chapter 18

Telecommunication Systems

The telephone system is the largest and most complex electronic communication system in the world. It uses just about every type of electronic communication technique available including virtually all the ones described in this book.

Although the primary purpose of the telephone system is to provide voice communication, it is also widely used for many other purposes, including facsimile transmission, TV transmission, and computer data transmission. Data transmission via modems and DSL was discussed in Chap. 11, so it will not be repeated here. The chapter concludes with coverage of the newest telephone technology called Voice over Internet Protocol (VoIP).

The wired telephone is gradually becoming history. Less than 50 percent of homes now have a standard wired phone. Most homes now have a least one wireless cell phone that is the sole telephone connection. The vast wired network will not go away, but the traditional telephone companies are now phasing out their standard voice service. That trend will continue into the future as wireless becomes the dominant phone service.

Objectives

After completing this chapter, you will be able to:

- Name and describe the components in conventional and electronic telephones.
- Describe the characteristics of the various signals used in telephone communication.
- State the general operation of a cordless telephone.
- Describe the operation of a PBX.
- Explain the hierarchy of signal transmission within the telephone system.
- Explain the operation of a facsimile machine.
- Describe the operation of an Internet Protocol telephone.

18-1 Telephones

The original telephone system was designed for full duplex analog communication of voice signals. Today, the telephone system is still primarily used for voice, but it employs mostly digital techniques, not only in signal transmission but also in control operations.

The telephone system permits any telephone to connect with any other telephone in the world. This means that each telephone must have a unique identification code—the 10-digit telephone number assigned to each telephone. The telephone system provides a means of recognizing each individual number and provides switching systems that can connect any two telephones.

The Local Loop

Standard telephones are connected to the telephone system by way of a two-wire, twisted-pair cable that terminates at the local exchange or central office. As many as 10,000 telephone lines can be connected to a single central office (see Fig. 18-1). The connections from the central office go to the "telephone system" represented in Fig. 18-1 by the large "cloud." This part of the system, which is mainly long distance, is described in Sec. 18-2. A call originating at telephone A will pass through the central office and then into the main system, where it is transmitted via one of many different routes to the central office connected to the desired location designated as B in Fig. 18-1. The connection between nearby local exchanges is direct rather than long distance.

Local loop (subscriber loop)

Last mile or first mile

The two-wire, twisted-pair connection between the telephone and the central office is referred to as the *local loop* or *subscriber loop*. You will also hear it referred to as the *last mile* or the *first mile*. The circuits in the telephone and at the central office form a complete electric circuit, or loop. This single circuit is analog and carries both dc and ac signals. The dc power for operating the telephone is generated at the central office and supplied to each telephone over the local loop. The ac voice signals are transmitted along with the dc power. Despite the fact that only two wires are involved, full duplex operation, i.e., simultaneous send and receive, is possible. All dialing and signaling operations are also carried on this single twisted-pair cable.

Telephone Set

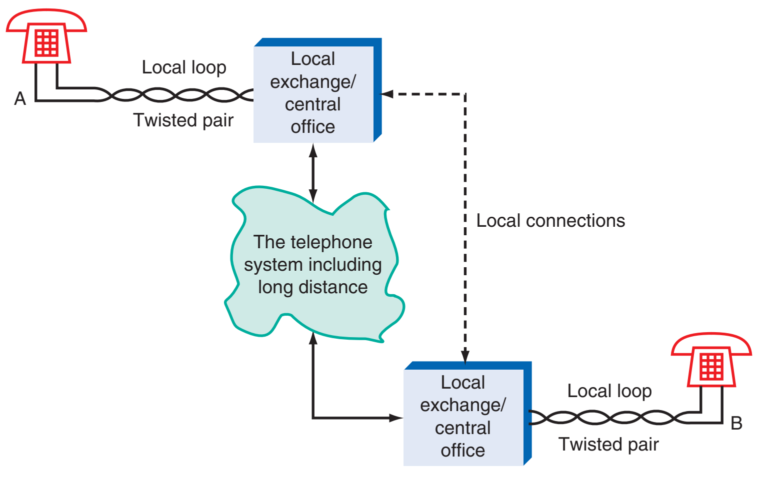
Telephone set

Transmitter

Receiver

A basic telephone or *telephone set* is an analog baseband transceiver. It has a handset that contains a microphone and a speaker, better known as a *transmitter* and a *receiver*.

Figure 18-1 The basic telephone system.



It also contains a ringer and a dialing mechanism. Overall, the telephone set fulfills the following basic functions.

The receive mode provides:

1. An incoming signal that rings a bell or produces an audio tone indicating that a call is being received
2. A signal to the telephone system indicating that the signal has been answered
3. Transducers to convert voice to electric signals and electric signals to voice

The transmit mode:

1. Indicates to the telephone system that a call is to be made when the handset is lifted
2. Indicates that the telephone system is ready to use by generating a signal called the *dial tone*
3. Provides a way of transmitting the telephone number to be called to the telephone system
4. Receives an indication that the call is being made by receiving a ringing tone
5. Provides a means of receiving a special tone indicating that the called line is busy
6. Provides a means of signaling the telephone system that the call is complete

Dial tone

All telephone sets provide these basic functions. Some of the more advanced electronic telephones have other features such as multiple line selection, hold, speaker phone, call waiting, and caller ID.

Fig. 18-2 is a basic block diagram of a telephone set. The function of each block is described below. Detailed circuits for each of the blocks and their operation are described later when the standard and electronic telephones are discussed in detail.

Ringer. The *ringer* is either a bell or an electronic oscillator connected to a speaker. It is continuously connected to the twisted pair of the local loop back to the central office. When an incoming call is received, a signal from the central office causes the bell or ringer to produce a tone.

Ringer

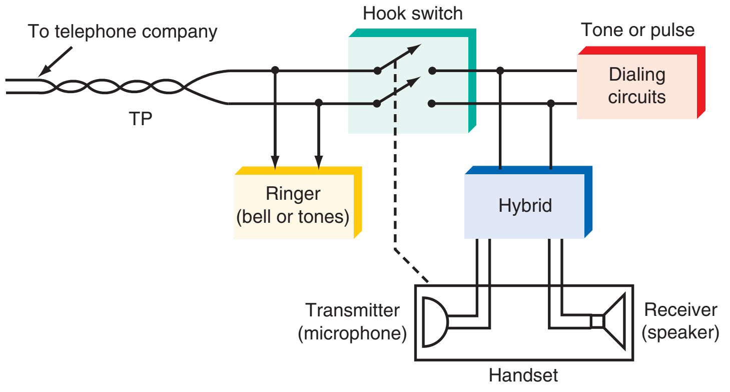
Switch Hook. A *switch hook* is a double-pole mechanical switch that is usually controlled by a mechanism actuated by the telephone handset. When the handset is "on the hook," the hook switch is open, thereby isolating all the telephone circuitry from the central office local loop. When a call is to be made or to be received, the handset is taken off the hook. This closes the switch and connects the telephone circuitry to the local loop. The direct current from the central office is then connected to the telephone, closing its circuits to operate.

Switch hook

Dialing Circuits. The *dialing circuits* provide a way for entering the telephone number to be called. In older telephones, a pulse dialing system was used. A rotary dial connected

Dialing circuit

Figure 18-2 Basic telephone set.



to a switch produced a number of on/off pulses corresponding to the digit dialed. These on/off pulses formed a simple binary code for signaling the central office.

Dual-tone multifrequency (DTMF) system

In most modern telephones, a tone dialing system is used. Known as the *dual-tone multifrequency (DTMF) system,* this dialing method uses a number of pushbuttons that generate pairs of audio tones that indicate the digits called.

Whether pulse dialing or tone dialing is used, circuits in the central office recognize the signals and make the proper connections to the dialed telephone.

Handset

Handset. This unit contains a microphone for the transmitter and a speaker or receiver. When you speak into the transmitter, it generates an electric signal representing your voice. When a received electric voice signal occurs on the line, the receiver translates it to sound waves. The transmitter and receiver are independent units, and each has two wires connecting to the telephone circuit. Both connect to a special device known as the hybrid.

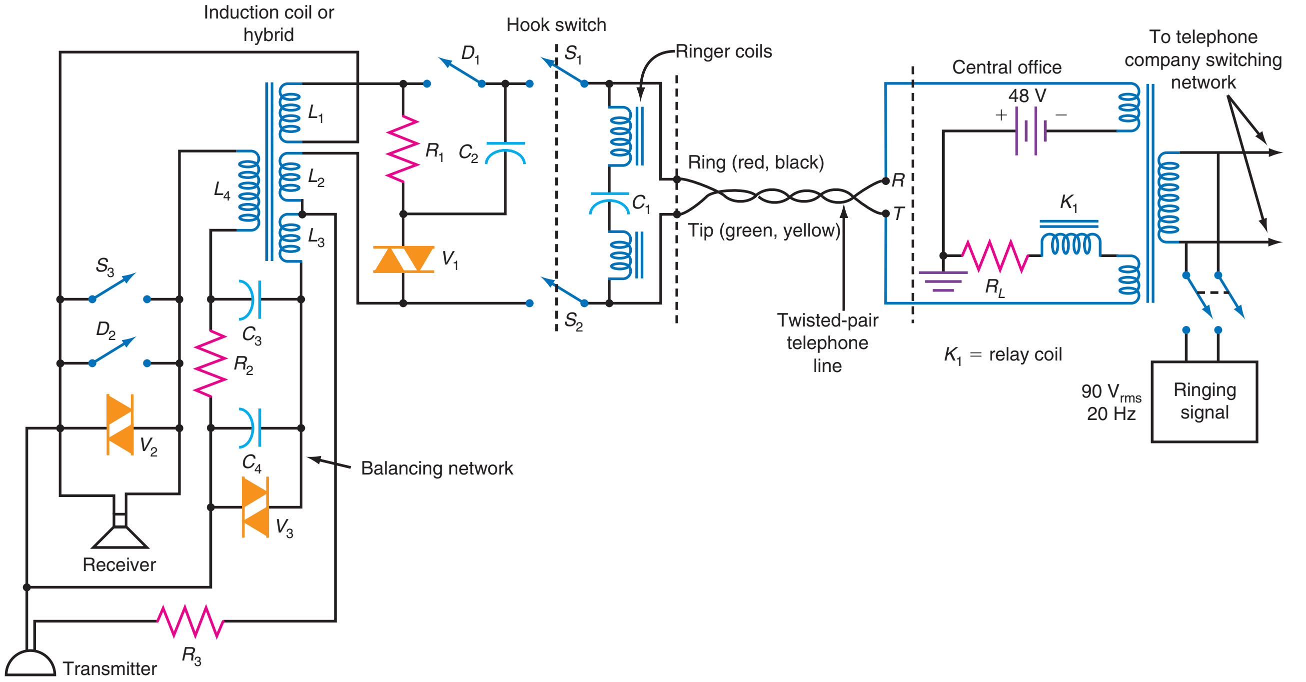
Hybrid circuit

Hybrid. The *hybrid circuit* is a special transformer used to convert signals from the four wires from the transmitter and receiver to a signal suitable for a single two-line pair to the local loop. The hybrid permits *full duplex,* i.e., simultaneous send and receive, analog communication on the two-wire line. The hybrid also provides a side tone from the transmitter to the receiver so that the speaker can hear her or his voice in the receiver. This feedback permits automatic voice-level adjustment.

Standard Telephone and Local Loop

Fig. 18-3 is a simplified schematic diagram of a conventional telephone and the local loop connections back to the central office. The circuitry at the central office is discussed in greater detail later. For now, note that the central office applies a dc voltage over the twisted-pair line to the telephone. This dc voltage is approximately −48 V with respect to ground in the open-circuit condition. When a subscriber picks up the telephone, the switch hook closes, connecting the circuitry to the telephone line. The load represented by the telephone circuitry causes current to flow in the local loop and the voltage inside the telephone to drop to approximately 5 to 6 V.

Figure 18-3 Standard telephone circuit diagram showing connection to central office.



The amount of current flowing in the local loop depends upon a number of factors. The dc voltage supplied by the central office may not be exactly −48 V. It can, in fact, vary many volts above or below the 48-V normal value.

As Fig. 18-3 shows, the central office also inserts some resistance R_L to limit the total current flow if a short circuit occurs on the line. This resistance can range from about 350 to 800 Ω. In Fig. 18-3, the total resistance is approximately 400 Ω.

The resistance of the telephone itself also varies over a relatively wide range. It can be as low as 100 Ω and as high as 400 Ω, depending upon the circuitry. The resistance varies because of the resistance of the transmitter element and because of the variable resistors called *varistors* used in the circuit to provide automatic adjustment of line level.

Varistor

The local loop resistance depends considerably on the length of the twisted pair between the telephone and the central office. Although the resistance of copper wire in the twisted pair is relatively low, the length of the wire between the telephone and the central office can be many miles long. Thus the resistance of the local loop can be anywhere from 1000 to 1800 Ω, depending upon the distance. The local loop length can vary from a few thousand feet up to about 18,000 ft.

Finally, the frequency response of the local loop is approximately 300 to 3400 Hz. This is sufficient to pass voice frequencies that produce full intelligibility. An unloaded twisted pair has an upper cutoff frequency of about 4000 Hz. But this cutoff varies considerably depending upon the overall length of the cable. When long runs of cable are used, special loading coils are inserted into the line to compensate for excessive roll-off at the higher frequencies.

Tip

Ring

The two wires used to connect telephones are labeled *tip* and *ring*. These designations refer to the plug used to connect telephones to one another at the central office. At one time, large groups of telephone operators at the central office used plugs and jacks at a switchboard to connect one telephone to another manually.

The tip wire is green and is usually connected to ground; the ring wire is red. Many telephone cables into a home or an office also contain a second twisted pair if a separate telephone line is to be installed. These wires are usually color-coded black and yellow. Black and yellow correspond to ring and tip, respectively, where yellow is ground. Other color combinations are used in telephone wiring.

