8-1 INTRODUCTION

Communications is the process of conveying information from one place to another. Communications requires a source of information, a transmitter, a receiver, a destination, and some form of transmission medium (connecting path) between the transmitter and the receiver. The transmission path may be quite short, as when two people are talking face to face with each other or when a computer is outputting information to a printer located in the same room. *Telecommunications* is long-distance communications (from the Greek word *tele* meaning "distant" or "afar"). Although the word "long" is an arbitrary term, it generally indicates that communications is taking place between a transmitter and a receiver that are too far apart to communicate effectively using only sound waves.

Although often taken for granted, the telephone is one of the most remarkable devices ever invented. To talk to someone, you simply pick up the phone and dial a few digits, and you are almost instantly connected with them. The telephone is one of the simplest devices ever developed, and the telephone connection has not changed in nearly a century. Therefore, a telephone manufactured in the 1920s will still work with today's intricate telephone system.

Although telephone systems were originally developed for conveying human speech information (voice), they are now also used extensively to transport data. This is accomplished using modems that operate within the same frequency band as human voice. Anyone who uses a telephone or a data modem on a telephone circuit is part of a global communications network called the *public telephone network* (PTN). Because the PTN interconnects subscribers through one or more switches, it is sometimes called the *public switched telephone network* (PSTN). The PTN is comprised of several very large corporations and hundreds of smaller independent companies jointly referred to as *Telco*.

The telephone system as we know it today began as an unlikely collaboration of two men with widely disparate personalities: Alexander Graham Bell and Thomas A. Watson. Bell, born in 1847 in Edinburgh, Scotland, emigrated to Ontario, Canada, in 1870, where he lived for only six months before moving to Boston, Massachusetts. Watson was born in a livery stable owned by his father in Salem, Massachusetts. The two met in 1874 and invented the telephone in 1876. On March 10, 1876, one week after his patent was allowed, Bell first succeeded in transmitting speech in his lab at 5 Exeter Place in Boston. At the time, Bell was 29 years old and Watson only 22. Bell's patent, number 174,465, has been called the most valuable ever issued.

The telephone system developed rapidly. In 1877, there were only six telephones in the world. By 1881, 3,000 telephones were producing revenues, and in 1883, there were over 133,000 telephones in the United States alone. Bell and Watson left the telephone business in 1881, as Watson put it, "in better hands." This proved to be a financial mistake, as the telephone company they left evolved into the telecommunications giant known officially as the American Telephone and Telegraph Company (AT&T). Because at one time AT&T owned most of the local operating companies, it was often referred to as the *Bell Telephone System* and sometimes simply as "*Ma Bell.*" By 1982, the Bell System grew to an unbelievable $155 billion in assets ($256 billion in today's dollars), with over one million employees and 100,000 vehicles. By comparison, in 1998, Microsoft's assets were approximately $10 billion.

AT&T once described the Bell System as "the world's most complicated machine." A telephone call could be made from any telephone in the United States to virtually any other telephone in the world using this machine. Although AT&T officially divested the Bell System on January 1, 1983, the telecommunications industry continued to grow at an unbelievable rate. Some estimate that more than 1.5 billion telephone sets are operating in the world today.

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8-2 THE SUBSCRIBER LOOP

The simplest and most straightforward form of telephone service is called *plain old telephone service* (POTS), which involves subscribers accessing the public telephone network through a pair of wires called the *local subscriber loop* (or simply *local loop*). The local loop is the most fundamental component of a telephone circuit. A local loop is simply an unshielded twisted-pair transmission line (cable pair), consisting of two insulated conductors twisted together. The insulating material is generally a polyethylene plastic coating, and the conductor is most likely a pair of 116- to 26-gauge copper wire. A subscriber loop is generally comprised of several lengths of copper wire interconnected at junction and cross-connect boxes located in manholes, back alleys, or telephone equipment rooms within large buildings and building complexes.

The subscriber loop provides the means to connect a telephone set at a subscriber's location to the closest telephone office, which is commonly called an *end office, local exchange office*, or *central office*. Once in the central office, the subscriber loop is connected to an *electronic switching system* (ESS), which enables the subscriber to access the public telephone network. The local subscriber loop is described in greater detail in Chapter 9.

8-3 STANDARD TELEPHONE SET

The word *telephone* comes from the Greek words *tele*, meaning "from afar," and *phone*, meaning "sound," "voice," or "voiced sound." The standard dictionary defines a telephone as follows:

> *An apparatus for reproducing sound, especially that of the human voice (speech), at a great distance, by means of electricity; consisting of transmitting and receiving instruments connected by a line or wire which conveys the electric current.*

In essence, *speech* is sound in motion. However, sound waves are acoustic waves and have no electrical component. The basic telephone set is a simple analog transceiver designed with the primary purpose of converting speech or acoustical signals to electrical signals. However, in recent years, new features such as multiple-line selection, hold, caller ID, and speakerphone have been incorporated into telephone sets, creating a more elaborate and complicated device. However, their primary purpose is still the same, and the basic functions they perform are accomplished in much the same way as they have always been.

The first telephone set that combined a transmitter and receiver into a single handheld unit was introduced in 1878 and called the Butterstamp telephone. You talked into one end and then turned the instrument around and listened with the other end. In 1951, Western Electric Company introduced a telephone set that was the industry standard for nearly four decades (the rotary dial telephone used by your grandparents). This telephone set is called the Bell System 500-type telephone and is shown in Figure 8-1a. The 500-type telephone set replaced the earlier 302-type telephone set (the telephone with the hand-crank magneto, fixed microphone, handheld earphone, and no dialing mechanism). Although there are very few 500-type telephone sets in use in the United States today, the basic functions and operation of modern telephones are essentially the same. In modern-day telephone sets, the rotary dial mechanism is replaced with a Touch-Tone keypad. The modern Touch-Tone telephone is called a 2500-type telephone set and is shown in Figure 8-1b.

The quality of transmission over a telephone connection depends on the received volume, the relative frequency response of the telephone circuit, and the degree of interference. In a typical connection, the ratio of the acoustic pressure at the transmitter input to the corresponding pressure at the receiver depends on the following:

- The translation of acoustic pressure into an electrical signal
- The losses of the two customer local loops, the central telephone office equipment, and the cables between central telephone offices

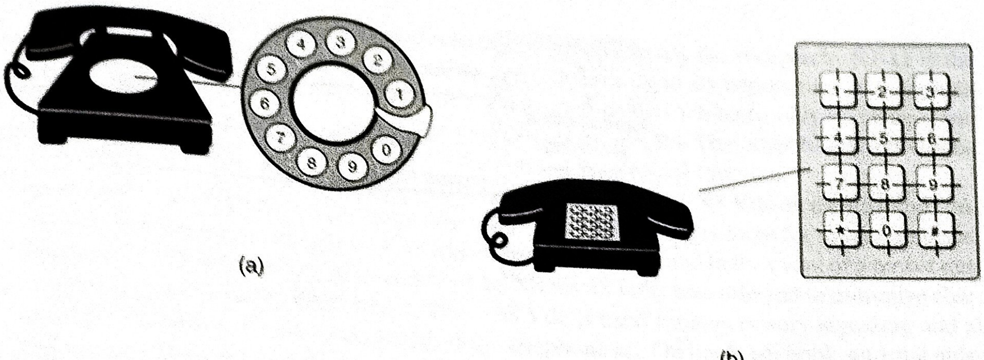


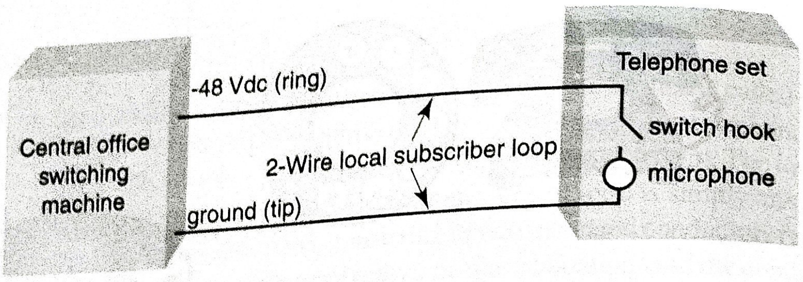
FIGURE 8-1 (a) 500-type telephone set; (b) 2500-type telephone set

The translation of the electrical signal at the receiving telephone set to acoustic pressure at the speaker output

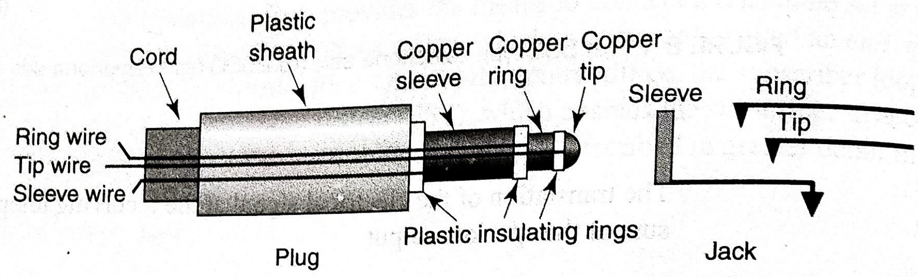
8-3-1 Functions of the Telephone Set

The basic functions of a telephone set are as follows:

1. Notify the subscriber when there is an incoming call with an audible signal, such as a bell, or with a visible signal, such as a flashing light. This signal is analogous to an interrupt signal on a microprocessor, as its intent is to interrupt what you are doing. These signals are purposely made annoying enough to make people want to answer the telephone as soon as possible.
2. Provide a signal to the telephone network verifying when the incoming call has been acknowledged and answered (i.e., the receiver is lifted off hook).
3. Convert speech (acoustical) energy to electrical energy in the transmitter and vice versa in the receiver. Actually, the microphone converts the acoustical energy to mechanical energy, which is then converted to electrical energy. The speaker performs the opposite conversions.
4. Incorporate some method of inputting and sending destination telephone numbers (either mechanically or electrically) from the telephone set to the central office switch over the local loop. This is accomplished using either rotary dialers (pulses) or Touch-Tone pads (frequency tones).
5. Regulate the amplitude of the speech signal the calling person outputs onto the telephone line. This prevents speakers from producing signals high enough in amplitude to interfere with other people's conversations taking place on nearby cable pairs (crosstalk).
6. Incorporate some means of notifying the telephone office when a subscriber wishes to place an outgoing call (i.e., handset lifted off hook). Subscribers cannot dial out until they receive a dial tone from the switching machine.
7. Ensure that a small amount of the transmit signal is fed back to the speaker, enabling talkers to hear themselves speaking. This feedback signal is sometimes called *sidetone* or *talkback*. Sidetone helps prevent the speaker from talking too loudly.
8. Provide an open circuit (idle condition) to the local loop when the telephone is not in use (i.e., on hook) and a closed circuit (busy condition) to the local loop when the telephone is in use (off hook).
9. Provide a means of transmitting and receiving call progress signals between the central office switch and the subscriber, such as on and off hook, busy, ringing, dial pulses, Touch-Tone signals, and dial tone.



