade-off?
8. What is the voice network hierarchy and what are the fundamental implications of such a hierarchical design?
9. What is out-of-band signaling and of what importance is such a technical ability in terms of emerging network services?
10. Why might a voice conversation not be totally transmitted via analog transmission even if the source and destination loops are analog?
11. How does sampling rate in voice digitization relate to the quality of the transmitted voice signal?
12. How does ADPCM accomplish digitized voice transmission in less than 64 Kbps?
13. Why is DS-0 considered a standard circuit for transmission of digitized voice?
14. How does PCM differ from other voice digitization techniques such as PAM?
15. What are the benefits of PCM over other voice digitization methods?
16. What is the role of a CODEC in voice digitization?
17. Compare the bandwidth of the PSTN with that of human hearing.
18. What is a channel bank?
19. What is toll quality and how is it related to ADPCM?
20. Show the mathematical proof of why a T-1 circuit is 1.544 Mbps.
21. What is a Signaling System?
22. What is the AIN and what role does SS7 play in such a network?
23. What is a tandem office?
24. What is required to transmit voice over the Internet?
25. What are some important features of VOIP client software?
26. Differentiate between H.323 and SIP.
27. Does VOIP require a computer be in place at each end of the call?
28. Why is quality of service so important to VOIP?
29. What is a voice gateway?
30. How is voice compression related to bandwidth requirements and delay?
31. What characteristic of frame relay must be overcome for effective voice transmission?
32. What are some of the issues surrounding voice over ATM?
33. What is the difference between CBR and VBR for voice over ATM?
34. Differentiate between Peak voice bit rate and guaranteed voice bit rate.
35. How does a FRAD assist in optimizing voice transmission over frame relay?
36. In terms of wireless phones, what is a cell? How do cells inter-relate?
37. What are the key limitations of AMPS?
38. Differentiate between analog and digital cellular transmission systems in terms of data transfer capabilities and equipment requirements.
39. Differentiate between TDMA and CDMA.
40. How is the notion of a personal phone number central to PCS and what changes in thinking about phone systems does it require?
41. What is GSM?
42. List six wireless data standards and their maximum speeds.
43. What are the requirements for a true 3G wireless network?
44. Compare and contrast GPRS and 1xRTT.
45. Compare and contrast EDGE and EV-DO.
46. What are the major architectural elements of a PBX?
47. Describe the basic architecture of a PBX. How is it like and unlike the switch in a CO?
48. What are some of the issues surrounding interoperability of PBXs from various vendors?
49. What are some important architectural trends in PBX design and what is the driving force behind these trends?
50. Explain how voice transmission can be integrated with computers to produce a service known as computer telephony integration.
51. What are some of the interoperability issues surrounding CTI?
52. How can the cost of call accounting systems be justified?
53. What are some the PBX features required to support a call accounting system?
54. What are some practical business applications of CTI?
55. What are some potential uses of voice processing or interactive voice response?

Case Study: For a business case study and questions on voice communication concepts and technology, go to www.wiley.com/college/goldman.