> **GOOD TO KNOW**
>
> When long runs of cable are used, special loading coils are inserted into the line to compensate for large amounts of roll-off at higher frequencies.

Ringer. In Fig. 18-3, the circuitry connected directly to the tip and ring local loop wires is the ringer. The ringer in most older telephones is an electromechanical bell. A pair of electromagnetic coils is used to operate a small hammer that alternately strikes two small metallic bells. When an incoming call is received, a voltage from the central office operates the electromagnetic coils, which in turn operate the hammer to ring the bells. The bells make the familiar tone produced by most standard telephones.

In Fig. 18-3, the ringing coils are connected in series with a capacitor C_1. This allows the ac ringing voltage to be applied to the coils but blocks the 48 V of direct current, thus minimizing the current drain on the 48 V of power supplied at the central office.

The ringing voltage supplied by the central office is a sine wave of approximately 90 V_{rms} at a frequency of about 20 Hz. These are the nominal values, because the actual ringing voltage can vary from approximately 80 to 100 V_{rms} with a frequency somewhere in the 15- to 30-Hz range. This ac signal is supplied by a generator at the central office. The ringing voltage is applied in series with the −48-V dc signal from the central office power supply. The ringing signal is connected to the local loop line by way of a transformer T_1. The transformer couples the ringing signal into its secondary winding where it appears in series with the 48-V dc supply voltage.

> **GOOD TO KNOW**
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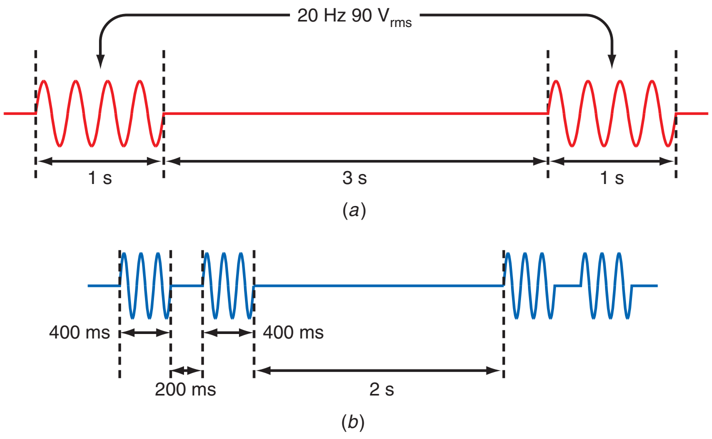
> In the United States, the telephone ringing voltage occurs for 1s followed by a 3-s break. Telephones in other countries may use different ringing sequences. Smaller office-based systems in the United States may also use other sequences for internal calls.

The standard ringing sequence is shown in Fig. 18-4. In U.S. telephones, the ringing voltage occurs for 1 s followed by a 3-s interval. Telephones in other parts of the world use different ringing sequences. For example, in the United Kingdom, the standard ring sequence is a higher-frequency tone occurring more frequently, and it consists of two ringing pulses 400 ms long, separated by 200 ms. This is followed by a 2-s interval of quiet before the tone sequence repeats.

Figure 18-4 Telephone ringing sequence. (*a*) United States and Europe. (*b*) United Kingdom.



Transmitter. The transmitter is the microphone into which you speak during a telephone call. In a standard telephone, this microphone uses a carbon element that effectively translates acoustical vibrations into resistance changes. The resistance changes, in turn, produce current variations in the local loop representing the speaker's voice. A dc voltage must be applied to the transmitter so that current flows through it during operation. The 48 V from the central office is used in this case to operate the transmitter.

The resulting ac voice signal produced on the telephone line is approximately 1 to 2 V_{rms}.

Receiver. The receiver, or earpiece, is basically a small permanent-magnet speaker. A thin metallic diaphragm is physically attached to a coil that rests inside a permanent magnet. Whenever a voice signal comes down a telephone line, it develops a current in the receiver coil. The coil produces a magnetic field that interacts with the permanent-magnet field. The result is vibration of the diaphragm in the receiver, which converts the electric signal to the acoustic energy that supplies the voice to the ear. As it comes in over the local loop lines, the voice signal has an amplitude of approximately 0.5 to 1 V_{rms}.

Hybrid. The hybrid is a transformerlike device that is used to simultaneously transmit and receive on a single pair of wires. The hybrid, which is also sometimes referred to as an *induction coil,* is really several transformers combined into a single unit. The windings on the transformers are connected in such a way that signals produced by the transmitter are put on the two-wire local loop but do not occur in the receiver. In the same way, the transformer windings permit a signal to be sent to the receiver, but the resulting voltage is not applied to the transmitter.

In practice, the hybrid windings are set up so that a small amount of the voice signal produced by the transmitter does occur in the receiver. This provides feedback to the speaker so that she or he may speak with normal loudness. The feedback from the transmitter to the receiver is referred to as the *side tone.* If the side tone were not provided, there would be no signal in the receiver and the person speaking would have the sensation that the telephone line was dead. By hearing his or her own voice in the receiver at a moderate level, the caller can speak at a normal level. Without the side tone, the speaker tends to speak more loudly, which is unnecessary.

Side tone

Automatic Voice Level Adjustment. Because of the wide variation in the different loop lengths of the two telephones connected to each other, the circuit resistances will vary considerably, thereby causing a wide variation in the transmitted and received voice signal levels. All telephones contain some type of component or circuit that provides *automatic voice level adjustment* so that the signal levels are approximately the same regardless of the loop lengths. In the standard telephone, this automatic loop length adjustment is handled by components called *varistors.* These are labeled V_1, V_2, and V_3 in Fig. 18-3.

Automatic voice level adjustment

Varistors

A varistor is a nonlinear resistance element whose resistance changes depending upon the amount of current passing through it. When the current passing through the varistor increases, its resistance decreases. A decrease in current causes the resistance to increase.

The varistors are usually connected across the line. In Fig. 18-3, varistor V_1 is connected in series with resistor R_1. This varistor automatically shunts some of the current away from the transmitter and the receiver. If the loop is long, the current will be relatively low and the voltage at the telephone will be low. This causes the resistance of the varistor to increase, thus shunting less current away from the transmitter and receiver. On short local loops, the current will be high and the voltage at the telephone will be high. This causes the varistor resistance to decrease; thus more current is shunted away from the transmitter and receiver. The result is a relatively constant level of transmitted or received speech.

Note that a second varistor V_3 is used in the balancing network. The balancing network (C_3, C_4, R_2) works in conjunction with the hybrid to provide the side tone discussed earlier. The varistor adjusts the level of the side tone automatically.

Pulse Dialing. The term *dialing* is used to describe the process of entering a telephone number to be called. In older telephones, a rotary dial was used. In more modern telephones, pushbuttons that generate electronic tones are used for "dialing."

The use of a rotary dialing mechanism produces what is known as *pulse dialing.* Rotating the dial and releasing it cause a switch contact to open and close at a fixed rate, producing current pulses in the local loop. These current pulses are detected by the central office and used to operate the switches that connect the dialing telephone to the called telephone. While most telephone companies still support pulse dialing, most dial phones have been long retired. Pulse dialing is no longer widely used.

Pulse dialing

Tone Dialing. Although some dial telephones are still in use and all central offices can accommodate them, most modern telephones use a dialing system known as *Touch-Tone.* It uses pairs of audio tones to create signals representing the numbers to be dialed. This dialing system is referred to as the *dual-tone multifrequency (DTMF) system.*

Tone dialing

Dual-tone multifrequency (DTMF) system

A typical DTMF keyboard on a telephone is shown in Fig. 18-5. Most telephones use a standard keypad with 12 buttons or switches for the numbers 0 through 9 and the special symbols * and #. The DTMF system also accommodates four additional keys for special applications.

In Fig. 18-5 numbers represent audio frequencies associated with each row and column of pushbuttons. For example, the upper horizontal row containing the keys for 1, 2, and 3 is labeled 697, which means that when any one of these three keys is depressed, a sine wave of 697 Hz is produced. Each of the four horizontal rows produces a different frequency. The horizontal rows generate what is generally known as the *low group of frequencies.*

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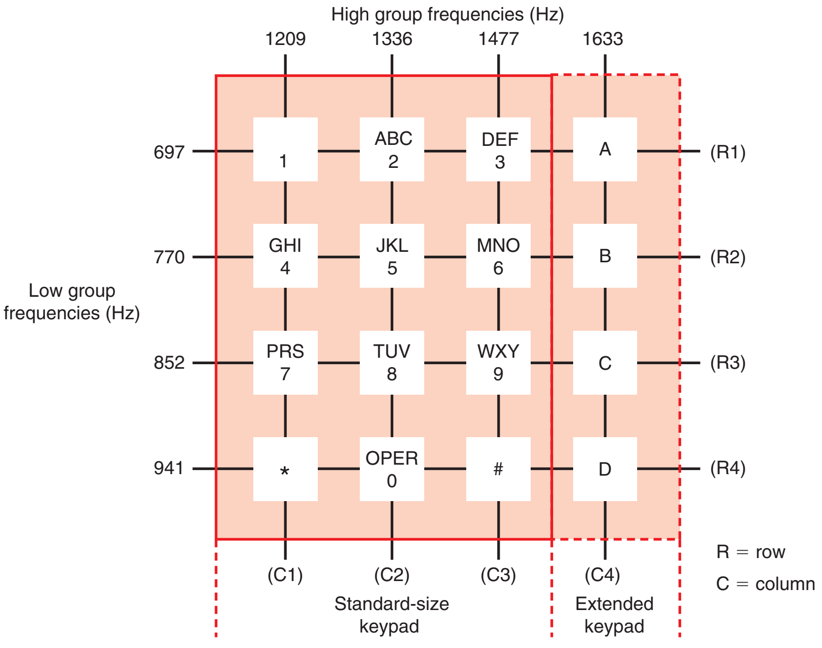
The dial tone is a continuous mix of 350 and 440 Hz at −13 dB.

A higher group of frequencies is associated with the vertical columns of keys. For example, the keys for the numbers 2, 5, 8, and 0 produce a frequency of 1336 Hz when depressed.

If the number 2 is depressed, two sine waves are generated simultaneously, one at 697 Hz and the other at 1336 Hz. These two tones are linearly mixed. This combination produces a unique sound and is easily detected and recognized at the central office as the signal representing the dialed digit 2. The tolerance on the generated frequencies is usually within ± 1.5 percent.

Figure 18-5 DTMF keypad.



Electronic Telephones

When solid-state circuits came along in the late 1950s, an electronic telephone became possible and practical. Today, all new telephones are electronic, and they use integrated-circuit technology.

The development of the microprocessor has also affected telephone design. Although simple electronic telephones do not contain a microprocessor, most multiple-line and full-feature telephones do. A built-in microprocessor permits automatic control of the telephone's functions and provides features such as telephone number storage and automatic dialing and redialing that are not possible in conventional telephones.

IC electronic telephone

Typical IC Electronic Telephone. The major components of a typical electronic telephone circuit are shown in Fig. 18-6. Most of the functions are implemented with circuits contained within a single IC.

In Fig. 18-6, note that the TouchTone keypad drives a DTMF tone generator circuit. An external crystal or ceramic resonator provides an accurate frequency reference for generating the dual dialing tones.

The tone ringer is driven by the 20-Hz ringing signal from the phone line and drives a piezoelectric sound element.

The IC also contains a built-in line voltage regulator. It takes the dc voltage from the local loop and stabilizes it to provide a constant voltage to the internal electronic circuits. An external zener diode and transistor provide bias to the electret microphone.

The internal speech network contains a number of amplifiers and related circuits that fully duplicate the function of a hybrid in a standard telephone. This IC also contains a microcomputer interface. The box labeled MPU is a single-chip *microprocessing unit.* Although it is not necessary to use a microprocessor, if automatic dialing and other functions are implemented, this circuit is capable of accommodating them.

Finally, note the bridge rectifier and hook switch circuit. The twisted pair from the local loop is connected to the tip and ring. Both the 48-V dc and 20-Hz ring voltages will be applied to this bridge rectifier. For direct current, the bridge rectifier provides polarity protection for the circuit, ensuring that the bridge output voltage is always positive. When the ac ringing voltage is applied, the bridge rectifier rectifies it into a

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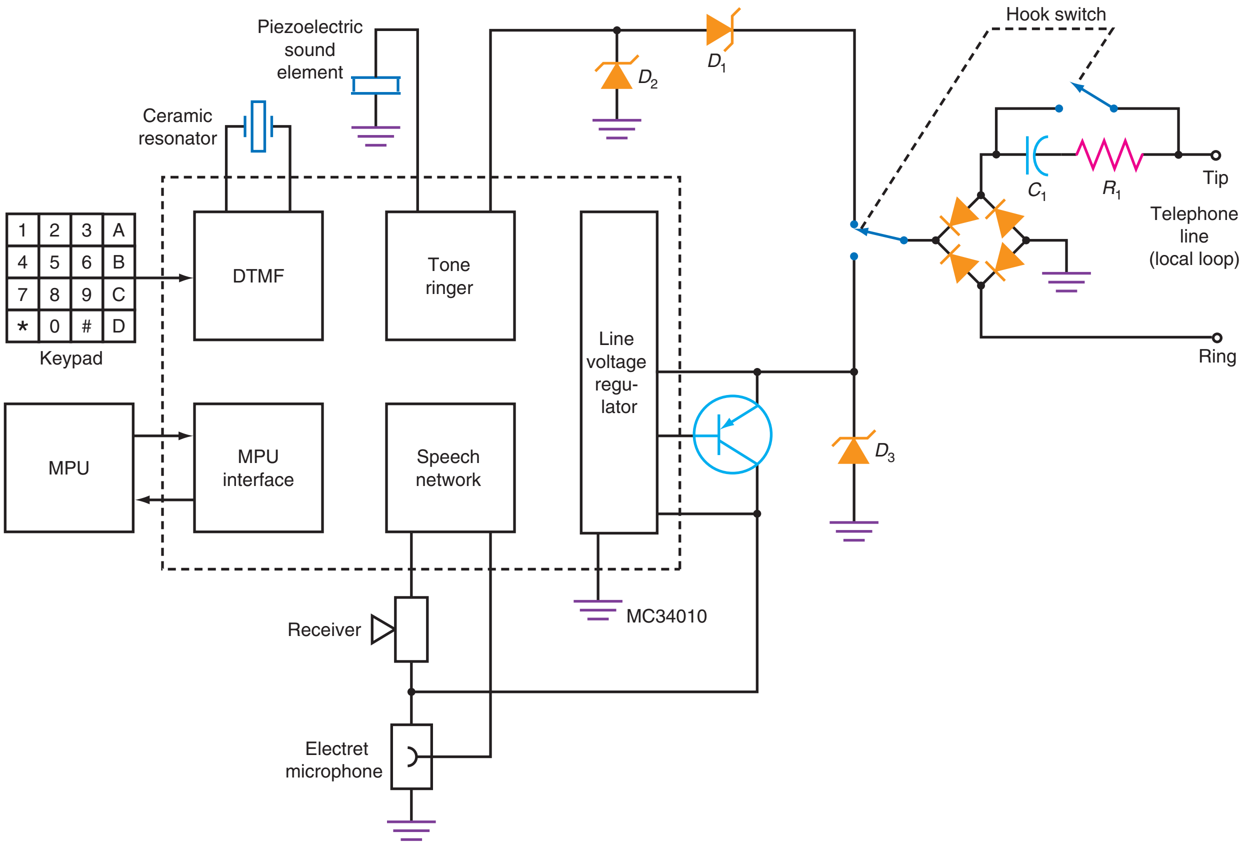
Newer telephones with built-in microprocessors are able to provide users with features such as telephone number storage and automatic dialing.