(a)



(b)

FIGURE 8-2 (a) Simplified two-wire loop showing telephone set hookup to a local switching machine; (b) plug and jack configurations showing tip, ring, and sleeve

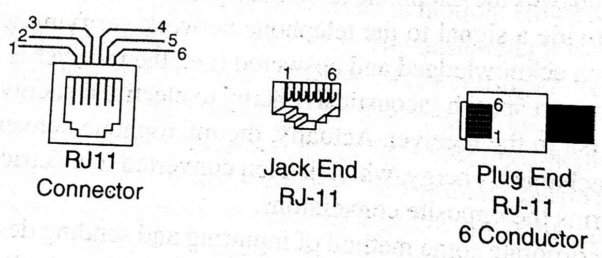


FIGURE 8-3 RJ-11 connector

8-3-2 Telephone Set, Local Loop, and Central Office Switching Machines

Figure 8-2a shows how a telephone set is connected to a central office switching machine (local switch). As shown in the figure, a basic telephone set requires only two wires (one pair) from the telephone company to operate. Again, the pair of wires connecting a subscriber to the closest telephone office is called the *local loop*. One wire on the local loop is called the *tip*, and the other is called the *ring*. The names *tip* and *ring* come from the ¼-inch-diameter two-conductor phone plugs and patch cords used at telephone company switchboards to interconnect and test circuits. The tip and ring for a standard plug and jack are shown in Figure 8-2b. When a third wire is used, it is called the *sleeve*.

Since the 1960s, phone plugs and jacks have gradually been replaced in the home with a miniaturized plastic plug known as RJ-11 and a matching plastic receptacle (shown in Figure 8-3). *RJ* stands for *registered jacks* and is sometimes described as RJ-XX. RJ is a series of telephone connection interfaces (receptacle and plug) that are registered with the U.S. Federal Communications Commission (FCC). The term *jack* sometimes describes both the receptacle and the plug and sometimes specifies only the receptacle. RJ-11 is the most common telephone jack in use today and can have up to six conductors. Although an RJ-11 plug is capable of holding six wires in a $\frac{5}{16}$-inch-by-$\frac{5}{16}$-inch body, only two wires (one pair) are necessary for a standard telephone circuit to operate. The other four wires can be used for a second telephone line and/or for some other special function.

As shown in Figure 8-2a, the switching machine outputs −48 Vdc on the ring and connects the tip to ground. A dc voltage was used rather than an ac voltage for several reasons: (1) to prevent power supply hum, (2) to allow service to continue in the event of a power outage, and (3) because people were afraid of ac. Minus 48 volts was selected to minimize electrolytic corrosion on the loop wires. The −48 Vdc is used for supervisory signaling and to provide talk battery for the microphone in the telephone set. On hook, off hook, and dial pulsing are examples of supervisory signals and are described in a later section of this chapter. It should be noted that −48 Vdc is the only voltage required for the operation of a standard telephone. However, most modern telephones are equipped with nonstandard (and often nonessential) features and enhancements and may require an additional source of ac power.

8-3-3 Block Diagram of a Telephone Set

A standard telephone set is comprised of a transmitter, a receiver, an electrical network for equalization, associated circuitry to control sidetone levels and to regulate signal power, and necessary signaling circuitry. In essence, a telephone set is an apparatus that creates an exact likeness of sound waves with an electric current. Figure 8-4 shows the functional block diagram of a *telephone set.* The essential components of a telephone set are the ringer circuit, on/off hook circuit, equalizer circuit, hybrid circuit, speaker, microphone, and a dialing circuit.

8-3-3-1 Ringer circuit. The telephone *ringer* has been around since August 1, 1878, when Thomas Watson filed for the first ringer patent. The *ringer circuit,* which was originally an electromagnetic bell, is placed directly across the tip and ring of the local loop. The purpose of the ringer is to alert the destination party of incoming calls. The audible tone from the ringer must be loud enough to be heard from a reasonable distance and offensive enough to make a person want to answer the telephone as soon as possible. In modern telephones, the bell has been replaced with an electronic oscillator connected to the speaker. Today, ringing signals can be any imaginable sound, including buzzing, beeping, chiming, or your favorite melody.

8-3-3-2 On/off hook circuit. The *on/off hook circuit* (sometimes called a *switch hook*) is nothing more than a simple single-throw, double-pole (STDP) switch placed across

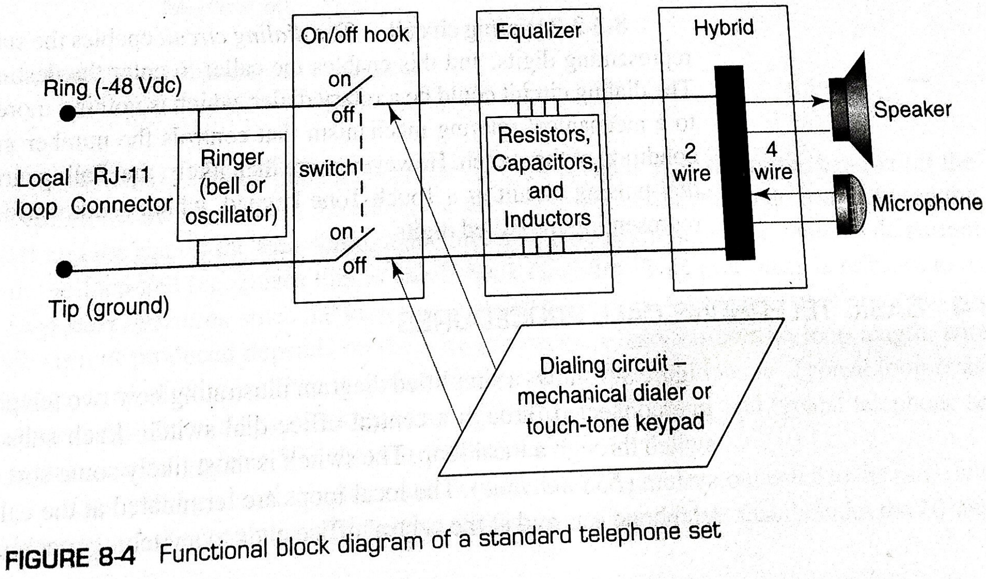


FIGURE 8-4 Functional block diagram of a standard telephone set

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the tip and ring. The switch is mechanically connected to the telephone handset so that when the telephone is idle (on hook), the switch is open. When the telephone is in use (off hook), the switch is closed, completing an electrical path through the microphone between the tip and ring of the local loop.

8-3-3-3 Equalizer circuit. *Equalizers* are combinations of passive components (resistors, capacitors, and so on) that are used to regulate the amplitude and frequency response of the voice signals. The equalizer helps solve an important transmission problem in telephone set design, namely, the interdependence of the transmitting and receiving efficiencies and the wide range of transmitter currents caused by a variety of local loop cables with different dc resistances.

8-3-3-4 Speaker. In essence, the *speaker* is the receiver for the telephone. The speaker converts electrical signals received from the local loop to acoustical signals (sound waves) that can be heard and understood by a human being. The speaker is connected to the local loop through the hybrid network. The speaker is typically enclosed in the *handset* of the telephone along with the microphone.

8-3-3-5 Microphone. For all practical purposes, the *microphone* is the transmitter for the telephone. The microphone converts acoustical signals in the form of sound pressure waves from the caller to electrical signals that are transmitted into the telephone network through the local subscriber loop. The microphone is also connected to the local loop through the hybrid network. Both the microphone and the speaker are transducers, as they convert one form of energy into another form of energy. A microphone converts acoustical energy first to mechanical energy and then to electrical energy, while the speaker performs the exact opposite sequence of conversions.

8-3-3-6 Hybrid network. The *hybrid network* (sometimes called a *hybrid coil* or *duplex coil*) in a telephone set is a special balanced transformer used to convert a two-wire circuit (the local loop) into a four-wire circuit (the telephone set) and vice versa, thus enabling full-duplex operation over a two-wire circuit. In essence, the hybrid network separates the transmitted signals from the received signals. Outgoing voice signals are typically in the 1-V to 2-V range, while incoming voice signals are typically half that value. Another function of the hybrid network is to allow a small portion of the transmit signal to be returned to the receiver in the form of a *sidetone*. Insufficient sidetone causes the speaker to raise his voice, making the telephone conversation seem unnatural. Too much sidetone causes the speaker to talk too softly, thereby reducing the volume that the listener receives.

8-3-3-7 Dialing circuit. The *dialing circuit* enables the subscriber to output signals representing digits, and this enables the caller to enter the destination telephone number. The dialing circuit could be a rotary dialer, which is nothing more than a switch connected to a mechanical rotating mechanism that controls the number and duration of the on/off condition of the switch. However, more than likely, the dialing circuit is either an electronic dial-pulsing circuit or a Touch-Tone keypad, which sends various combinations of tones representing the called digits.