Figure 18-6 Single-chip electronic telephone.



pulsating dc voltage. The hook switch is shown with the telephone on the hook or in the "hung-up" position. Thus the dc voltage is not connected to the circuit at this time. However, the ac ringing voltage will be coupled through the resistor and capacitor to the bridge, where it will be rectified and applied to the two zener diodes D_1 and D_2 that drive the tone ringer circuit.

When the telephone is taken off the hook, the hook switch closes, providing a dc path around the resistor and capacitor R_1 and C_1. The path to the tone ringer is broken, and the output of the bridge rectifier is connected to zener diode D_3 and the line voltage regulator. Thus the circuits inside the IC are powered up, and calls may be received or made.

Microprocessor Control. All modern electronic telephones contain a built-in microcontroller. Like any microcontroller, it consists of the CPU, a ROM in which a control program is stored, a small amount of random access read-write memory, and I/O circuits. The microcontroller, usually a single-chip IC, may be directly connected to the telephone IC, or some type of intermediate interface circuit may be used.

The functions performed by the microcomputer include operating the keyboard and any LCD display, if present. Some other functions involve storing telephone numbers and automatically redialing. Many advanced telephones have the capability of storing 10 or more commonly called numbers. The user puts the telephone into a program mode and uses the TouchTone keypad to enter the most frequently dialed numbers. These are stored in the microcontroller's RAM. To automatically dial one of the numbers, the user depresses a pushbutton on the front of the telephone. This may be one of the TouchTone pushbuttons, or it may be a separate set of pushbuttons provided for the purpose. When one of the pushbuttons is depressed, the microcontroller supplies a preprogrammed set of binary codes to the DTMF circuitry in the telephone IC. Thus the number is automatically dialed. Other features implemented by the microcontroller are caller ID and an answering machine.

Answering machine

Voice mail

Voice Mail. Previously called an *answering machine,* this feature is implemented on most electronic phones. The microcontroller automatically answers the call after a preprogrammed number of rings and saves the voice message. In older answering machines, the message was recorded on a tape cassette. But in modern phones, the voice message is digitized, compressed, and then stored in a small flash ROM ready for replay. The outgoing message is also stored there.

Caller ID (calling line identification service)

Caller ID. *Caller ID,* also known as the *calling line identification service,* is a feature that is now widely implemented on most electronic telephones. To make use of this service, you must sign up and pay for it monthly. With this feature, any calling number will be displayed on an LCD readout when the phone is ringing. This allows you to identify the caller.

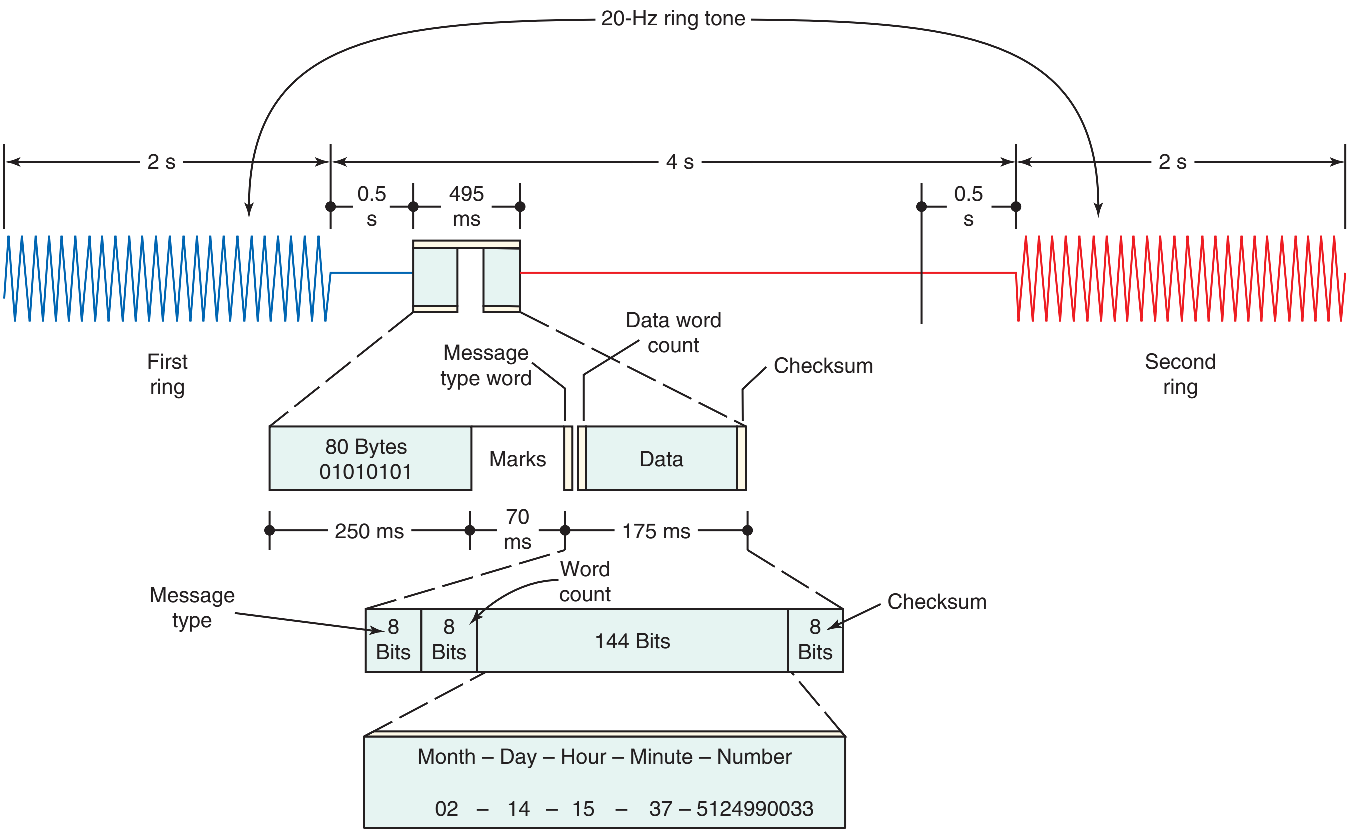
The caller ID service sends a digitized version of the calling number to your phone during the first and second rings. The data transmitted includes the date, time, and calling number. Data is transmitted by FSK, where a binary 1 (mark) is a 1200-Hz tone and a binary 0 (space) is a 2200-Hz tone. The data rate is 1200 bps.

Single-data message format (SDMF)

Multiple-data message format (MDMF)

There are two message formats in use, the *single-data message format (SDMF)* and the *multiple-data message format (MDMF).* The SDMF is illustrated in Fig. 18-7. One-half second after the first ring, 80 bytes of alternating 0s and 1s (hex 05) is transmitted for 250 ms followed by 70 ms of mark symbols. These two signals provide initialization and synchronization of the caller ID circuitry in the phone. This is followed by 1 byte describing the message type. This is usually a binary 4 (00000100), indicating the SDMF. This is followed by a byte containing the message length, usually the number of digits in the calling number. Next the data is transmitted. This is the date, time, and the 10-digit phone number transmitted as ASCII bytes with the least significant digit first. The data format is 2 digits for the month, 2 digits for the day, 2 digits for the hour (military time), 2 digits for the minutes, and up to 10 digits for the calling number. For example, if the

Figure 18-7 The caller ID transmission format.



date is February 14, the time is 3:37 p.m., and the calling number is 512-499-0033, the data sequence would be 021415375124990033. The final byte in the message is the checksum that is used for error detection. The checksum is the 2s complement sum (XOR) of all the data bytes not including the initialization and sync signals.

If the calling number is outside the calling area, the system will display an O on the LCD rather than the calling number. Furthermore, a caller may also have his or her number blocked. This can be done by setting it up with the service provider in advance or by dialing *67 prior to making the call. This will cause a P to be displayed on the LCD instead of the calling number.

A more advanced data format is the MDMF. It is similar to the SDMF but includes an extra field for the name of the calling party plus additional identification bytes.

Line Interface. Most telephones are connected by way of a thin multiwire cable to a wall jack. A special connector on the cable, called an *RJ-11 modular connector,* plugs into the matching wall jack. Two local loops are available if needed.

RJ-11 modular connector

The wall jack is connected by way of wiring inside the walls to a central wiring point called the *subscriber interface.* Also known as the *wiring block* or *modular interface,* this is a small plastic housing containing all the wiring that connects the line from the telephone company to all the telephone wires in the house. Many houses and apartments are wired so that there is a wall jack in every room.

Subscriber interface

Wiring block or modular interface

Fig. 18-8 is a general diagram of the modular interface. The line from the telephone company usually passes through a protector that provides lightning protection. It then terminates at the interface box. An RJ-11 jack and plug are provided to connect to the rest of the wiring. This gives the telephone company a way to disconnect the incoming line from the rest of the house wiring and makes testing and troubleshooting easier.

All the wiring is made by way of screw terminals. For a single-line house, the green and red tip and ring connections terminate at the terminals, and all wiring to the room wall jacks is connected in parallel at these terminals.

If a second line is installed, the black and yellow wires, which are the tip and ring connections, are also terminated at screw terminals. They are then connected to the inside house wiring.

Figure 18-8 Subscriber interface.

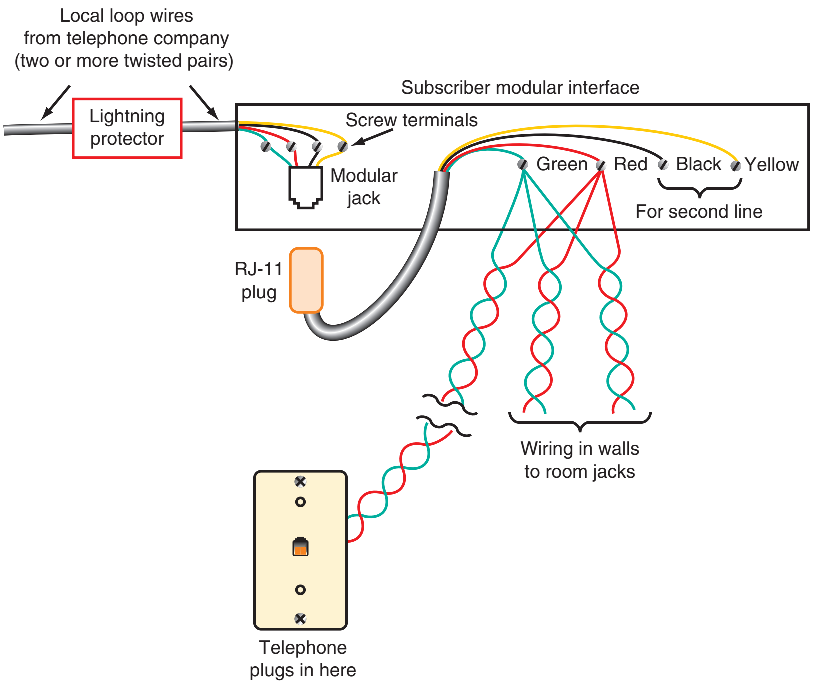
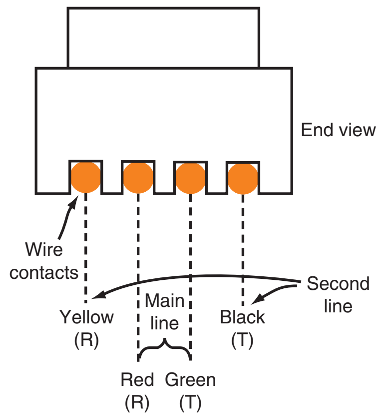


Figure 18-9 Connections to modular plug.



Connections on the RJ-11 connector are shown in Fig. 18-9. The red and green wires terminate at the two center connections, and the black and yellow wires terminate at the two outside connections. Most telephone wire and RJ-11 connectors have four wires and connections. Some cables have only the two inner wires. With four wires a two-line phone can be accommodated.

Cordless Telephones

Virtually all offices and most homes now have two or more telephones, and many homes and apartments have a standard telephone jack in every room. This permits a single phone to be moved easily from one place to another, and it permits multiple (extension) phones. However, the ultimate convenience is a *cordless telephone,* which uses two-way radio transmission and provides total portability. Today, most homes have a cordless unit.