8-4 BASIC TELEPHONE CALL PROCEDURES

Figure 8-5 shows a simplified diagram illustrating how two telephone sets (subscribers) are interconnected through a central office dial switch. Each subscriber is connected to the switch through a local loop. The switch is most likely some sort of an electronic switching system (*ESS machine*). The local loops are terminated at the calling and called stations in telephone sets and at the central office ends to switching machines.

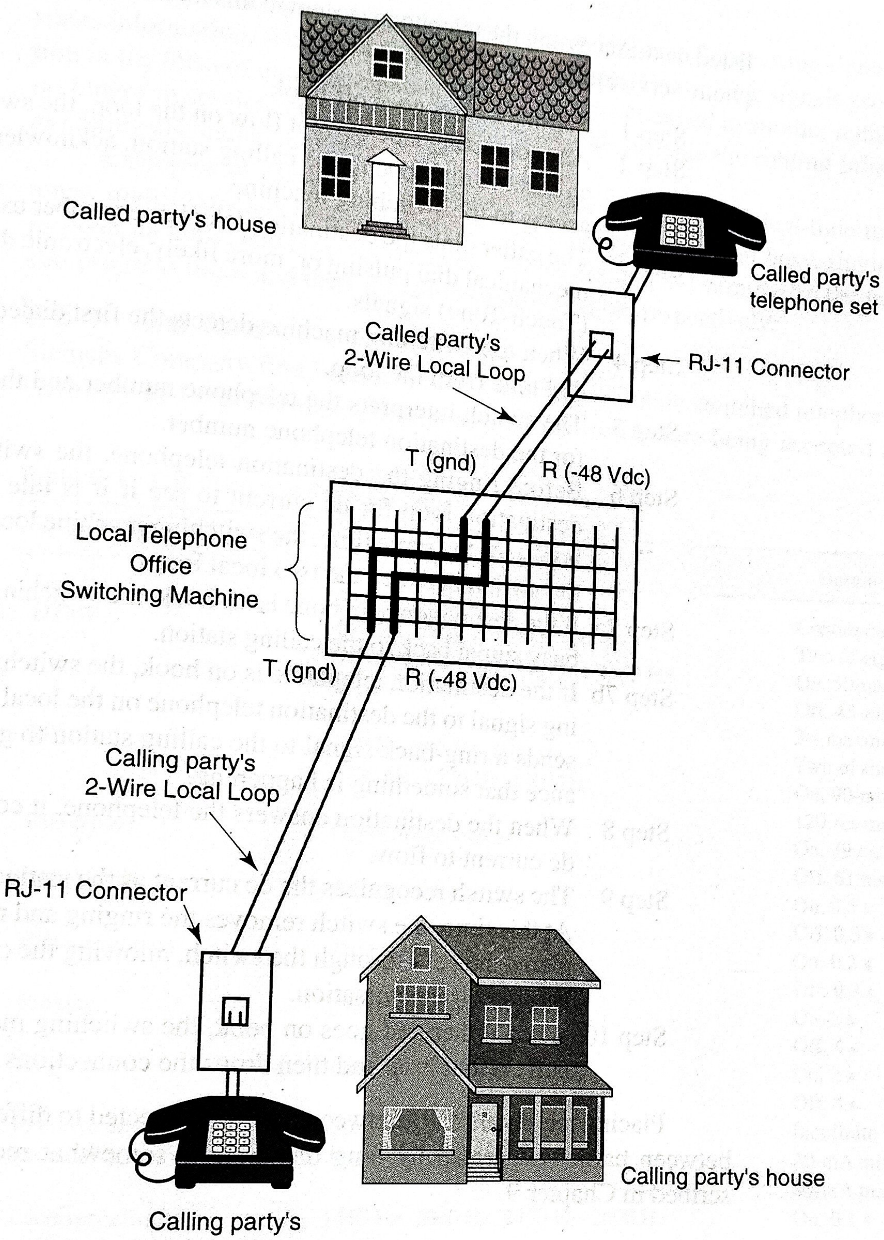

FIGURE 8-5 Telephone call procedures

When the calling party's telephone set goes off hook (i.e., lifting the handset off the cradle), the switch hook in the telephone set is released, completing a dc path between the tip and the ring of the loop through the microphone. The ESS machine senses a dc current in the loop and recognizes this as an off-hook condition. This procedure is referred to as *loop start operation* since the loop is completed through the telephone set. The amount of dc current produced depends on the wire resistance, which varies with loop length, wire gauge, type of wire, and the impedance of the subscriber's telephone. Typical loop resistance ranges from a few ohms up to approximately 1300 ohms, and typical telephone set impedances range from 500 ohms to 1000 ohms.

Completing a local telephone call between two subscribers connected to the same telephone switch is accomplished through a standard set of procedures that includes the 10 steps listed next. Accessing the telephone system in this manner is known as POTS (plain old telephone service):

Step 1 Calling station goes off hook.

Step 2 After detecting a dc current flow on the loop, the switching machine returns an audible dial tone to the calling station, acknowledging that the caller has access to the switching machine.

Step 3 The caller dials the destination telephone number using one of two methods: mechanical dial pulsing or, more likely, electronic dual-tone multifrequency (Touch-Tone) signals.

Step 4 When the switching machine detects the first dialed number, it removes the dial tone from the loop.

Step 5 The switch interprets the telephone number and then locates the local loop for the destination telephone number.

Step 6 Before ringing the destination telephone, the switching machine tests the destination loop for dc current to see if it is idle (on hook) or in use (off hook). At the same time, the switching machine locates a signal path through the switch between the two local loops.

Step 7a If the destination telephone is off hook, the switching machine sends a station-busy signal back to the calling station.

Step 7b If the destination telephone is on hook, the switching machine sends a ringing signal to the destination telephone on the local loop and at the same time sends a ring-back signal to the calling station to give the caller some assurance that something is happening.

Step 8 When the destination answers the telephone, it completes the loop, causing dc current to flow.

Step 9 The switch recognizes the dc current as the station answering the telephone. At this time, the switch removes the ringing and ring-back signals and completes the path through the switch, allowing the calling and called parties to begin their conversation.

Step 10 When either end goes on hook, the switching machine detects an open circuit on that loop and then drops the connections through the switch.

Placing telephone calls between parties connected to different switching machines or between parties separated by long distances is somewhat more complicated and is described in Chapter 9.

8-5 CALL PROGRESS TONES AND SIGNALS

Call progress tones and *call progress signals* are acknowledgment and status signals that ensure the processes necessary to set up and terminate a telephone call are completed in an orderly and timely manner. Call progress tones and signals can be sent from machines to machines, machines to people, and people to machines. The people are the subscribers (i.e., the calling and the called party), and the machines are the electronic switching systems in the telephone offices and the telephone sets themselves. When a switching machine outputs a call progress tone to a subscriber, it must be audible and clearly identifiable.

Signaling can be broadly divided into two major categories: *station signaling* and *interoffice signaling*. Station signaling is the exchange of signaling messages over local loops between stations (telephones) and telephone company switching machines. On the other hand, interoffice signaling is the exchange of signaling messages between switching machines. Signaling messages can be subdivided further into one of four categories: *alerting*, *supervising*, *controlling*, and *addressing*. Alerting signals indicate a request for service, When a telephone is *on hook*, it is [illegible] telephone *state*. The term *on hook* was derived in the early days of [illegible] when the handset was literally placed on a hook (the hook eventually evolved into a cradle). When the telephone set is on hook, the local loop is open, and there is no current flowing on the loop. An on-hook signal is also used to terminate a call and initiate a disconnect.

When the telephone set is taken *off hook*, a switch closes in the telephone that completes a dc path between the two wires of the local loop. The switch closure causes a dc current to flow on the loop (nominally between 20 mA and 80 mA, depending on loop length and wire gauge). The switching machine in the central office detects the dc current and recognizes it as a receiver off-hook condition (sometimes called a *seizure* or *request for service*). The receiver off-hook condition is the first step to completing a telephone call. The switching machine will respond to the off-hook condition by placing an audible dial tone on the loop. The off-hook signal is also used at the destination end as an *answer signal* to indicate that the called party has answered the telephone. This is sometimes referred to as a *ring trip* because when the switching machine detects the off-hook condition, it removes (or trips) the ringing signal.

8-5-10 Other Nonessential Signaling and Call Progress Tones

There are numerous additional signals relating to initiating, establishing, completing, and terminating a telephone call that are nonessential, such as *call waiting tones, caller waiting tones, calling card service tones, comfort tones, hold tones, intrusion tones, stutter dial tone* (for voice mail), and *receiver off-hook tones* (also called *howler tones*).

CORDLESS TELEPHONES

Cordless telephones are simply telephones that operate without cords attached to the handset. Cordless telephones originated around 1980 and were quite primitive by today's standards. They originally occupied a narrow band of frequencies near 1.7 MHz, just above the AM broadcast band, and used the 117-vac, 60-Hz household power line for an antenna. These early units used frequency modulation (FM) and were poor quality and susceptible to interference from fluorescent lights and automobile ignition systems. In 1984, the FCC reallocated cordless telephone service to the 46-MHz to 49-MHz band. In 1990, the FCC extended cordless telephone service to the 902-MHz to 928-MHz band, which appreciated a superior signal-to-noise ratio. Cordless telephone sets transmit and receive over narrowband FM (NBFM) channels spaced 30 kHz to 100 kHz apart, depending on the modulation and frequency band used. In 1998, the FCC expanded service again to the 2.4-GHz band. Adaptive differential pulse code modulation and spread spectrum technology (SST) are used exclusively in the 2.4-GHz band, while FM and SST digital modulation are used in the 902-MHz to 928-MHz band. Digitally modulated SST telephones offer higher quality and more security than FM telephones.