Cordless telephone

Cordless Telephone Concepts. A cordless telephone is a full duplex, two-way radio system made up of two units, the portable unit or handset and the base unit. The base unit is wired to the telephone line by way of a modular connector. It receives its power from the ac line. The base unit is a complete transceiver in that it contains a transmitter that sends the received audio signal to the portable unit and receives signals transmitted by the portable unit and retransmits them on the telephone line. It also contains a battery charger that rejuvenates the battery in the handheld unit.

The portable unit is also a battery-powered transceiver. This unit is designed to rest in the base unit where its battery can be recharged. Both units have an antenna.

The transceivers in both the portable and the base units use full duplex operation. To achieve this, the transmitter and receiver must operate on different frequencies.

Fig. 18-10 shows simplified block diagrams of the base and portable units of a typical cordless telephone. Both the base unit and the handset contain an embedded microcontroller that controls all operations, including the keyboard and display. A high percentage of cordless units also contain a caller ID function, and many contain a voice mail feature. An analog-to-digital converter translates a received voice message to digital; it is compressed by the microcontroller and then stored in a flash memory connected to the microcontroller.

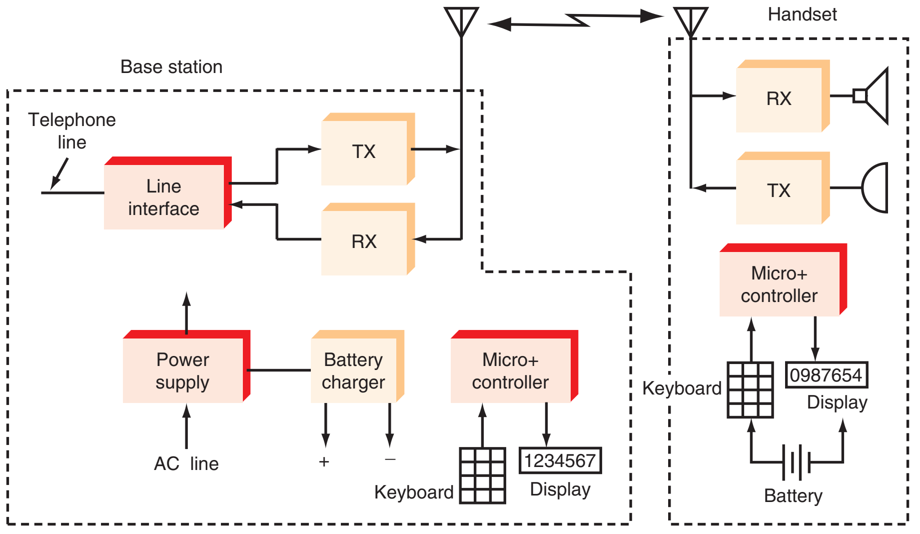
GOOD TO KNOW

The base unit and receiver unit in cordless telephones each contain a separate full duplex, complete transceiver.

Frequency allocation

Frequency Allocations. The FCC has set aside four primary frequency bands for cordless telephones: 43 to 50 MHz, 902 to 928 MHz, 2.4 to 2.45 GHz, and 5.8 GHz. The older analog phones used 25 assigned duplex frequency pairs in the 43- to 50-MHz range. In the 902- to 928-MHz ISM band, there are more channels, but the number depends upon the technology used. The 2.4-GHz band has up to 100 wide channels where many spread spectrum signals can exist concurrently and channels are determined by a

Figure 18-10 General block diagram of a cordless telephone.



pseudorandom code. The 5.8-GHz band is the most recent addition with plenty of spectrum space for multiple channels. The phones are programmed to automatically seek a channel pair with no activity and minimum noise.

Cordless Phone Features, Capabilities, and Limitations. The frequency range defines the three basic classes of cordless telephones available today, but there are other considerations. Here is a summary of the three basic types.

The simplest and least expensive cordless phones use the 43- to 50-MHz range. They are analog phones using frequency modulation. The transmitter output power is limited to 500 mW, and this, in turn, limits the transmission range to a maximum of about 1000 ft, depending upon the environment. The FCC created these limitations deliberately to reduce the amount of interference with nearby cordless telephones as well as the many wireless baby monitors and toy walkie-talkies using the same frequencies. While some 43- to 50-MHz phones are still available, for the most part they have been replaced by the newer digital phones.

Although these older phones work well enough, they are susceptible to noise and their range is limited. If higher quality and longer range are desired, phones in the 900-MHz, 2.4-GHz, or 5.8-GHz range can be used.

Three types of 900-MHz phones are available. These are analog, digital, and spread spectrum. The analog phones use FM. Although they can transmit over a longer distance, they are still susceptible to noise. A digital 900-MHz phone is also available. It uses *Gaussian FSK (GFSK)* modulation. The best 900-MHz phones use *direct-sequence spread spectrum (DSSS)*. With a power of up to 1 W, the transmission distance is a maximum of about 5000 to 7000 ft, depending on the environment and terrain. Both types of digital phones are highly immune to noise.

Gaussian FSK (GFSK)

Direct-sequence spread spectrum (DSSS)

The newer and perhaps the best cordless phones use DSSS in the 2.4-GHz or 5.8-GHz bands. Their maximum range is nearly 7000 ft, and they are virtually immune to local noise. Although these phones are far more expensive, they offer the highest-quality sound and greatest reliability.

For the most part, cordless phones in the United State have used proprietary designs rather than those conforming to a particular standard. Since the phones are only intended to work in a home or small office setting and there is no requirement that the phone interoperate with other cordless phones, any technology will work as long as it meets the FCC's frequency and operating mode guidelines. The situation is different in Europe where standards for cordless phones have existed for many years. The newest standard

Digital Enhanced Cordless Telecommunications (DECT)

created by the European Telecommunications Standards Institute (ETSI), called *Digital Enhanced Cordless Telecommunications (DECT)* has now been approved for use in the United States. DECT works in the 1.8- to 1.9-GHz band for U.S. use.

The DECT phones are digital, using Gaussian FSK modulation. Instead of using frequency-division duplexing (FDD) with two channels, DECT uses only a single channel and time-division duplexing (TTD). In a single channel, time-division multiplexing permits 12 users per channels. Typically 10 channels are available. The raw data rate is 1.152 Mbps. The latest version of the DECT phone is 6.0, and these phones are available in the United States.

18-2 Telephone System

Most of us take telephone service for granted, as we do other so-called utilities, e.g., electric power. In the United States, telephone service is excellent. But this is certainly not the case in many other countries in the world.

Telephone system

When we refer to the *telephone system,* we are talking about the organizations and facilities involved in connecting your telephone to the called telephone regardless of where it might be in United States or anywhere else in the world. The telephone system is called the Public Switched Telephone Network (PSTN). You will sometimes hear the telephone system referred to as the Plain Old Telephone Service (POTS). A number of different companies are involved in long-distance calls, although a single company is usually responsible for local calls in a given area. These companies make up the telephone system, and they design, build, maintain, and operate all the facilities and equipment used in providing universal telephone service. A vast array of equipment and technology are employed. Practically every conceivable type of electronic technology is used to implement worldwide telephone service, and that continues to change as Internet calling known as Voice over Internet Protocol (VoIP) grows.

Telephone

The *telephone,* a small but relatively complex entity, is nothing compared to the massive system that backs it up. The telephone system can connect any two telephones in the world, and most people can only speculate on the method by which this connection takes place. It takes place on many levels and involves an incredible array of systems and technology. Obviously, it is difficult to describe such a massive system here. However, in this brief section, we attempt to describe the technical complexities of interconnecting telephones, the central office and the subscriber line interface that connect each user to the telephone system, the hierarchy of interconnections within the telephone system, and the major elements and general operation of the telephone system. Long-distance operation and special telephone interconnection systems such as the PBX are also discussed. VoIP is introduced.

Subscriber Interface

Most telephones are connected to a local central office by way of the two-line, twisted-pair local loop cable. The central office contains all the equipment that operates the telephone and connects it to the telephone system that makes the connection to any other telephone.

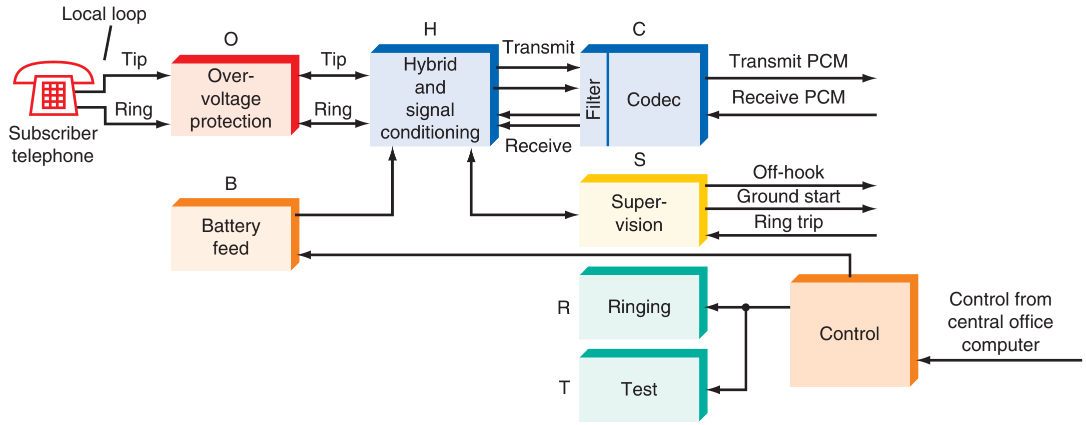
Subscriber interface or subscriber line interface circuit (SLIC)

Each telephone connected to the central office is provided with a group of basic circuits that power the telephone and provide all the basic functions, such as ringing, dial tone, and dialing supervision. These circuits are collectively referred to as the *subscriber interface* or the *subscriber line interface circuit (SLIC).* In older central office systems, the subscriber interface circuits used discrete components. Today, most functions of the subscriber line interface are implemented by one or perhaps two integrated circuits plus supporting equipment. The subscriber line interface is also referred to as the *line side interface.*

BORSCHT

The SLIC provides seven basic functions generally referred to as *BORSCHT* (representing the first letters of the functions *battery, overvoltage protection, ringing,*

Figure 18-11 BORSCHT functions in the subscriber line interface at the central office.



supervision, coding, hybrid, and *test*). A general block diagram of the subscriber interface and BORSCHT functions is given in Fig. 18-11.

Battery

Battery. The subscriber line interface at the central office must provide a dc voltage to the subscriber to operate the telephone. In the United States, this is typically −48 V dc with respect to ground. The actual voltage can be anything between approximately −20 and −80 V when the phone is on the hook, i.e., disconnected. The voltage at the telephone drops to approximately 6 V when the phone is taken off the hook. The large difference between the on-hook and off-hook voltages has to do with the large voltage drop that occurs across the components in the telephone and the long local loop cable.

Overvoltage protection

Overvoltage Protection. The circuits and components that protect the subscriber line interface circuits from electrical damage are referred to collectively as *overvoltage protection.* The phone lines are vulnerable to many types of electrical problems. Lightning is by far the worst threat, although other hazards exist, including accidental connection to an electric power line or some type of misconnection that would occur during installation. Induced disturbances from other sources of noise can also cause problems. Overvoltage protection ensures reliable telephone operation even under such conditions.

Ringing

Ringing. When a specific telephone is receiving a call, the telephone local office must provide a ringing signal. As indicated earlier, this is commonly a 90-V_{rms} ac signal at approximately 20 Hz. The SLIC must connect the ringing signal to the local loop when a call is received. This is usually done by closing relay contacts that connect the ringing signal to the line. The SLIC must also detect when the phone is picked up (off hook) so that the ringing signal can be disconnected.

Supervision

Supervision. *Supervision* refers to a group of functions within the subscriber line interface that monitor local loop conditions and provide various services. For example, the supervision circuits in the SLIC detect when a telephone is picked up to initiate a new call. A sensing circuit recognizes the off-hook condition and signals circuits within the SLIC to connect a dial tone. The caller then dials the desired number, which causes interconnection through the telephone system.

The supervision circuits continuously monitor the line during the telephone call. The circuits sense when the call is terminated and provide the connection of a busy signal if the called number is not available.

GOOD TO KNOW

On-hook and off-hook signals differ in resistance across tip and ring conductors. On-hook minimum dc resistance is 30,000 Ω, and off-hook maximum resistance is 200 Ω.

Coding. *Coding* is another name for A/D conversion and D/A conversion. Today, many telephone transmissions are made by way of serial digital data methods. The SLIC may

Coding

contain codec that converts the analog voice signals to serial PCM format or converts received digital calls back to analog signals to be placed on the local loop. Transmission over trunk lines to other central offices or toll offices or for use in long-distance transmission is typically by digital PCM signals in modern systems.

Hybrid. Recall that in the telephone, a hybrid circuit (also known as a *two-wire to four-wire circuit*), usually a transformer, provides simultaneous two-way conversations on a single pair of wires. The hybrid combines the signal from the telephone transmitter with the received signal to the receiver on the single twisted-pair cable. It keeps the signals separate within the telephone.

A hybrid is also used at the central office. It effectively translates the two-wire line to the subscriber back into four lines, two each for the transmitted and received signals. The hybrid provides separate transmit and receive signals. Although a single pair of lines is used in the local loop to the subscriber, all other connections to the telephone system treat the transmitted and received signals separately and have independent circuits for dealing with them along the way.

Test signal

Test Signals. To check the status and quality of subscriber lines, the phone company often puts special test tones on the local loop and receives resulting tones in return. These can give information about the overall performance of the local loop. The SLIC provides a way to connect the test signals to the local loop and to receive the resulting signals for measurement.

BORSCHT Functions. The basic BORSCHT functions are usually divided into two groups, high voltage and low voltage. The high-voltage parts of the system are the battery feed, overvoltage protection, ringing circuits, and test circuits. The low-voltage group includes the supervision, coding, and hybrid functions. In older systems, all the functions were implemented with discrete component circuits. Today, these functions are generally divided between two ICs, one for the high-voltage functions and the other for the low-voltage functions. However, single-chip SLIC BORSCHT ICs are now available.

Telephone hierarchy

Telephone Hierarchy

Whenever you make a telephone call, your voice is connected through your local exchange to the telephone system. From there it passes through at least one other local exchange, which is connected to the telephone you are calling. Several other facilities may provide switching, multiplexing, and other services required to transmit your voice. The telephone system is referred to as the *public switchod telephone network (PSTN)*. The organization of this hierarchy in the United States is discussed in the next sections.

Central office (local exchange)

Central Office. The *central office* or *local exchange* is the facility to which your telephone is directly connected by a twisted-pair cable. Also known as an *end office (EO),* the local exchange can serve up to 10,000 subscribers, each of whom is identified by a four-digit number from 0000 through 9999 (the last four digits of the telephone number).

The local exchange also has an exchange number. These are the three additional digits that make up a telephone number. Obviously, there can be as many as 1000 exchanges with numbers from 000 through 999. These exchanges become part of an area code region, which is defined by an additional three-digit number. Each area code is fully contained within one of the geographic areas assigned to one of the regional operating companies.

Local exchange companies (or carriers; LECs)

These companies are called *local exchange carriers,* or *local exchange companies (LECs).*

Local access and transport areas (LATAs)

Operational Relationships. The LECs provide telephone services to designated geographic areas referred to as *local access and transport areas (LATAs).* The United States is divided into approximately 200 LATAs. The LATAs are defined within the individual states making up the seven operating regions. The LECs provide the telephone service for the LATAs within their regions but do not provide long-distance service for the LATAs.

Long-distance service is provided by long-distance carriers known as *interexchange carriers (IXCs).* The IXCs are the familiar long-distance carriers, such as AT&T, Verizon and Sprint. Long-distance carriers must be used for the interconnection for any inter-LATA connections. The LECs can provide telephone service within the LATAs that are part of their operating region, but links between LATAs within a region, even though they may be directly adjacent to one another, must be made through an IXC.

Interexchange carriers (IXCs)

Each LATA contains a *serving,* or *point-of-presence (POP), office* that is used to provide the interconnections to the IXCs. The local exchanges communicate with one another via individual trunks. And all local exchanges connect to an LEC central office, which provides trunks to the POP. At the POP, the long-distance carriers can make their interface connections. The POPs must provide equal access for any long-distance carrier desiring to connect. Many POPs are connected to multiple IXCs, but in many areas, only one IXC serves a POP.

Point of presence (POP) office

Fig. 18-12 summarizes the hierarchy just discussed. Individual telephones within a LATA connect to the local exchange or central office by way of the two-wire local loop. The central offices within an LATA are connected to one another by trunks. These trunks may be standard baseband twisted-pair cables run underground or on telephone poles, but they may also be coaxial cable, fiber-optic cable, or microwave radio links. In some areas, two or more central offices are located in the same building or physical facility. Trunk interconnections are usually made by cables.

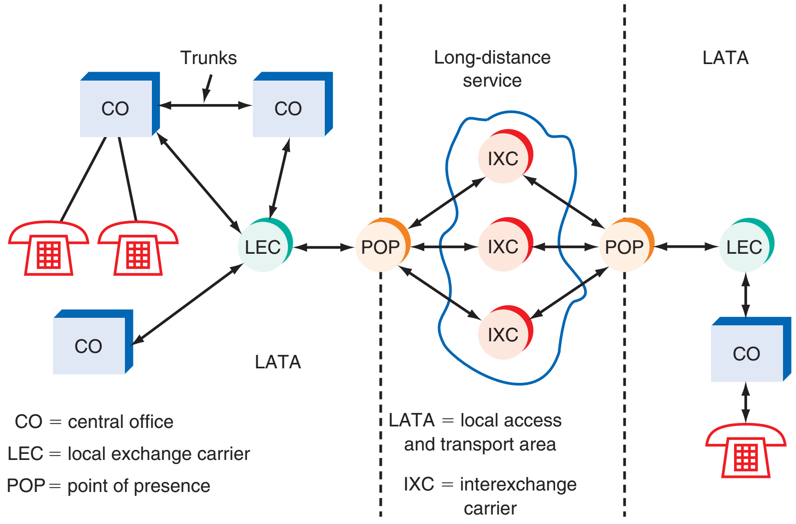
The local exchanges are also connected to an LEC central office when a connection cannot be made between two local exchanges that are not directly trunked. The call passes from the local exchange to the LEC central office, where the connection is made to the other local exchange.

The LEC central office is also connected to the POP. Depending upon the organization of the LEC within the LATA, the LEC central office may contain the POP.

Note in Fig. 18-12 that the POP provides the connections to the long-distance carriers, or IXCs. The "cloud" represents the long-distance networks of the IXCs. The long-distance network connects to the remote POPs, which in turn are connected to other central offices and local exchanges.

Most other long-distance carriers have their own specific hierarchical arrangements. A variety of switching offices across the country are linked by trunks using fiber-optic cable or microwave relay links. Multiplexing techniques are used throughout to provide many simultaneous paths for telephone calls.

Figure 18-12 Organization of the telephone system in the United States.



In all cases, the various central offices and routing centers provide switching services. The whole idea is to permit any one telephone to directly connect with any other specific telephone. The purpose of all the different levels in the telephone system hierarchy is to provide the interconnecting trunk lines as well as switching equipment that makes the desired interconnection.

The connections between central offices, central offices and LEC and POPs are digital and use the T1 and T3 multiplexing schemes described in Chap. 12. The transmission method in long distance is fiber-optic cable using protocols known as the asynchronous transfer mode (ATM), the synchronous optical network (SONET), and the optical transport network (OTN). These systems are described in Chap. 12.

System Signaling

Signaling refers to the process of setting up and disconnecting calls on the network. Signaling uses digital packets to perform all of the various operations necessary to establish a connection and tear it down. Typical functions include billing, call management (such as call-forwarding, number display, three-way calling, and 800 and 900 calls), and routing of calls from one point to another. The signaling system used in the United States and other parts of the world is called Signaling System No. 7 or SS7.

Besides the T1, T3, and other connections between central offices and other facilities, telephone companies have built a separate signaling network made up of digital links that run at 56 kbps or 64 kbps. Faster connections also exist at rates of 1.536 kbps and 1.984 kbps. These links carry the digital packets with control words that tell the system what to do when a call is made.

In addition a formal protocol has been established using the familiar OSI model. There are definitions for the four kinds of layers: physical (1 layer), data link (2), network (3), and application (7). The SS7 protocol is standardized by the ITU-T and is designated Q.700. Multiple versions of SS7 exist around the world.

Private Telephone System

Telephone service provided to companies or large organizations with many employees and many telephones is considerably different from basic local loop service provided for individuals. Depending upon the size of the organization, there may be dozens, hundreds, or even thousands of telephones required. It is simply not economical to provide each telephone in the organization with its own separate local loop connection to the central office. It is also an inefficient use of expensive facilities to use a remote central office for intercompany communication. For example, an individual in one office often may need to make an intercompany call to a person in another office, which may be only a few doors down the hall or a couple of floors away. Making this connection through the local exchange is wasteful.

Private telephone system

This problem is solved by the use of *private telephone systems* within a company or organization. Private telephone systems implement telephone service among the telephones in the organization and provide one or more local loop connections to the central office. The two basic types of private telephone systems are known as *key systems* and *private branch exchanges.*

Key system

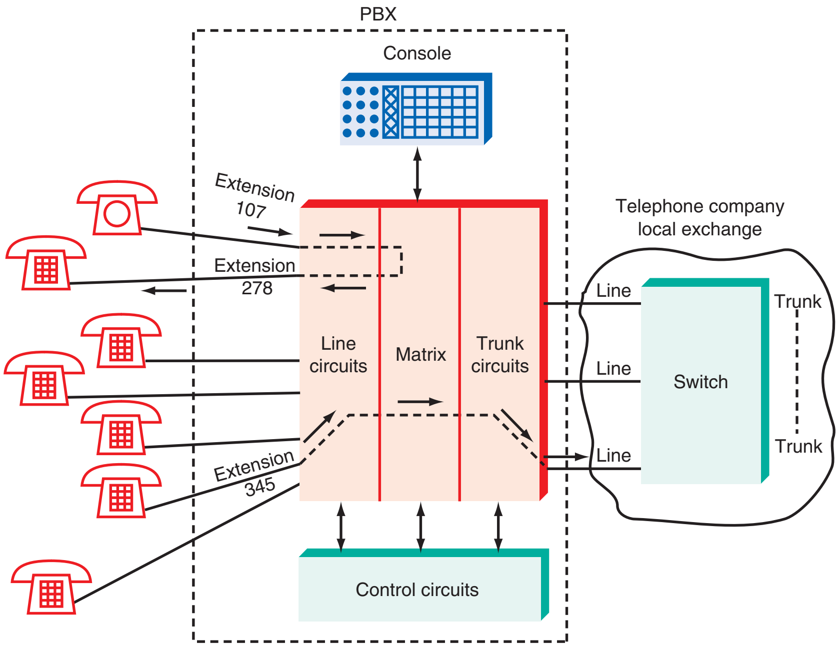
Key Systems.

Key systems are small telephone systems designed to serve from 2 to 50 user telephones within an organization. Commercially available systems usually have provisions for 6, 10, 12, or 50 telephones.

Simple key telephone systems are made up of the individual telephone units generally referred to as *stations,* all of which are connected to a central answering station. The central answering station is connected to one or more local loop lines known as *trunks* back to the local exchange. Most systems also contain a central electronic switching unit that makes all the internal and external connections.

The telephone sets in a key system typically have a group of pushbuttons that allow each telephone to select two or more outgoing trunking lines. Phone calls are made in the usual way.

Figure 18-13 A PBX.



Private Branch Exchange. A *private branch exchange,* or *PBX,* as it is known, is a private telephone system for larger organizations. Most PBXs are set up to handle 50 or more telephone interconnections. They can handle thousands of individual telephones within an organization. These systems may also be referred to as *private automatic branch exchanges (PABXs)* or *computer branch exchanges (CPXs).* Of the three terms, the expression *PBX* is the most widely known and used.

Private branch exchange (PBX)

Private automatic branch exchange (PABX)

Computer branch exchange (CPX)

A PBX (see Fig. 18-13) is, in effect, a miniature complete telephone system. It provides baseband interconnections to all the telephones in an organization. All the telephones connect to a central switching system that makes intercompany connections as well as external connections to multiple trunk lines to the central office.

Like the key system, the PBX offers the advantages of efficiency and cost reduction when many telephones are required. Interoffice calls can be completed by the PBX system without accessing the local exchange. Furthermore, it is more economical to limit the number of trunk lines to the central office, for not all telephones in the organization will be attempting to access an outside line at one time.

The modern PBX is usually fully automated by computer control. Although no operator is required, most large organizations have one or more operators who answer incoming telephone calls and route them appropriately with a control console. However, some PBXs are automated so that the individual user's telephone whose extension is the last four digits of the telephone number can be called directly from outside.

As you can see from Fig. 18-13, the PBX is made up of line circuits that are similar to the subscriber line interface circuits discussed earlier. The matrix is the electronic switch that connects any phone to any other phone in the system. It also permits conference calls. The trunk circuits interface to the local loop lines to the central office. All the circuits are under the control of a central computer dedicated to the operation of the PBX.

An alternative to the PBX is known as *Centrex.* This service, normally provided by the local telephone company, performs the function of a PBX but uses special equipment, and most of the switching is carried out by the local exchange switching equipment over special trunk lines. Its advantage over a standard PBX is that the high initial cost of PBX equipment can be avoided by leasing the Centrex equipment from the telephone company.

Centrex

Today, as more companies adopt VoIP systems, the older style PBX systems are gradually disappearing in favor of an equivalent system that uses VoIP standards. These systems attach to the company's LAN system that typically uses Ethernet to connect phones to a base or key unit for distribution and calling features such as voice mail and PBX-like answering capability. Most of these functions are implemented in software with a server dedicated to this function.

18-3 Facsimile

Facsimile (fax)

Facsimile, or *fax,* is an electronic system for transmitting graphic information by wire or radio. Facsimile is used to send printed material by scanning it and converting it to electronic signals that modulate a carrier to be transmitted over the telephone lines. Since modulation is involved, fax transmission can also take place by radio. With facsimile, documents such as letters, photographs, line drawings, or any printed information can be converted to an electric signal and transmitted with conventional communication techniques. The components of a fax system are illustrated in Fig. 18-14.

Although facsimile is used to transmit pictures, it is not TV because it does not transmit sound messages or live scenes and motion. However, it does use scanning techniques that are generally similar to those used in TV. A scanning process is used to break a printed document up into many horizontal scan lines that can be transmitted and reproduced serially.

How Facsimile Works

The early facsimile machines scanned the document to be transmitted with a light and photocell arrangement. A scanning head consists of a light source and a photocell. A light source, focused to a tiny point with a lens system, was used to scan the document. The lens was also used to focus the reflected light from on the document onto the photocell. As the light scanned the letters and numbers in a typed or printed document or the gray scale in a photograph, the photocell produced a varying electronic signal whose output amplitude was proportional to the amount of reflected light. This baseband signal was then used to amplitude- or frequency-modulate a carrier in the audio frequency range. This permitted the signal to be transmitted over the telephone lines.

Fig. 18-15 shows how a printed letter might have been scanned. Assume that the letter F is black on a white background. The output of a photodetector as it scans across line *a* is shown in Fig. 18-15(*a*). The output voltage is high for white and low for black. The output of the photodetector is also shown for scan lines *b* and *c*. The output of the photodetector is used to modulate a carrier, and the resulting signal is put on the telephone line.

The resolution of the transmission is determined by the number of scan lines per vertical inch. The greater the number of lines scanned, the finer the detail transmitted and the higher the quality of reproduction. Older systems had a resolution of 96 lines per inch (LPI), and the new systems have 200 LPI.