In essence, a cordless telephone is a full-duplex, battery-operated, portable radio transceiver that communicates directly with a stationary transceiver located somewhere in

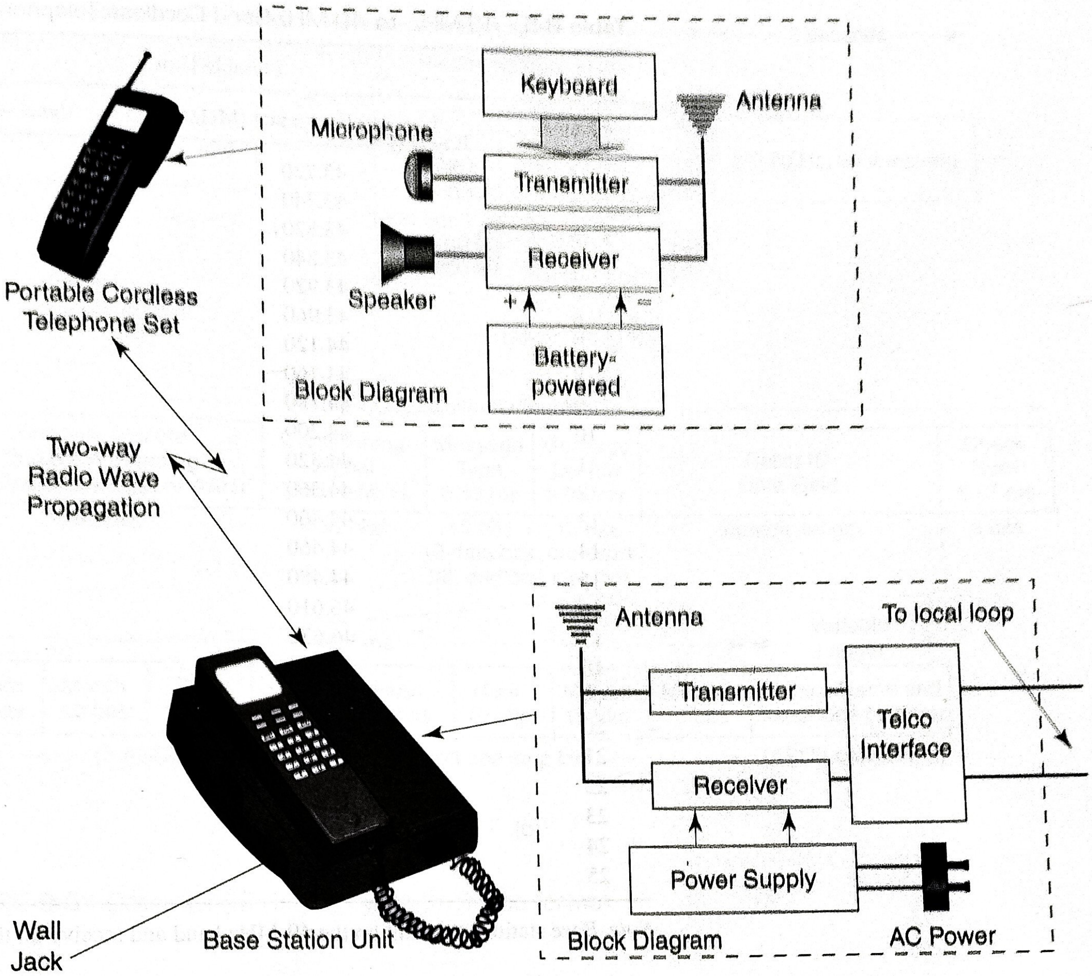

FIGURE 8-8 Cordless telephone system

the subscriber's home or office. The basic layout for a cordless telephone is shown in Figure 8-8. The base station is an ac-powered stationary radio transceiver (transmitter and receiver) connected to the local loop through a cord and telephone company interface unit. The interface unit functions in much the same way as a standard telephone set in that its primary function is to interface the cordless telephone with the local loop while being transparent to the user. Therefore, the base station is capable of transmitting and receiving both supervisory and voice signals over the subscriber loop in the same manner as a standard telephone. The base station must also be capable of relaying voice and control signals to and from the portable telephone set through the wireless transceiver. In essence, the portable telephone set is a battery-powered, two-way radio capable of operating in the full-duplex mode.

Because a portable telephone must be capable of communicating with the base station in the full-duplex mode, it must transmit and receive at different frequencies. In 1984, the FCC allocated 10 full-duplex channels for 46-MHz to 49-MHz units. In 1995 to help relieve congestion, the FCC added 15 additional full-duplex channels and extended the frequency band to include frequencies in the 43-MHz to 44-MHz band. Base stations transmit on high-band frequencies and receive on low-band frequencies, while the portable unit transmits on low-band frequencies and receives on high-band frequencies. The frequency assignments are listed in Table 8-5. Channels 16 through 25 are the original 10 full-duplex carrier frequencies. The maximum transmit power for both the portable unit and the base station is 500 mW. This stipulation limits the useful range of a cordless telephone to within 100 feet or less of the base station.

Table 8-5 43-MHz- to 49-MHz-Band Cordless Telephone Frequencies

	Portable Unit	
Channel	Transmit Frequency (MHz)	Receive Frequency (MHz)
1	43.720	48.760
2	43.740	48.840
3	43.820	48.860
4	43.840	48.920
5	43.920	49.920
6	43.960	49.080
7	44.120	49.100
8	44.160	49.160
9	44.180	49.200
10	44.200	49.240
11	44.320	49.280
12	44.360	49.360
13	44.400	49.400
14	44.460	49.460
15	44.480	49.500
16	46.610	49.670
17	46.630	49.845
18	46.670	49.860
19	46.710	49.770
20	46.730	49.875
21	46.770	49.830
22	46.830	49.890
23	46.870	49.930
24	46.930	49.970
25	46.970	49.990

Note. Base stations transmit on the 49-MHz band and receive on the 46-MHz band.

Cordless telephones using the 2.4-GHz band offer excellent sound quality utilizing digital modulation and twin-band transmission to extend their range. With twin-band transmission, base stations transmit in the 2.4-GHz band, while portable units transmit in the 902-MHz to 928-MHz band.

8-7 CALLER ID

Caller ID (identification) is a service originally envisioned by AT&T in the early 1970s, although local telephone companies have only recently offered it. The basic concept of caller ID is quite simple. Caller ID enables the destination station of a telephone call to display the name and telephone number of the calling party before the telephone is answered (i.e., while the telephone is ringing). This allows subscribers to screen incoming calls and decide whether they want to answer the telephone.

The caller ID message is a simplex transmission sent from the central office switch over the local loop to a caller ID display unit at the destination station (no response is provided). The caller ID information is transmitted and received using Bell System 202-compatible modems (ITU V.23 standard). This standard specifies a 1200-bps FSK (frequency-shift keying) signal with a 1200-Hz mark frequency (f_m) and a 2200-Hz space frequency (f_m). The FSK signal is transmitted in a burst between the first and second 20-Hz, 90-Vrms ringing signals as shown in Figure 8-9a. Therefore, to ensure detection of the caller ID signal, the telephone must ring at least twice before being answered. The caller ID signal does not begin until 500 ms after the end of the first ring and must end 500 ms before the beginning of the second ring. Therefore, the caller ID signal has a 3-second window in which it must be transmitted.

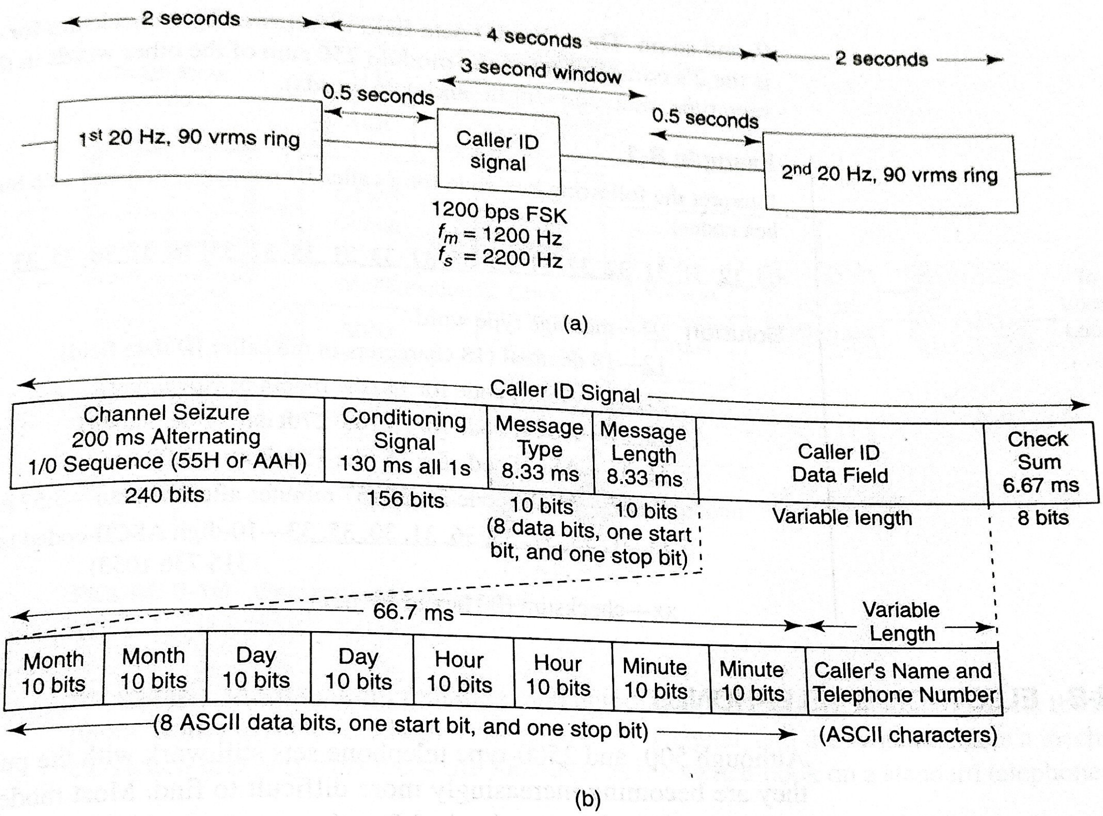

FIGURE 8-9 Caller ID: (a) ringing cycle; (b) frame format

The format for a caller ID signal is shown in Figure 8-9b. The 500-ms delay after the first ringing signal is immediately followed by the *channel seizure field,* which is a 200-ms-long sequence of alternating logic 1s and logic 0s (240 bits comprised of 120 pairs of alternating 1/0 bits, either 55 hex or AA hex). A *conditioning signal field* immediately follows the channel seizure field. The conditioning signal is a continuous 1200-Hz tone lasting for 130 ms, which equates to 156 consecutive logic 1 bits.