Figure 18-14 Components of a facsimile system.

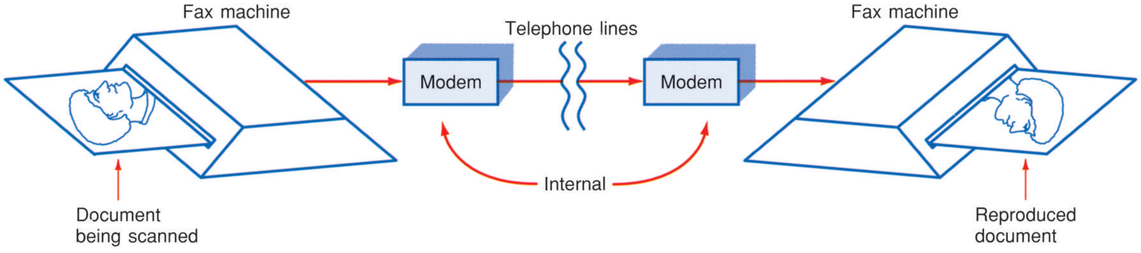
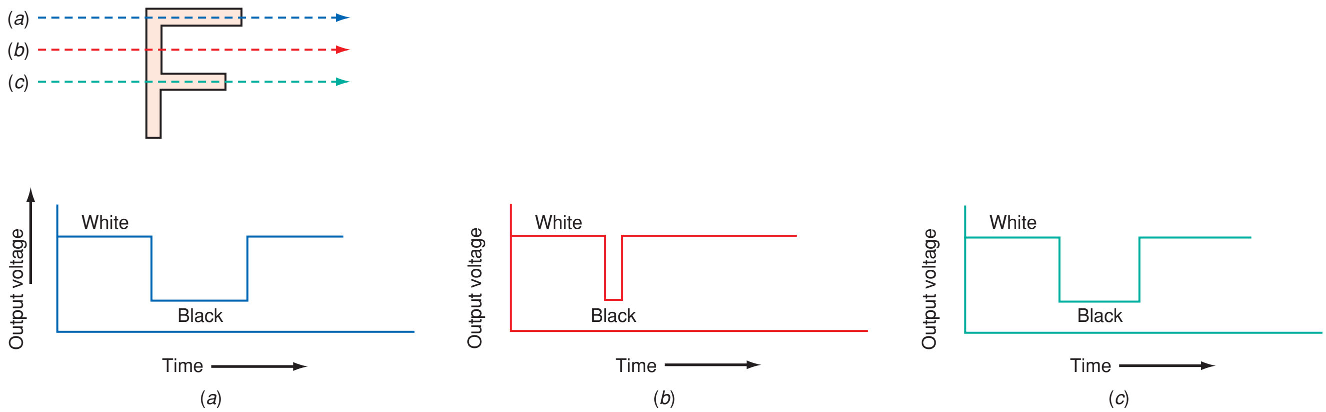


Figure 18-15 Output of a photosensitive detector during different scans.



On the receiving end, a demodulator recovered the original signal information, which was then applied to a stylus. The purpose of the stylus was to redraw the original information on a blank sheet of paper. A typical stylus converted the electric signal to heat variations that burned the image into heat-sensitive paper. Other types of printing mechanisms were used.

Today's modern fax machine is a high-tech electrooptical machine. Scanning is done electronically, and the scanned signal is converted to a binary signal. Then digital transmission with standard modem techniques is used.

Fig. 18-16 is a block diagram of a modern fax machine. The transmission process begins with an image scanner that converts the document to hundreds of horizontal scan lines. Many different techniques are used, but they all incorporate a photo- (light-) sensitive device to convert light variations along one scanned line into an electrical voltage. The resulting signal is then processed in various ways to make the data smaller and thus faster to transmit. The resulting signal is sent to a modem where it modulates a carrier set to the middle of the telephone voice spectrum bandwidth. The signal is then transmitted to the receiving fax machine over the public switched telephone network.

The receiving fax machine's modem demodulates the signal that is then processed to recover the original data. The data is decompressed and then sent to a printer, which reproduces the document. Because all fax machines can transmit as well as receive, they are referred to as *transceivers*. The transmission is half duplex because only one machine may transmit or receive at a time.

Most fax machines have a built-in telephone, and the printer can also be used as a copy machine. An embedded microcomputer handles all control and operation, including paper handling.

GOOD TO KNOW

Facsimile utilizes scanning techniques similar to those used in TV. Through this scanning, the document is broken up into many horizontal scan lines that are transmitted and reproduced serially.

Figure 18-16 Block diagram of modem fax machine.

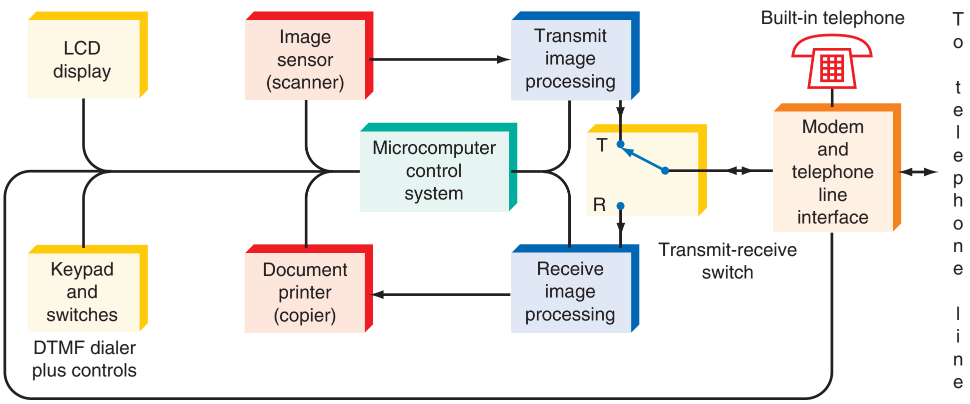


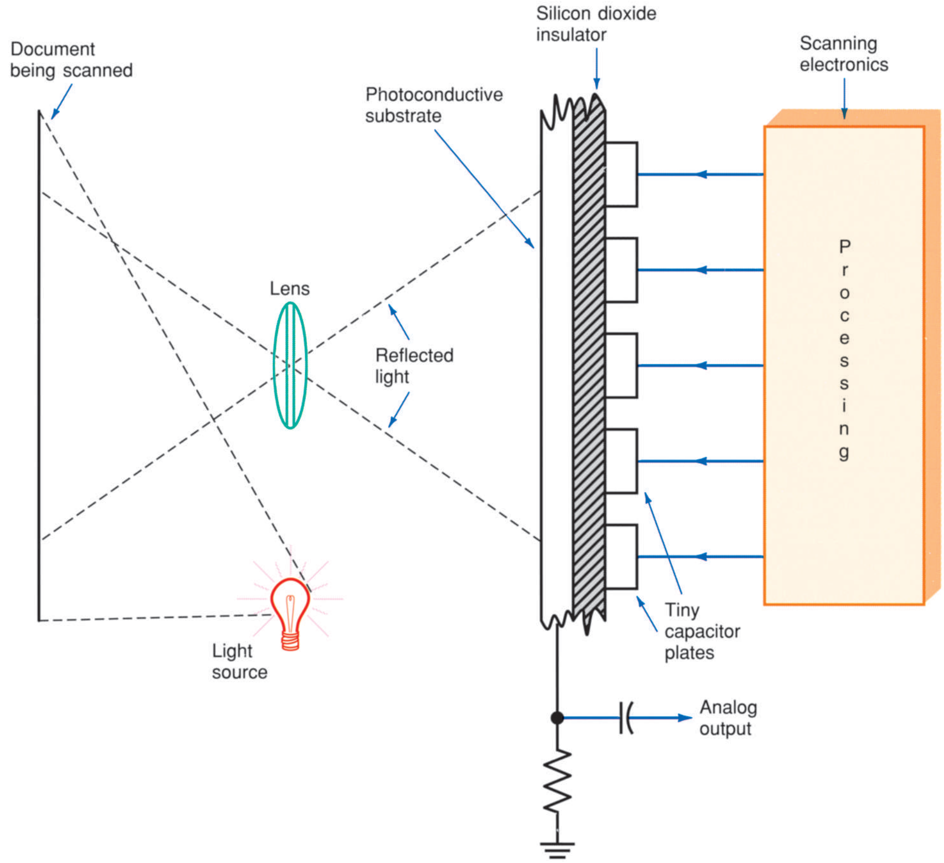
Image processing

Charge-coupled device (CCD)

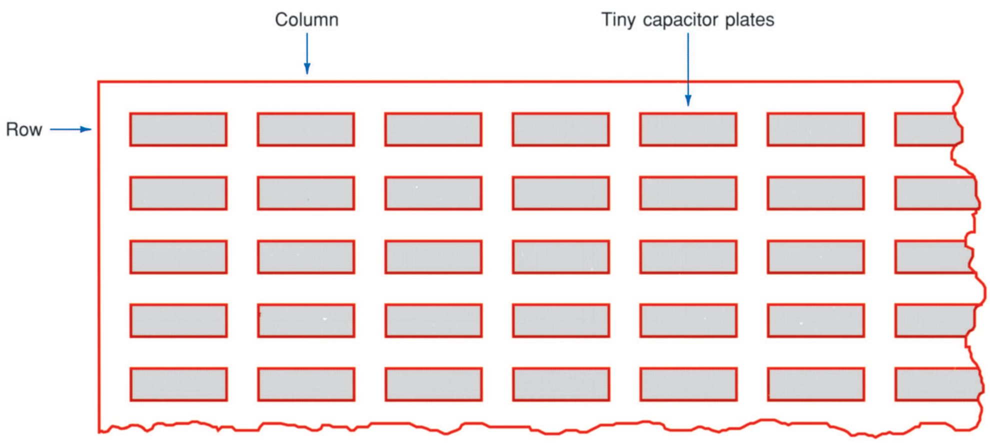
Image Processing

Most fax machines use *charge-coupled devices (CCDs)* for scanning. A CCD is a light-sensitive semiconductor device that converts varying light amplitudes to an electric signal. The typical CCD is made up of many tiny reverse-biased diodes that act as capacitors, which are manufactured in a matrix on a silicon chip (see Fig. 18-17). The base forms one large plate of a capacitor that is electrically separated by a dielectric

Figure 18-17 A charge-coupled device is used to scan documents in modern fax machines. (*a*) Cross section. (*b*) Detail of capacitor matrix.



(*a*)



(*b*)

from many thousands of tiny capacitor plates, as shown. When the CCD is exposed to light, the CCD capacitors charge to a value proportional to the light intensity. The capacitors are then scanned or sampled electronically to determine their charge. This creates an analog output signal that accurately depicts the image focused on the CCD.

A CCD is actually a device that breaks up any scene or picture into *individual picture elements,* or *pixels.* The greater the number of CCD capacitors, or pixels, the higher the resolution and the more faithfully a scene, photograph, or document can be reproduced. CCDs are available with a matrix of many thousands of pixels, thereby permitting very high-resolution picture transmission. CCDs are widely used in modern video cameras in place of the more delicate and more expensive vidicon tubes. In the video camera (camcorder), the lens focuses the entire scene on a CCD matrix. This same approach is used in some fax machines. In one type of fax machine, the document to be transmitted is placed face down as it might be in a copy machine. The document is then illuminated with brilliant light from a xenon or fluorescent bulb. A lens system focuses the reflected light on a CCD. The CCD is then scanned, and the resulting output is an analog signal whose amplitude is proportional to the amplitude of the reflected light.

Pixels

In most desktop fax machines, the entire document is not focused on a single CCD. Instead, only a narrow portion of the document is lighted and examined as it is moved through the fax machine with rollers. A complex system of mirrors is used to focus the lighted area on the CCD (see Fig. 18-18).

The more modern fax machines use another type of scanning mechanism that does not use lenses. The scanning mechanism is an assembly made up of an LED array and a CCD array. These are arranged so that the entire width of a standard 8½ × 11 in page is scanned simultaneously one line at a time. The LED array illuminates a narrow portion of the document. The reflected light is picked up by the CCD scanner. A typical scanner has 2048 light sensors forming one scan line. Fig. 18-19 shows a side view of the scanning mechanism. The 2048 pixels of light are converted to voltages proportional to the light variations on one scanned line. These voltages are converted from a parallel format to a serial voltage signal. The resulting analog signal is amplified and sent to an AGC circuit and an S/H amplifier. The signal is then sent to an A/D converter where the light signals are translated to binary data words for transmission.

Figure 18-18 Scanning mechanism in a fax machine.