The protocol used for the next three fields—*message type field, message length field,* and *caller ID data field*—specifies asynchronous transmission of 16-bit characters (without parity) framed by one start bit (logic 0) and one stop bit (logic 1) for a total of 10 bits per character. The message type field is comprised of a 16-bit hex code, indicating the type of service and capability of the data message. There is only one message type field currently used with caller ID (04 hex). The message type field is followed by a 16-bit message length field, which specifies the total number of characters (in binary) included in the caller ID data field. For example, a message length code of 15 hex (0001 0101) equates to the number 21 in decimal. Therefore, a message length code of 15 hex specifies 21 characters in the caller ID data field.

The caller ID data field uses extended ASCII coded characters to represent a month code (01 through 12), a two-character day code (01 through 31), a two-character hour code in local military time (00 through 23), a two-character minute code (00 through 59), and a variable-length code, representing the caller's name and telephone number. ASCII coded digits are comprised of two independent hex characters (eight bits each). The first hex character is always 3 (0011 binary), and the second hex character represents a digit between 0 and 9 (0000 to 1001 binary). For example, 30 hex (0011 0000 binary) equates to the digit 0, 31 hex (0011 0001 binary) equates to the digit 1, 39 hex (0011 1001) equates to the digit

9, and so on. The caller ID data field is followed by a checksum for error detection, which is the 2's complement of the module 256 sum of the other words in the data message (message type, message length, and data words).

Example 8-1

Interpret the following hex code for a caller ID message (start and stop bits are not included in the hex codes):

04 12 31 31 32 37 31 35 35 37 33 31 35 37 33 36 31 30 35 33 xx

Solution 04—message type word

12—18 decimal (18 characters in the caller ID data field)

31, 31—ASCII code for 11 (the month of November)

32, 37—ASCII code for 27 (the 27th day of the month)

31, 35—ASCII code for 15 (the 15th hour—3:00 P.M.)

35, 37—ASCII code for 57 (57 minutes after the hour—3:57 P.M.)

33, 31, 35, 37, 33, 36, 31, 30, 35, 33—10-digit ASCII-coded telephone number (315 736 1053)

xx—checksum (00 hex to FF hex)

8-8 ELECTRONIC TELEPHONES

Although 500- and 2500-type telephone sets still work with the public telephone network, they are becoming increasingly more difficult to find. Most modern-day telephones have replaced many of the mechanical functions performed in the old telephone sets with electronic circuits. Electronic telephones use integrated-circuit technology to perform many of the basic telephone functions as well as a myriad of new and, and in many cases, nonessential functions. The refinement of microprocessors has also led to the development of multiple-line, full-feature telephones that permit automatic control of the telephone set's features, including telephone number storage, automatic dialing, redialing, and caller ID. However, no matter how many new gadgets are included in the new telephone sets, they still have to interface with the telephone network in much the same manner as telephones did a century ago.

Figure 8-10 shows the block diagram for a typical electronic telephone comprised of one multifunctional integrated-circuit chip, a microprocessor chip, a Touch-Tone keypad, a speaker, a microphone, and a handful of discrete devices. The major components included in the multifunctional integrated circuit chip are DTMF tone generator, MPU (microprocessor unit) interface circuitry, random access memory (RAM), tone ringer circuit, speech network, and a line voltage regulator.

The Touch-Tone keyboard provides a means for the operator of the telephone to access the DTMF tone generator inside the multifunction integrated-circuit chip. The external crystal provides a stable and accurate frequency reference for producing the dual-tone multifrequency signaling tones.

The tone ringer circuit is activated by the reception of a 20-Hz ringing signal. Once the ringing signal is detected, the tone ringer drives a piezoelectric sound element that produces an electronic ring (without a bell).

The voltage regulator converts the dc voltage received from the local loop and converts it to a constant-level dc supply voltage to operate the electronic components in the telephone. The internal speech network contains several amplifiers and associated components that perform the same functions as the hybrid did in a standard telephone.

The microprocessor interface circuit interfaces the MPU to the multifunction chip. The MPU, with its internal RAM, controls many of the functions of the telephone, such as num-

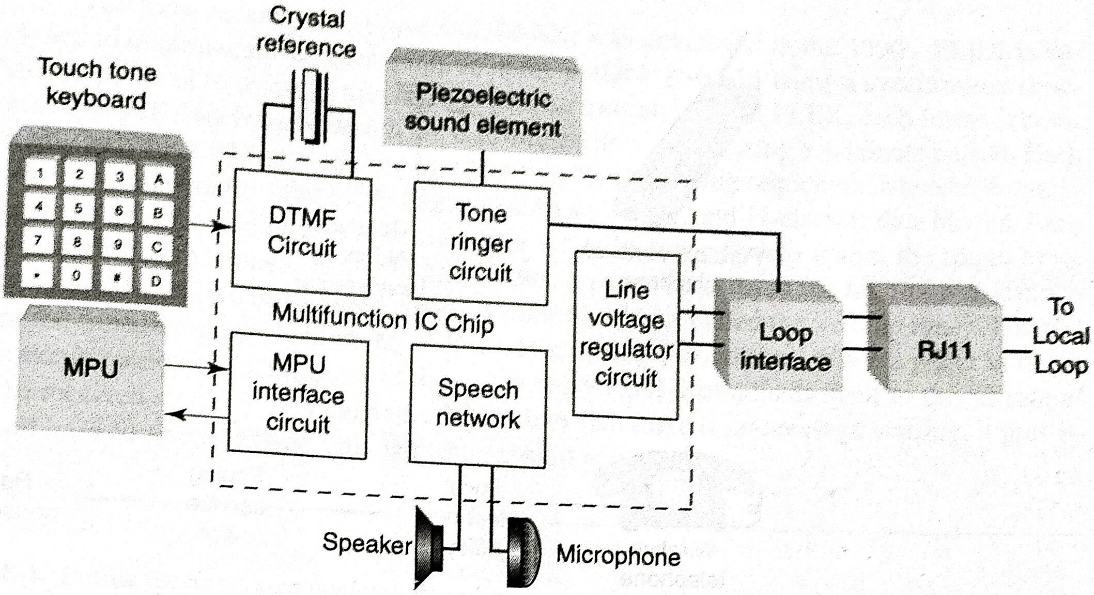

FIGURE 8-10 Electronic telephone set

ber storage, speed dialing, redialing, and autodialing. The bridge rectifier protects the telephone from the relatively high-voltage ac ringing signal, and the switch hook is a mechanical switch that performs the same functions as the switch hook on a standard telephone set.

8-9 PAGING SYSTEMS

Most *paging systems* are simplex wireless communications systems designed to alert subscribers of awaiting messages. Paging transmitters relay radio signals and messages from wire-line and cellular telephones to subscribers carrying portable receivers. The simplified block diagram of a paging system is shown in Figure 8-11. The infrastructure used with paging systems is somewhat different than the one used for cellular telephone systems. This is because standard paging systems are one way, with signals transmitted from the paging system to portable pager and never in the reverse direction. There are narrow-, mid-, and wide-area pagers (sometimes called local, regional, and national). Narrow-area paging systems operate only within a building or building complex, mid-area pagers cover an area of several square miles, and wide-area pagers operate worldwide. Most pagers are mid-area where one centrally located high-power transmitter can cover a relatively large geographic area, typically between 6 and 10 miles in diameter.

To contact a person carrying a pager, simply dial the telephone number assigned that person's portable pager. The paging company receives the call and responds with a query requesting the telephone number you wish the paged person to call. After the number is entered, a *terminating signal* is appended to the number, which is usually the # sign. The caller then hangs up. The paging system converts the telephone number to a digital code and transmits it in the form of a digitally encoded signal over a wireless communications system. The signal may be simultaneously sent from more than one radio transmitter (sometimes called *simulcasting* or *broadcasting*), as is necessary in a wide-area paging system. If the paged person is within range of a broadcast transmitter, the targeted pager will receive the message. The message includes a notification signal, which either produces an audible beep or causes the pager to vibrate, and the number the paged unit should call is shown on an alphanumeric display. Some newer paging units are also capable of displaying messages as well as the telephone number of the paging party.