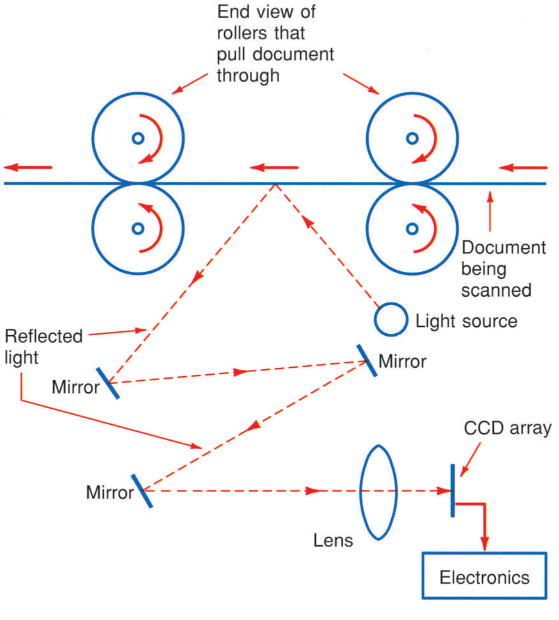
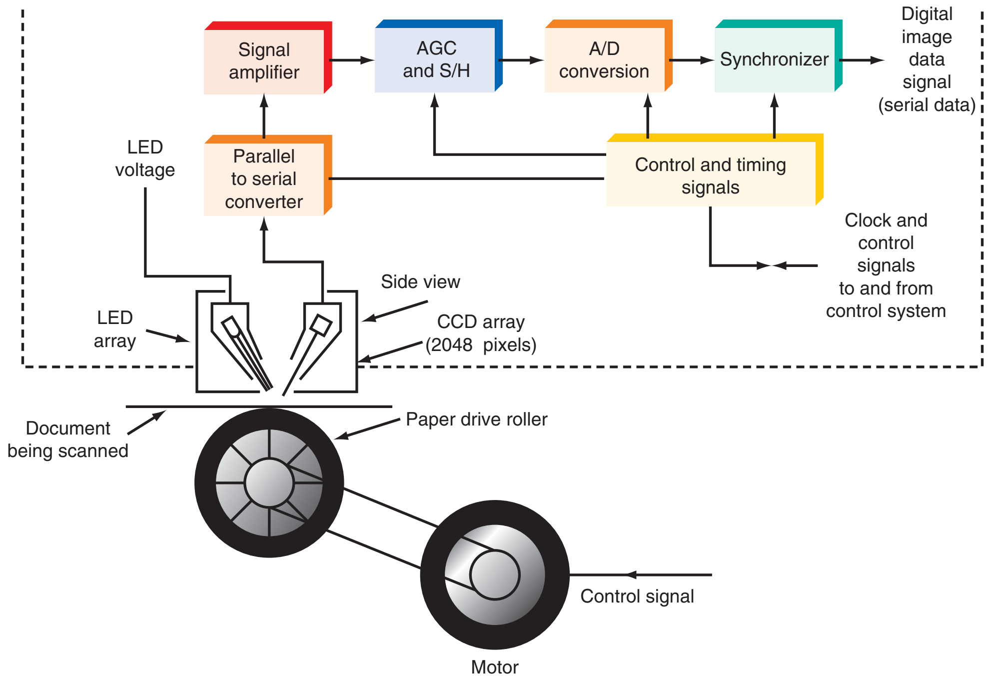


Figure 18-19 LED/CCD scanner mechanism in a modern fax machine.



Data Compression

Data compression techniques

An enormous amount of data is generated by scanning one page of a document. A typical 8½ × 11 in page represents about 40,000 bytes of data. This can be shortened by a factor of 10 or more with *data compression techniques.* Furthermore, because of the narrow bandwidth of telephone lines, data rates are limited. That is why it takes so long to transmit one page of data. Developments in high-speed modems have helped reduce the transmission time, but the most important developments are data compression techniques that reduce the overall amount of data, which significantly decreases the transmission time and telephone charges.

Data compression is a digital data processing technique that looks for redundancy in the transmitted signal. White space or continuous segments of the page that are the same shade produce continuous strings of data words that are the same. These can be eliminated and transmitted as a special digital code that is significantly faster to transmit. Other forms of data compression use various mathematical algorithms to reduce the amount of data to be transmitted.

GOOD TO KNOW

Because of the large amount of data generated through scanning, data compression techniques can be used to shorten facsimile transmission by a factor of 10 or more.

The data compression is carried out by a *digital signal processing (DSP) chip.* This is a high-speed microprocessor with embedded ROM containing the compression program. The digital data from the A/D converter is passed through the DSP chip, from which comes a significantly shorter string of data that represents the scanned image. This is what is transmitted, and in far less time than the original data could be transmitted.

At the receiving end, the demodulated signal is decompressed. Again, this is done through a DSP chip especially programmed for this function. The original data signal is recovered and sent to the printer.

Modems

Modem

Every fax machine contains a built-in *modem* that is similar to a conventional data modem for computers. These modems are optimized for fax transmission and reception. And they follow international standards so that any fax machine can communicate with any other fax machine.

A number of different modulation schemes are used in fax systems. Analog fax systems use AM or FM. Digital fax uses PSK or QAM. To ensure compatibility between fax machines of different manufacturers, *facsimile standards* have been developed for speed, modulation methods, and resolution by the *International Telegraph and Telephone Consultative Committee,* better known by its French abbreviation, *CCITT.* The CCITT is now known as the *ITU-T,* or *International Telecommunications Union.* The ITU-T fax standards are divided into four groups:

Facsimile standards

International Telegraph and Telephone Consultative Committee (CCITT)

International Telecommunications Union (ITU-T)

1. ***Group 1 (G1 or GI):*** Analog transmission using frequency modulation where white is 1300 Hz and black is 2100 Hz. Most North American equipment uses 1500 Hz for white and 2300 Hz for black. The scanning resolution is 96 lines per inch (LPI). Average transmission speed is 6 minutes per page (8½ × 11 in or A4 metric size, which is slightly longer than 11 in).
2. ***Group 2 (G2 or GII):*** Analog transmission using FM or vestigial sideband AM. The vestigial sideband AM uses a 2100-Hz carrier. The lower sideband and part of the upper sideband are transmitted. Resolution is 96 LPI. Transmission speed is 3 min or less for an 8½ × 11 in or A4 page.
3. ***Group 3 (G3 or GIII):*** Digital transmission using PCM black and white only or up to 32 shades of gray. PSK or QAM to achieve transmission speeds of up to 9600 Bd. Resolution's 200 LPI. Transmission speed is less than 1 minute per page, with 15 to 30 s being typical.
4. ***Group 4 (G4 or GIV):*** Digital transmission, 56 kbps, resolution up to 400 LPI, and speed of transmission less than 5 s.

The older G1 and G2 machines are no longer used. The most common configuration is group 3. Most G3 machines can also read the G2 format.

The G4 machines are not yet widely used. They are designed to use digital transmission only with no modem over very wideband dedicated digital-grade telephone lines. Both G3 and G4 formats also employ digital data compression methods that shorten the binary data stream considerably, thereby speeding up page transmission. This is important because shorter transmission times cut long-distance telephone charges and reduce operating costs.

Fax Machine Operation

Fig. 18-20 is a simplified block diagram of the transmitting circuits in a modern G3 fax transceiver. The analog output from the CCD array is serialized and fed to an A/D converter that translates the continuously varying light intensity into a stream of binary numbers. Sixteen gray scale values between white and black are typical. The binary data is sent to a DSP digital data compression circuit as described earlier. The binary output in serial data format is used to modulate a carrier that is transmitted over the telephone lines. The techniques are similar to those employed in modems. Speeds of 2400/4800 and 7200/9600 Bd are common. Most systems use some form of PSK or QAM to achieve very high data rates on voice-grade lines.

GOOD TO KNOW

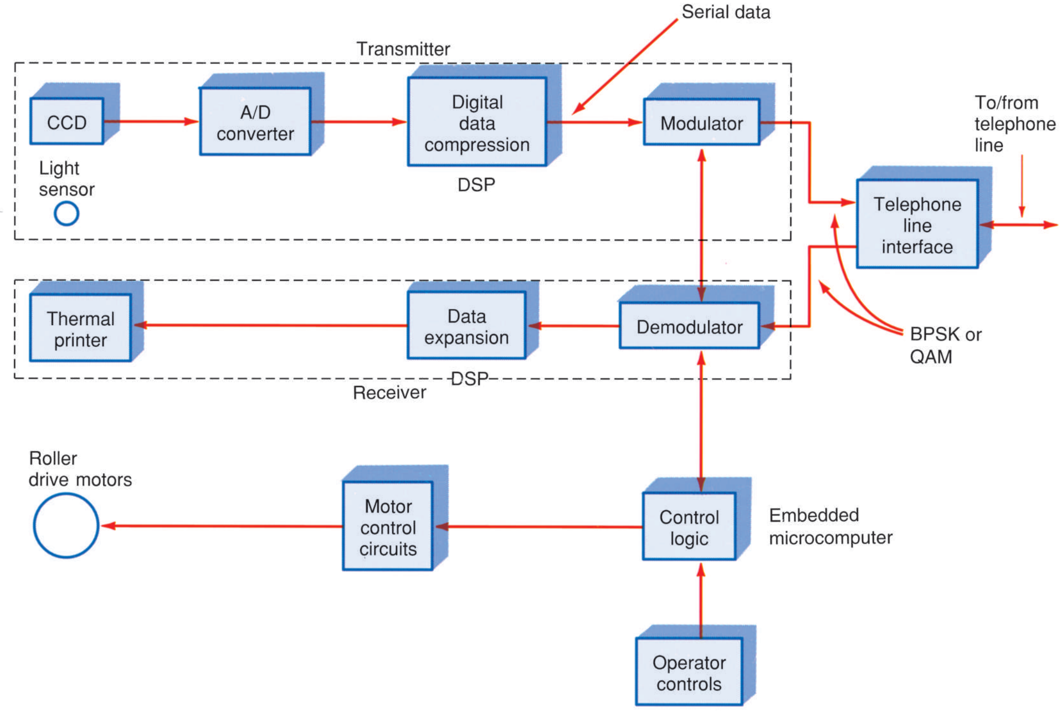
The analog signal from the charge-coupled device is converted to digital values that are sent to a DSP chip.

In the receiving portion of the fax machine, the received signal is demodulated and then sent to DSP circuits, where the data compression is removed and the binary signals are restored to their original form. The signal is then applied to a printing mechanism.

The most common fax printer today is an ink jet printer like those popularly used with PCs. In the high-priced machines, laser scanning of an electrosensitive drum, similar to the drum used in laser printers, produces output copies by using the proven techniques of xerography.

The control logic in Fig. 18-20 is usually an embedded microcomputer. Besides all the internal control functions it implements, it is used for "handshaking" between the two machines that will communicate. This ensures compatibility. Handshaking is usually carried out by exchanging different audio tones. The called machine responds with tones designating its capability. The calling machine compares this to its own standards and then either initiates the transmission or terminates it because of incompatibility. If the

Figure 18-20 Block diagram of a facsimile machine.



transmission proceeds, the calling machine sends synchronizing signals to ensure that both machines start at the same time. The called machine acknowledges the receipt of the sync signal, and transmission begins. All the protocols for establishing communication and sending and receiving the data are standardized by the ITU-T. Transmission is half duplex.

As improvements have been made in picture resolution quality, transmission speed, and cost, facsimile machines have become much more popular. The units can be easily attached with standard RJ-11 modular connectors to any telephone system. In most business applications, the fax machine is typically dedicated to a single line. Most fax machines feature fully automatic operation with microprocessor-based control. A document can be sent to a fax machine automatically. The sending machine simply dials the receiving machine and initiates the transmission. The receiving machine answers the initial call and then reproduces the document before hanging up.

Most fax machines have a built-in telephone and are designed to share a single line with conventional voice transmission. The built-in telephone usually features Touch-Tone dialing and number memory plus automatic redial and other modern telephone features. Most fax machines also have automatic send and receive features for fully unattended operation.

Fax machines are slowly fading away as technology changes. Today, most computer printers incorporate a scanner and a printer. The fax function including a data-only telephone with RJ-11 connection is built into the printer. A scanned document the digitized and sent using the fax procedures described earlier.

18-4 Internet Telephony

Internet Protocol (IP)

Voice over Internet Protocol (VoIP)

Internet telephony, also called *Internet Protocol (IP)* telephony or *Voice over Internet Protocol (VoIP),* uses the Internet to carry digital voice telephone calls. VoIP, in effect, for the most part, bypasses the existing telephone system, but not completely. It has been in development for over a decade, but only recently has it become practical and popular. VoIP is a highly complex digital voice system that relies on high-speed Internet connections from cable TV companies, phone companies supplying DSL, and other broadband systems including wireless. It uses the Internet's vast fiber-optic cabling network to carry phone calls without phone company charges. This new telephony system is slowly replacing traditional phones, especially in large companies. It offers the benefits of lower long-distance calling charges and reduces the amount of new equipment needed, because phone service is essentially provided over the same local-area network (LAN) that interconnects the PCs in an organization. VoIP is rapidly growing in use and in the future is expected to replace standard phones in many companies and homes. While the legacy PSTN will virtually never go away, over time it will play a smaller and smaller role as VoIP is more widely adopted or as more and more individuals choose a cell phone as their main telephone service.

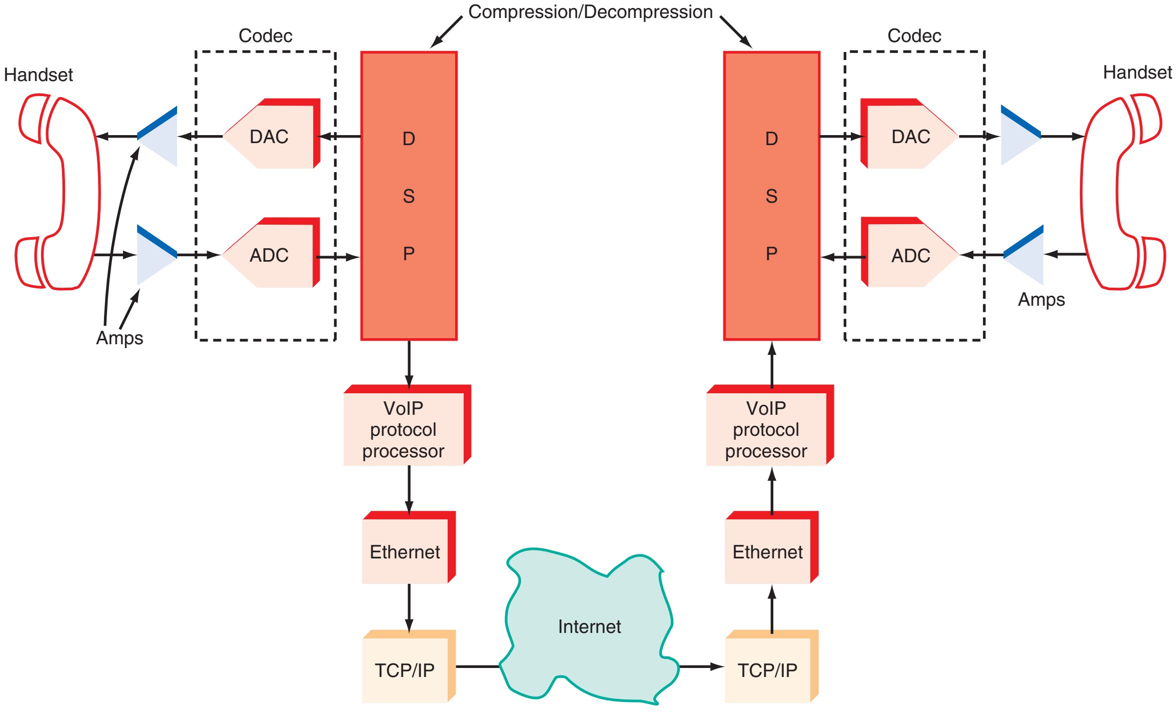
VoIP Fundamentals

There are two basic parts to an IP phone call: the "dialing" process, which establishes an initial connection, and the voice signal flow.