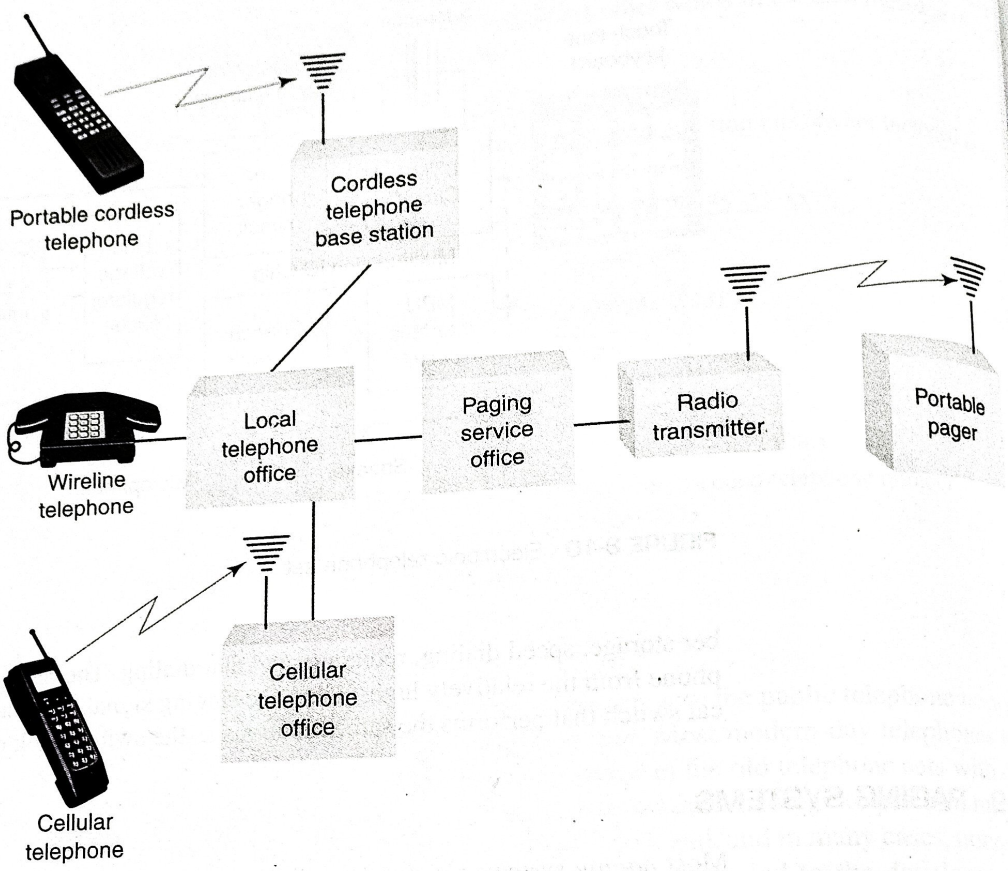

FIGURE 8-11 Simplified block diagram of a standard simplex paging system

Early paging systems used FM; however, most modern paging systems use FSK or PSK. Pagers typically transmit bit rates between 200 bps and 6400 bps with the following carrier frequency bands: 138 MHz to 175 MHz, 267 MHz to 284 MHz, 310 MHz to 330 MHz, 420 MHz to 470 MHz, and several frequency slots within the 900-MHz band.

Each portable pager is assigned a special code, called a *cap code,* which includes a sequence of digits or a combination of digits and letters. The cap code is broadcast along with the paging party's telephone number. If the portable paging unit is within range of the broadcasting transmitter, it will receive the signal, demodulate it, and recognize its cap code. Once the portable pager recognizes its cap code, the callback number and perhaps a message will be displayed on the unit. Alphanumeric messages are generally limited to between 20 and 40 characters in length.

Early paging systems, such as one developed by the British Post Office called Post Office Code Standardization Advisory Group (POCSAG), transmitted a two-level FSK signal. POCSAG used an asynchronous protocol, which required a long preamble for synchronization. The preamble begins with a long *dotting sequence* (sometimes called a *dotting comma*) to establish clock synchronization. Data rates for POCSAG are 512 bps, 1200 bps, and 2400 bps. With POCSAG, portable pagers must operate in the *always-on mode* all the time, which means the pager wastes much of its power resources on nondata preamble bits.

In the early 1980s, the European Telecommunications Standards Institute (ETSI) developed the ERMES protocol. ERMES transmitted data at a 6250 bps rate using four-level FSK (3125 baud). ERMES is a synchronous protocol, which requires less time to synchronize. ERMES supports 16 25-kHz paging channels in each of its frequency bands.

9-1 INTRODUCTION

A *telephone circuit* is comprised of two or more facilities, interconnected in tandem, to provide a transmission path between a source and a destination. The interconnected facilities may be temporary, as in a standard telephone call, or permanent, as in a dedicated private-line telephone circuit. The facilities may be metallic cable pairs, optical fibers, or wireless carrier systems. The information transferred is called the *message*, and the circuit used is called the *message channel*.

Telephone companies offer a wide assortment of message channels ranging from a basic 4-kHz voice-band circuit to wideband microwave, satellite, or optical fiber transmission systems capable of transferring high-resolution video or wideband data. The following discussion is limited to basic voice-band circuits. In telephone terminology, the word *message* originally denoted speech information. However, this definition has been extended to include any signal that occupies the same bandwidth as a standard voice channel. Thus, a message channel may include the transmission of ordinary speech, supervisory signals, or data in the form of digitally modulated carriers (FSK, PSK, QAM, and so on). The network bandwidth for a standard voice-band message channel is 4 kHz; however, a portion of that bandwidth is used for *guard bands* and signaling. Guard bands are unused frequency bands located between information signals. Consequently, the effective channel bandwidth for a voice-band message signal (whether it be voice or data) is approximately 300 Hz to 3000 Hz.

THE LOCAL SUBSCRIBER LOOP

The *local subscriber loop* is the only facility required by all voice-band circuits, as it is the means by which subscriber locations are connected to the local telephone company. In essence, the sole purpose of a local loop is to provide subscribers access to the public telephone network. The local loop is a metallic transmission line comprised of two insulated copper wires (a pair) twisted together. The local loop is the primary cause of *attenuation* and *phase distortion* on a telephone circuit. Attenuation is an actual loss of signal strength, and phase distortion occurs when two or more frequencies undergo different amounts of phase shift.

The *transmission characteristics* of a cable pair depend on the wire diameter, conductor spacing, dielectric constant of the insulator separating the wires, and the conductivity of the wire. These physical properties, in turn, determine the inductance, resistance, capacitance, and conductance of the cable. The resistance and inductance are distributed along the length of the wire, whereas the conductance and capacitance exist between the two wires. When the insulation is sufficient, the effects of conductance are generally negligible. Figure 9-1a shows the electrical model for a copper-wire transmission line.

The electrical characteristics of a cable (such as inductance, capacitance, and resistance) are uniformly distributed along its length and are appropriately referred to as *distributed parameters*. Because it is cumbersome working with distributed parameters, it is common practice to lump them into discrete values per unit length (i.e., millihenrys per mile, microfarads per kilometer, or ohms per 1000 feet). The amount of attenuation and phase delay experienced by a signal propagating down a metallic transmission line is a function of the frequency of the signal and the electrical characteristics of the cable pair.

There are seven main component parts that make up a traditional local loop:

Feeder cable (F1). The largest cable used in a local loop, usually 3600 pair of copper wire placed underground or in conduit.

Serving area interface (SAI). A cross-connect point used to distribute the larger feeder cable into smaller distribution cables.

Biomi ① GET: HTTP (standard) Put, delete
② POST (Response)

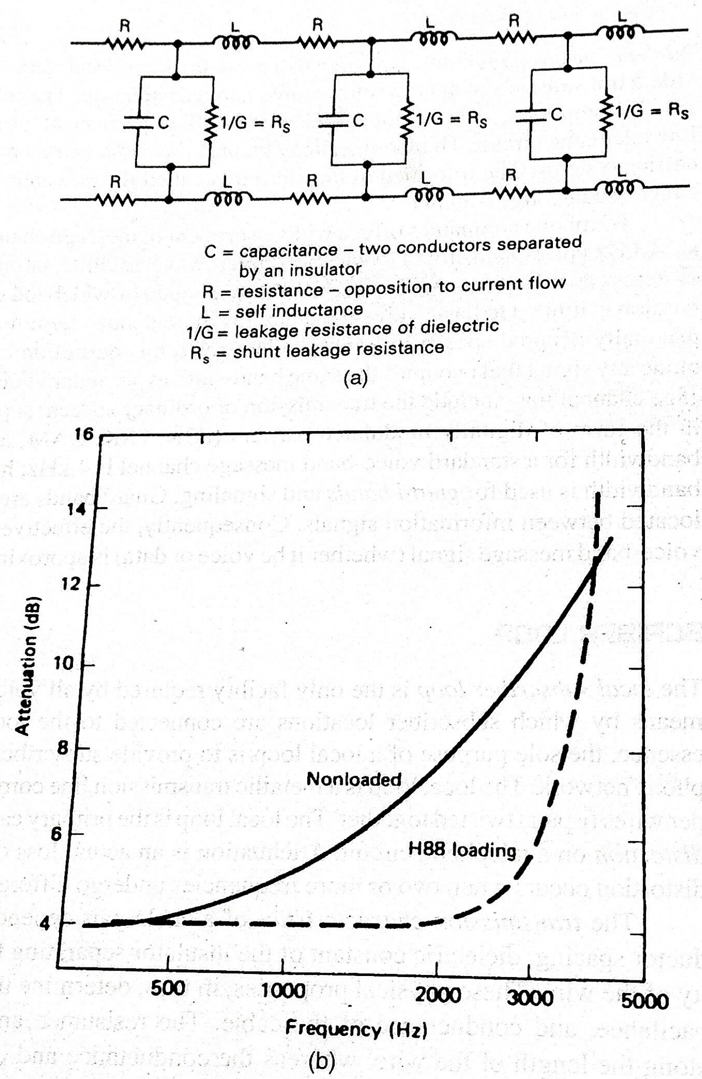

FIGURE 9-1 (a) Electrical model of a copper-wire transmission line; (b) frequency-versus-attenuation characteristics for unloaded and loaded cables

Distribution cable (F2). A smaller version of a feeder cable containing less wire pairs.

Subscriber or standard network interface (SNI). A device that serves as the demarcation point between local telephone company responsibility and subscriber responsibility for telephone service.

Drop wire. The final length of cable pair that terminates at the SNI.

Aerial. That portion of the local loop that is strung between poles.

Distribution cable and drop-wire cross-connect point. The location where individual cable pairs within a distribution cable are separated and extended to the subscriber's location on a drop wire.

Two components often found on local loops are loading coils and bridge taps.

of a local loop using 19-gauge wire ... wire.