Voice Signal Flow. Fig. 18-21 shows the signal flow and major operations that take place during an IP phone call. The voice signal is first amplified and digitized by an analog-to-digital converter (ADC) that is part of a coder-decoder (codec) circuit, which also includes a digital-to-analog converter (DAC). The ADC usually samples the voice signal at 8 kHz and produces an 8-bit word for each sample. These samples occur one after another serially and therefore produce a 64-kbps digital signal. A relatively wide bandwidth is needed to transmit this bit stream (64 kHz or more). To reduce the data rate and the need for bandwidth, the bit stream is processed by a voice encoder that compresses the voice signal. This compression is usually done by DSP either in a separate DSP processor chip or as hardwired logic on a larger chip. The output is at a greatly reduced serial digital data rate.

The type of compression used is determined by International Telecommunications Union standards. Various mathematical algorithms beyond the scope of this text are used.

Figure 18-21 Signal flow in a VoIP system.



The 64-kbps digital signal is designated as standard G.711 and is better known as *pulse-code modulation (PCM),* covered earlier in this book. Standard G.729a is probably the most common compression standard used and results in an 8-kbps digital voice signal. Another popular standard is G.723, which produces an even more highly compressed 5.3-kbps signal at the expense of some voice quality. Numerous other compression standards are used, and they are selected based upon the application. Most VoIP phones contain all the common compression standard algorithms in the DSP memory for use as called for. The signal is also processed in the DSP to provide echo cancellation, a problem in digital telephony.

The resulting serial digital signal is put into a special packet by a microcomputer processor running a VoIP protocol and then transmitted by Ethernet over a LAN or via a high-speed Internet connection such as is available from a cable TV company or on DSL. From there the signal travels over standard available Internet connections using TCP/IP through multiple servers and routers until it comes to the desired location.

At the receiving phone, the process is reversed. The Internet signal gets converted back to Ethernet, and then the VoIP processor recovers the original packet. From there, the compressed data is extracted, decompressed by a DSP, and sent to the DAC in the codec where the original voice is heard.

One of the main problems with VoIP is that it takes a relatively long time to transmit the voice data over the Internet. The packets may take different routes through the Internet. They all do eventually arrive at their intended destination, but often the packets are out of sequence. The receiving phone must put them back together in the correct sequence. This takes time.

Latency

Furthermore, even though the signals traverse the high-speed optical Internet lines at gigabit speeds, the packets pass through numerous routers and servers, each adding transit time or latency. *Latency* is the delay between the time the signal is transmitted and the time it is received. It has been determined that the maximum acceptable latency is about 150 ms. Any longer time is noticeable by the user. One party may have to wait a short time before responding to avoid talking while the signal is still be received. This annoying wait is unacceptable to most. Keeping the latency below 150 ms minimizes this problem.

Session initiation protocol (SIP)

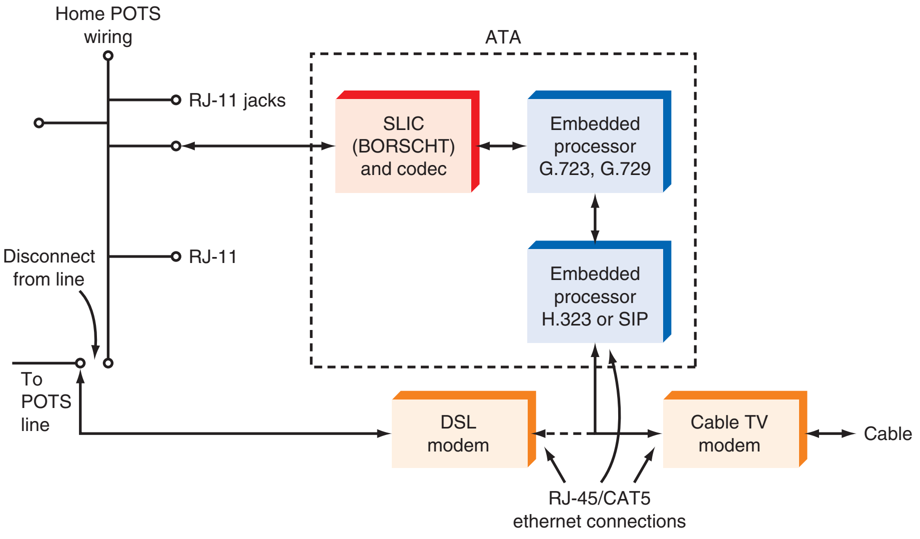
Link Establishment. In the PSTN, the dialing process initiates multiple levels of switching that literally connects the calling phone to the called phone. That link is maintained for the duration of the call because the switches stay in place and the electronic paths stay dedicated to the call. In Internet telephony, no such temporary dedicated link is established because of the packetized nature of the system. Yet some method must be used to get the voice data to the desired phone. This is taken care of by a special protocol developed for this purpose. The initial protocol used was the ITU H.323. Today, however, a newer protocol established by the Internet Engineering Task Force (IETF) called the *session initiation protocol (SIP)* has been adopted as the de facto standard. In both cases, the protocol sets up the call and then makes sure that the voice packets produced by the calling phone get sent to the receiving phone in a timely manner.

Internet Phone Systems

There are two basic types of IP phones: those used in the home and those used in larger organizations. The concepts as described above are the same for both, but the details are slightly different.

Home VoIP. To establish IP phone service in the home, the subscriber must have some form of high-speed Internet service. Cable TV provides this service in most homes, but it can also be provided over the standard POTS local loop with DSL. In addition, the subscriber must have a VoIP interface. This is called different things by the different service providers. A common example is the Analog Terminal Adapter (ATA). This device

Figure 18-22 Analog terminal adapter (ATA) or VoIP gateway.



connects the standard home telephone to the existing broadband Internet modem. Another configuration is a VoIP gateway that contains the ATA circuitry as well as the broadband modem.

A general block diagram of an ATA is shown in Fig. 18-22. Notice that the ATA allows standard telephones and cordless phones to attach to the ATA via the usual RJ-11 modular plug. In fact, the input to the ATA is the phone wiring in the home. The home wiring is disconnected from the subscriber interface at the connection provided outside the home by the phone company. In this way, any of the available home phones can be used with the ATA over installed wiring. Note in the figure that because standard phones are used, they must be provided with SLIC BORSCHT functions. The SLIC circuitry is usually packaged in a single IC chip, and often the codec is also contained on this chip.

The codec inputs and outputs go to one or more processors where the H.323 or SIP protocol is implemented and where the DSP functions for compression and decompression reside. An Ethernet interface is also provided. The Ethernet signal connects to the broadband modem for cable TV service or DSL. If the cable modem is used, the POTS and last mile local loop are simply not used. However, if DSL service provides the broadband connection, the POTS connection is used for the DSL modem. The home phone wiring must be disconnected from the POTS line as described earlier.

Enterprise IP Phones. IP phones in companies or large organizations are especially designed for VoIP service. The telephone set contains all the ATA circuitry except for the SLIC and connects directly to the available Ethernet connection usually supplied to each desk. No broadband modem is needed. Since most employees will also have a PC connected to the LAN, a two-port Ethernet switch in the phone or PC provides a single Ethernet connection to the LAN that the phone and PC share.

A major benefit of IP phones is that they may also use wireless Ethernet connections. Wireless Ethernet, generally called Wi-Fi or the IEEE standard designation 802.11, is widely used to extend the LANs in most companies. If the IP phone is equipped with a wireless Ethernet transceiver, then no wired connection is needed. Already some cell phone manufacturers are including Wi-Fi VoIP in some models. In this way, a person's cell phone works outside the company with the standard cell site service but also serves as the person's company phone with a wireless Ethernet connection when inside the company. Wireless systems such as this are covered in Chap. 22.

CHAPTER REVIEW

Online Activity

18-1 The Future of the Telephone System

Objective: To investigate the trends related to the PSTN.

Procedure:

1. Search on the terms *PSTN, POTS, future of the telephone system,* and similar expressions.
2. Answer the questions below.

Questions:

1. What one factor has resulted in the decline in the number of households with standard POTS subscriptions?
2. Approximately what percentage of households have a standard wired phone?
3. When will analog phone service be phased out?
4. How will the local loop be used if analog phone service is phased out?
5. What wired service is gradually replacing the local loop?

Questions

1. Define specifically what is meant by the *local loop*.
2. What type of power supply is used to power a standard telephone, what are its specifications, and where is it located?
3. State the characteristics of the ringing signal supplied by the telephone company.
4. What is a hybrid?
5. True or false? Most telephone companies can still accommodate pulse dial telephones.
6. What type of transmitter (microphone) is used in a standard telephone, and how does it work?
7. Define what is meant by *tip* and *ring* and state the colors used to represent them.
8. What is the name of the TouchTone dialing system?
9. What two tone frequencies are generated when you press the pound key (#) on a TouchTone phone?
10. What is the name of the building or facility to which every telephone is connected?
11. What kind of microphone is used in an electronic telephone?
12. What is the purpose of the bridge rectifier circuit at the input to the connection of the telephone to the line to the telephone company?
13. Name one type of low-cost sound device used to implement the bell or ringer in an electronic telephone.
14. True or false? In an electronic telephone, the hybrid is a special type of transformer.
15. Give two names or designations for the standard connector used on telephones.
16. State the four frequency ranges used by cordless telephones, and tell which type of modulation is used.
17. What do you call the circuits that make up the connections to each telephone at the telephone office?
18. What is a key telephone system?
19. Does your local telephone company supply long-distance service?
20. Define POTS and PSTN.
21. True or false? Fax can transmit photographs and drawings as well as printed text.
22. What is the most common transmission medium for fax signals? What other medium is commonly used?
23. True or false? Facsimile was invented before radio.
24. Who sets the standards for fax transmission?
25. Vestigial sideband AM is used in what group type of fax machines?
26. What is the name of the semiconductor photosensitive device used in most modern fax machines to convert a scanned line to an analog signal?
27. What is the group designation given to most modern fax machines?
28. To ensure compatibility between sending and receiving fax machines, the control logic carries out a procedure by using audio tones to establish communications. What is this process called?
29. What circuit in the fax machine makes the fax signal compatible with the telephone line?
30. What is the upper speed limit of a G3 fax machine over the telephone lines?
31. What is the resolution of a G3 fax machine in lines per inch?
32. Is fax transmission usually full duplex or half duplex?
33. Are fax signals representing the image to be transmitted before they are prepared for the telephone lines analog or digital?
34. True or false? Group 4 transmissions do not use the standard telephone lines.
35. What are the speed and resolution of group 4 fax transmissions?
36. What does caller ID do?
37. Explain how caller ID data is transmitted to the telephone.
38. Describe the modulation and data rate used in caller ID.
39. Name two ways that caller ID can be blocked.

40. Name the three frequency ranges used by cordless telephones.
41. What are the three primary disadvantages of the older analog cordless phones?
42. Name three types of 900-MHz cordless phone.
43. What type of modulation is used in the best digital cordless phones?
44. Why are the 2.4-GHz and 5.8-GHz cordless phones inherently secure?
45. What is the approximate maximum distance over which a digital cordless phone can transmit?
46. What is the designation and name of the newest digital cordless telephone standard?
47. Define signaling in the PSTN.
48. Name the signaling system used in the United States.
49. Do VoIP phone systems use the PSTN? Explain.
50. What is the name of the circuit that does the data conversion in an IP phone?
51. What are the designations of the basic protocol standard used to implement VoIP?
52. What are some of the standards for compression and decompression used in IP phones?
53. Why is compression needed in IP phones?
54. How are compression and decompression accomplished in an IP phone?
55. What two transmission protocols are widely used in sending and receiving IP phone calls?
56. Give two names of the equipment used in a home to implement VoIP.
57. True or false? A home VoIP system can use regular analog phones.
58. True or false? An office IP phone needs an SLIC.
59. What effect greatly degrades the quality of a VoIP call and makes it annoying? What causes this problem?

Problems

1. What is an SLIC?
2. List the basic BORSCHT functions that are performed by the telephone company for every telephone. ◆
3. Explain what each of the number groups in a 10-digit telephone number mean.
4. Describe the types of possible links between telephone exchanges. Are they two-line or four-line? ◆
5. Briefly define the terms *LATA, LEC, POP,* and *IXC.* Which one means "long-distance carrier"?
6. State the name and basic specifications and benefits of the newest class of digital cordless telephone.
7. Explain the process and hardware used to convert images to be transmitted to electric signals in a fax machine. ◆
8. Describe two methods of scanning used in modern fax machines.
9. What is the most commonly used type of printer in a fax machine?
10. Describe how fax signals are processed to speed up transmission.
11. What is the basic frequency response of the telephone local loop? Can it carry digital as well as analog signals?
12. What is the serial data speed of a G.711 VoIP signal?
13. What are the data rates of G.723 and G.729a compressed VoIP voice signals?
14. State the maximum allowable latency in an IP phone call.

◆ *Answers to Selected Problems follow Chap. 22.*

Critical Thinking

1. Discuss how it would be possible to use the standard ac power lines for telephone voice transmission. What circuits might be needed, and what would be the limitations of this system?
2. Explain the factors that limit the range of a cordless telephone.