The total attenuation of a local loop is generally limited to a maximum value of 7.5 dB with a maximum dc resistance of 1300 Ω, which includes the resistance of the telephone (approximately 120 Ω). The dc resistance of 26-gauge copper wire is approximately 41 Ω per 1000 feet, which limits the round-trip loop length to approximately 5.6 miles. The maximum distance for lower-gauge wire is longer of course.

The dc loop resistance for copper conductors is approximated by

$$R_{dc} = \frac{0.1095}{d^2} \tag{9-1}$$

where R_{dc} = dc loop resistance (ohms per mile)
d = wire diameter (inches)

9-3 TELEPHONE MESSAGE-CHANNEL NOISE AND NOISE WEIGHTING

The *noise* that reaches a listener's ears affects the degree of annoyance to the listener and, to some extent, the intelligibility of the received speech. The total noise is comprised of room *background noise* and noise introduced in the circuit. Room background noise on the listening subscriber's premises reaches the ear directly through leakage around the receiver and indirectly by way of the sidetone path through the telephone set. Room noise from the talking subscriber's premises also reaches the listener over the communications channel. Circuit noise is comprised mainly of thermal noise, nonlinear distortion, and impulse noise, which are described in a later section of this chapter.

The measurement of interference (noise), like the measurement of volume, is an effort to characterize a complex signal. Noise measurements on a telephone message channel are characterized by how annoying the noise is to the subscriber rather than by the absolute magnitude of the average noise power. Noise interference is comprised of two components: annoyance and the effect of noise on intelligibility, both of which are functions of frequency. Noise signals with equal interfering effects are assigned equal magnitudes. To accomplish this effect, the American Telephone and Telegraph Company (AT&T) developed a weighting network called *C-message* weighting.

When designing the C-message weighting network, groups of observers were asked to adjust the loudness of 14 different frequencies between 180 Hz and 3500 Hz until the sound of each tone was judged to be equally annoying as a 1000-Hz reference tone in the absence of speech. A 1000-Hz tone was selected for the reference because empirical data indicated that 1000 Hz is the most annoying frequency (i.e., the best frequency response) life Cycle (SDLC)

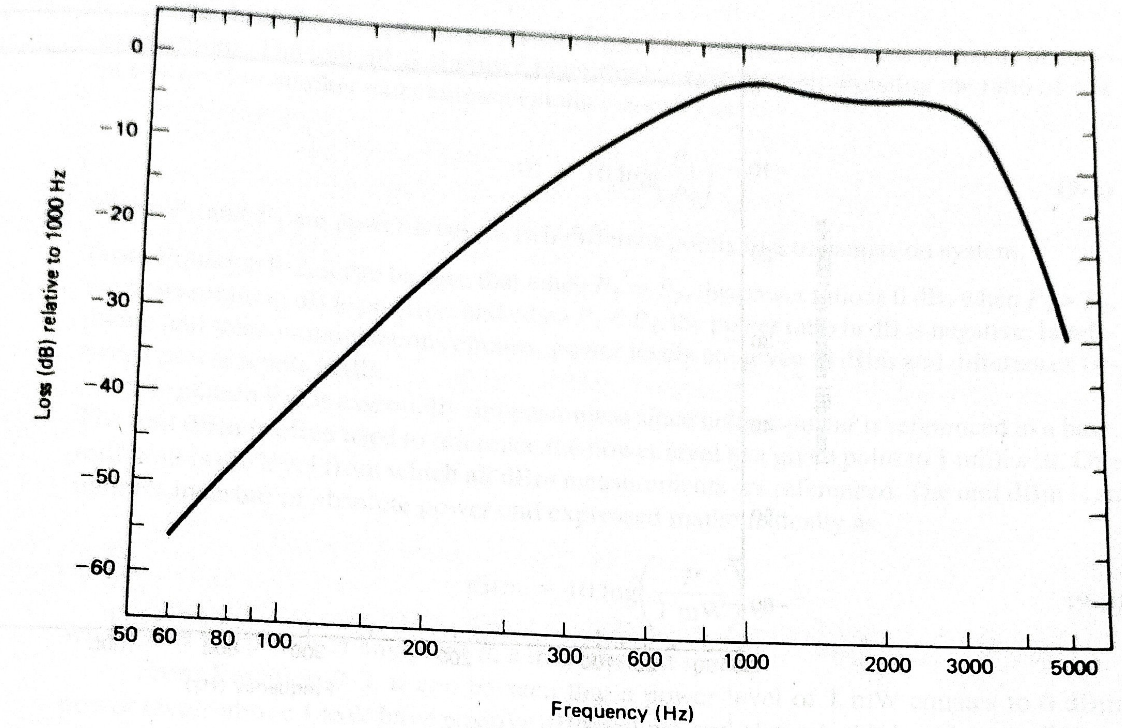

FIGURE 9-2 C-message weighting curve

to humans. The same people were then asked to adjust the amplitude of the tones in the presence of speech until the effect of noise on articulation (annoyance) was equal to that of the 1000-Hz reference tone. The results of the two experiments were combined, smoothed, and plotted, resulting in the C-message weighting curve shown in Figure 9-2. A 500-type telephone set was used for these tests; therefore, the C-message weighting curve includes the frequency response characteristics of a standard telephone set receiver as well as the hearing response of an average listener.

The significance of the C-message weighting curve is best illustrated with an example. From Figure 9-2, it can be seen that a 200-Hz test tone of a given power is 25 dB less disturbing than a 1000-Hz test tone of the same power. Therefore, the C-message weighting network will introduce 25 dB more loss for 200 Hz than it will for 1000 Hz.

When designing the C-message network, it was found that the additive effect of several noise sources combine on a root-sum-square (RSS) basis. From these design considerations, it was determined that a telephone message-channel noise measuring set should be a voltmeter with the following characteristics:

Readings should take into consideration that the interfering effect of noise is a function of frequency as well as magnitude.

When dissimilar noise signals are present simultaneously, the meter should combine them to properly measure the overall interfering effect.

It should have a transient response resembling that of the human ear. For sounds shorter than 200 ms, the human ear does not fully appreciate the true power of the sound. Therefore, noise-measuring sets are designed to give full-power indication only for bursts of noise lasting 200 ms or longer.

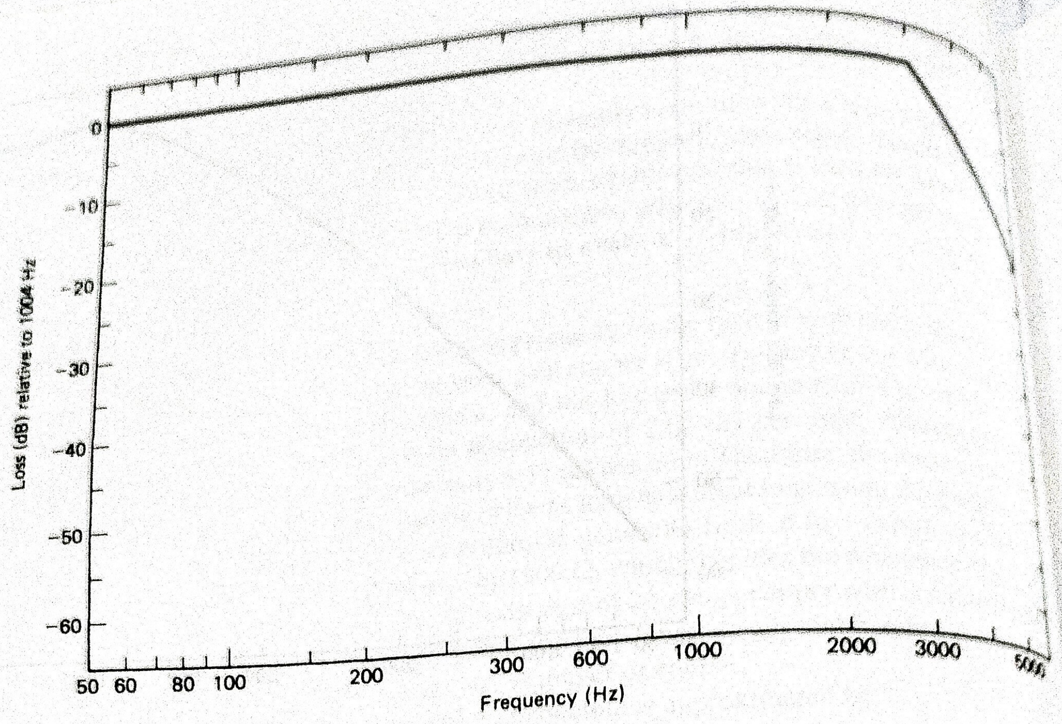

FIGURE 9-3 3-kHz flat response curve

When different types of noise cause equal interference as determined in subjective tests, use of the meter should give equal readings.

The reference established for performing message-channel noise measurements is −90 dBm (10^{-12} watts). The power level of −90 dBm was selected because, at the time, power levels could not measure levels below −90 dBm, and therefore it would not be necessary to deal with negative values when reading noise levels. Thus, a 1000-Hz tone with a power level of −90 dBm is equal to a noise reading of 0 dBrn. Conversely, a 1000-Hz tone with a power level of 0 dBm is equal to a noise reading of 90 dBrn, and a 1000-Hz tone with a power level of −40 dBm is equal to a noise reading of 50 dBrn.

When appropriate, other weighting networks can be substituted for C-message. For example, a *3-kHz flat* network is used to measure power density of white noise. This network has a nominal low-pass frequency response down 3 dB at 3 kHz and rolls off at 12 dB per octave. A 3-kHz flat network is often used for measuring high levels of low-frequency noise, such as power supply hum. The frequency response for a 3-kHz flat network is shown in Figure 9-3.

9-4 UNITS OF POWER MEASUREMENT

9-4-1 dB and dBm

To specify the amplitudes of signals and interference, it is often convenient to define them at some reference point in the system. The amplitudes at any other physical location can then be related to this reference point if the loss or gain between the two points is known. For example, sea level is generally used as the reference point when comparing elevations. By referencing two mountains to sea level, we can compare the two elevations regardless of where the mountains are located. A mountain peak in Colorado 12,000 feet above sea level is 4000 feet higher than a mountain peak in France 8000 feet above sea level.

(SDLC)

The decibel (dB) is the basic yardstick used for making power measurements in communications. The unit dB is simply a logarithmic expression representing the ratio of one power level to another and expressed mathematically as

$$\text{dB} = 10 \log\left(\frac{P_1}{P_2}\right) \tag{9-2}$$

where P_1 and P_2 are power levels at two different points in a transmission system.

From Equation 9-2, it can be seen that when $P_1 = P_2$, the power ratio is 0 dB; when $P_1 > P_2$, the power ratio in dB is positive; and when $P_1 < P_2$, the power ratio in dB is negative. In telephone and telecommunications circuits, power levels are given in dBm and differences between power levels in dB.

Equation 9-2 is essentially dimensionless since neither power is referenced to a base. The unit dBm is often used to reference the power level at a given point to 1 milliwatt. One milliwatt is the level from which all dBm measurements are referenced. The unit dBm is an indirect measure of absolute power and expressed mathematically as

$$\text{dBm} = 10 \log\left(\frac{P}{1\text{ mW}}\right) \tag{9-3}$$

where P is the power at any point in a transmission system.

From Equation 9-3, it can be seen that a power level of 1 mW equates to 0 dBm, power levels above 1 mW have positive dBm values, and power levels less than 1 mW have negative dBm values.

Example 9-1

Determine

a. The power levels in dBm for signal levels of 10 mW and 0.5 mW.
b. The difference between the two power levels in dB.

Solution a. The power levels in dBm are determined by substituting into Equation 9-3:

$$\text{dBm} = 10 \log\left(\frac{10\text{ mW}}{1\text{ mW}}\right) = 10\text{ dBm}$$

$$\text{dBm} = 10 \log\left(\frac{0.5\text{ mW}}{1\text{ mW}}\right) = -3\text{ dBm}$$

b. The difference between the two power levels in dB is determined by substituting into Equation 9-2:

$$\text{dB} = 10 \log\left(\frac{10\text{ mW}}{0.5\text{ mW}}\right) = 13\text{ dB}$$

or

$$10\text{ dBm} - (-3\text{ dBm}) = 13\text{ dB}$$

The 10-mW power level is 13 dB higher than a 0.5-mW power level.

Experiments indicate that a listener cannot give a reliable estimate of the loudness of a sound but can distinguish the difference in loudness between two sounds. The ear's sensitivity to a change in sound power follows a logarithmic rather than a linear scale, and the dB has become the unit of this change.

9-4-2 Transmission Level Point, Transmission Level, and Data Level Point

Transmission level point (TLP) is defined as the optimum level of a test tone on a channel at some point in a communications system. The numerical value of the TLP does not describe the total signal power present at that point—it merely defines what the ideal level should be.

9-4-4 Psophometric Noise Weighting

Psophometric noise weighting is used primarily in Europe. Psophometric weighting assumes a perfect receiver; therefore, its weighting curve corresponds to the frequency response of the human ear only. The difference between C-message weighting and psophometric weighting is so small that the same conversion factor may be used for both.

9-5 TRANSMISSION PARAMETERS AND PRIVATE-LINE CIRCUITS

Transmission parameters apply to dedicated *private-line data circuits* that utilize the private sector of the public telephone network—circuits with bandwidths comparable to those of standard voice-grade telephone channels that do not utilize the public switched telephone network. Private-line circuits are direct connections between two or more locations. On private-line circuits, transmission facilities and other telephone company–provided equipment are hardwired and available only to a specific subscriber. Most private-line data circuits use four-wire, full-duplex facilities. Signal paths established through switched lines are inconsistent and may differ greatly from one call to another. In addition, telephone lines provided through the public switched telephone network are two wire, which limits high-speed data transmission to half-duplex operation. Private-line data circuits have several advantages over using the switched public telephone network:

Transmission characteristics are more consistent because the same facilities are used with every transmission.

The facilities are less prone to noise produced in telephone company switches.

Line conditioning is available only on private-line facilities.

Higher transmission bit rates and better performance is appreciated with private-line data circuits.

Private-line data circuits are more economical for high-volume circuits.

Transmission parameters are divided into three broad categories: bandwidth parameters, which include attenuation distortion and envelope delay distortion; interface parameters, which include terminal impedance, in-band and out-of-band signal power, test signal power, and ground isolation; and facility parameters, which include noise measurements, frequency distortion, phase distortion, amplitude distortion, and nonlinear distortion.

9-5-1 Bandwidth Parameters

The only transmission parameters with limits specified by the FCC are attenuation distortion and envelope delay distortion. *Attenuation distortion* is the difference in circuit gain experienced at a particular frequency with respect to the circuit gain of a reference frequency. This characteristic is sometimes referred to as *frequency response, differential gain,* and *1004-Hz deviation. Envelope delay distortion* is an indirect method of evaluating the phase delay characteristics of a circuit. FCC tariffs specify the limits for attenuation distortion and envelope delay distortion. To reduce attenuation and envelope delay distortion

9-5-3-9 Phase intercept distortion. *Phase intercept distortion* occurs in coherent SSBSC systems, such as those using frequency-division multiplexing when the received carrier is not reinserted with the exact phase relationship to the received signal as the transmit carrier possessed. This impairment causes a constant phase shift to all frequencies, which is of little concern for data modems using FSK, PSK, or QAM. Because these are practically the only techniques used today with voice-band data modems, no limits have been set for phase intercept distortion.

9-5-3-10 Peak-to-average ratio. The difficulties encountered in measuring true phase distortion or envelope delay distortion led to the development of *peak-to-average ratio* (PAR) tests. A signal containing a series of distinctly shaped pulses with a high peak voltage-to-average voltage ratio is transmitted. Differential delay distortion in a circuit has a tendency to spread the pulses, thus reducing the peak voltage-to-average voltage ratio. Low peak-to-average ratios indicate the presence of differential delay distortion. PAR measurements are less sensitive to attenuation distortion than EDD tests and are easier to perform.

9-5-3-11 Facility parameter summary. Table 9-3 summarizes facility parameter limits.

9-6 VOICE-FREQUENCY CIRCUIT ARRANGEMENTS

Electronic communications circuits can be configured in several ways. Telephone instruments and the voice-frequency facilities to which they are connected may be either *two wire* or *four wire*. Two-wire circuits have an obvious economic advantage, as they use only half as much copper wire. This is why most local subscriber loops connected to the public switched telephone network are two wire. However, most private-line data circuits are configured four wire.

9-6-1 Two-Wire Voice-Frequency Circuits

As the name implies, *two-wire transmission* involves two wires (one for the signal and one for a reference or ground) or a circuit configuration that is equivalent to using only two wires. Two-wire circuits are ideally suited to simplex transmission, although they are often used for half- and full-duplex transmission.

Figure 9-19 shows the block diagrams for four possible two-wire circuit configurations. Figure 9-19a shows the simplest two-wire configuration, which is a passive circuit consisting of two copper wires connecting a telephone or voice-band modem at one

(SDLC)

Table 9-3 Facility Parameter Limits

Parameter	Limit	
1. 1004-Hz loss variation	Not more than ±4 dB long term	
2. C-message noise	Maximum rms noise at modem receiver (nominal − 16 dBm point)	
Facility miles	*dBm*	*dBrnc0*
0–50	−61	32
51–100	−59	34
101–400	−58	35
401–1000	−55	38
1001–1500	−54	39
1501–2500	−52	41
2501–4000	−50	43
4001–8000	−47	46
8001–16,000	−44	49
3. C-notched noise	(minimum values)	
(a) Standard voice-band channel	24-dB signal to C-notched noise	
(b) High-performance line	28-dB signal to C-notched noise	
4. Single-frequency interference	At least 3 dB below C-message noise limits	
5. Impulse noise		
Threshold with respect to 1004-Hz holding tone	*Maximum counts above threshold allowed in 15 minutes*	
0 dB	15	
+4 dB	9	
+8 dB	5	
6. Frequency shift	±5 Hz end to end	
7. Phase intercept distortion	No limits	
8. Phase jitter	No more than 10° peak to peak (end-to-end requirement)	
9. Nonlinear distortion (D-conditioned circuits only)		
Signal to second order	At least 35 dB	
Signal to third order	At least 40 dB	
10. Peak-to-average ratio	Reading of 50 minimum end to end with standard PAR meter	
11. Phase hits	8 or less in any 15-minute period greater than ±20 peak	
12. Gain hits	8 or less in any 15-minute period greater than ±3 dB	
13. Dropouts	2 or less in any 15-minute period greater than 12 dB	

station through a telephone company interface to a telephone or voice-band modem at the destination station. The modem, telephone, and circuit configuration are capable of two-way transmission in either the half- or the full-duplex mode.

Figure 9-19b shows an active two-wire transmission system (i.e., one that provides gain). The only difference between this circuit and the one shown in Figure 9-19a is the addition of an amplifier to compensate for transmission line losses. The amplifier is unidirectional and, thus, limits transmission to one direction only (simplex).

Figure 9-19c shows a two-wire circuit using a digital T carrier for the transmission medium. This circuit requires a T carrier transmitter at one end and a T carrier receiver at the other end. The digital T carrier transmission line is capable of two-way transmission; however, the transmitter and receiver in the T carrier are not. The transmitter encodes the analog voice or modem signals into a PCM code, and the decoder in the receiver performs the opposite operation, converting PCM codes back to analog. The digital transmission medium is a pair of copper wires.

Figures 9-19a, b, and c are examples of *physical two-wire circuits*, as the two stations are physically interconnected with a two-wire metallic transmission line. Figure 9-19d shows an *equivalent two-wire circuit*. The transmission medium is earth's atmosphere, and there are no copper wires between the two stations. Although earth's atmosphere is capable