CHAPTER 17.7
TERMINAL EQUIPMENT

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TELEPHONES

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Telephone equipment ranges from the familiar desk or wall telephone set to the versatile communications system terminal of the information age. Telecommunications has merged with compute technologies in telephones to make available the entire spectrum of voice, data, video, and graphics. Terminal equipment now allows the exchange of this information over the telephone network.

The Telephone Set

The basic functions of the telephone set include signaling, alerting, central office supervision, and transmission of voice communications. In a typical call sequence, when the caller (the near-end party) picks up the handset, the telephone draws loop current (the telephone line is known as a *loop*) from the central office battery, which signals the central office (CO) that it wants service. The loop current also provides power for telephone functions. The caller then dials, sending address signals to the central office by either pulse or tone dialing. The CO collects the address signal in registers and sets up a transmission path with the CO for the number being called. The called CO sends an alerting signal to the called telephone, causing it to ring. When the called or far-end party picks up their handset, loop current is drawn signaling the CO to trip (interrupt) ringing and complete the talking circuit.

The functional elements of traditional telephones (Fig. 17.7.1) include a carbon transmitter to convert acoustic energy to an electrical voice signal, an electromagnetic receiver to convert the electrical voice signal back into acoustic energy, a switch hook to turn the telephone on and off, rotary dial contacts, which make and break loop current, a loop-equalizer circuit to compensate for loop resistance, a balance circuit, a hybrid transformer for coupling the transmitter and receiver to the telephone line, and an electromechanical ringer. The two-wire telephone line connections are known as Tip and Ring. The loop equalizer, balance circuit, and hybrid transformer are collectively known as the speech network. Such traditional speech networks are called *passive* networks. Electronic speech networks using solid-state components are called *active* networks.

The ringer is shown bridged across the telephone line. The capacitor C_1 blocks the flow of loop current through the ringer. Resistor R_1 and varistor V_1 constitute the loop-equalizer circuit. On long loops with low loop current, varistor V_1 maintains a high resistance and takes little current away from the rest of the speech network. On short loops, higher levels of loop circuit result in a lower resistance of V_1, thereby reducing the transmit and receive levels.

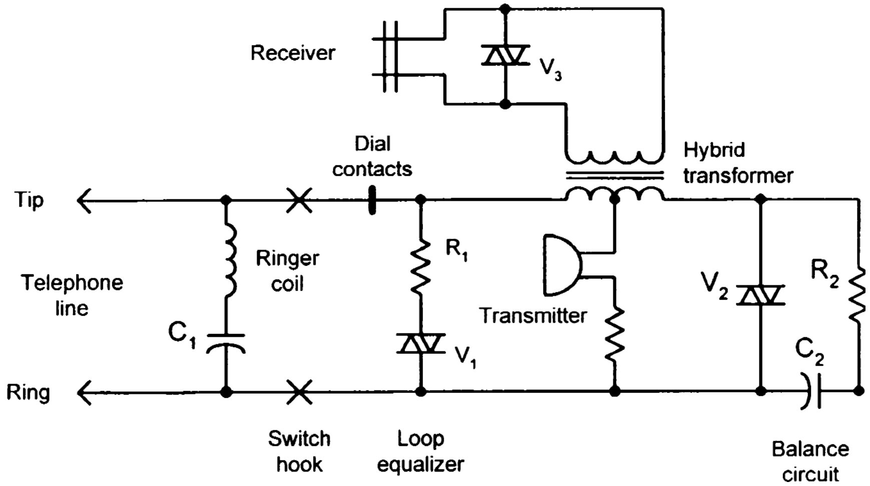

FIGURE 17.7.1 Traditional passive network telephone set.

The combination of a three-winding hybrid transformer and impedance balancing circuitry provides the means of coupling the transmitter and receiver to the loop independently. This is called an *antisidetone* network. Sidetone is that portion of the transmitted signal that is heard in the receiver while talking. Sidetone is subjectively desirable because it provides the live quality of face-to-face conversation. The antisidetone network is designed to provide a sidetone signal at about the same level as received speech. If the sidetone level is too high, the talker tends to speak softly to keep the sidetone level pleasant, which results in signal strength too low for good transmission. If the sidetone level is too low, the talker perceives the telephone as dead or inoperative.

Incoming voice signals from the telephone loop are transformer coupled to the receiver. The induction voltages are such that most of the incoming signal power is delivered to the receiver with little power to the balance network. Outgoing voice signals generated by the transmitter induce voltages in two of the transformer windings that cancel each other, so that most of the signal power is divided between the balance circuit resistor R_2 and loop impedances with little to the receiver. The choice of impedance and turns ratios provides a compromise in sidetone balance and impedance matching to the telephone line. Capacitor C_2 prevents dc power from being dissipated in R_2. Varistor V_2 helps match the balance circuit impedance to the loop impedance.

The main advantage of an active over a passive network is its smaller physical volume, lower cost, and greater versatility. An active network also provides power gain, thus allowing the use of microphones such as electrets. In the active network the gain of the transmit and receiver amplifiers can be automatically adjusted, depending on the loop current, to provide loop equalization.

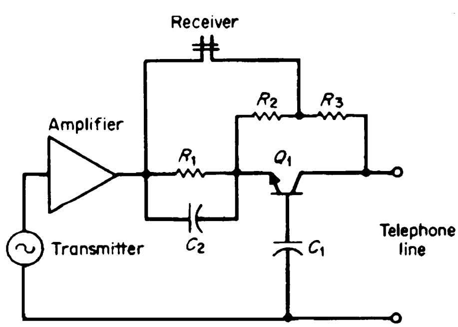

FIGURE 17.7.2 Typical active-network circuit.

A basic active network is shown in Fig. 17.7.2. The base of Q_1 is returned to common (at voice frequencies) by capacitor C_1. The emitter of Q_1 is virtual ground, since its low base impedance is divided by the transistor's beta. A received signal appearing on the telephone line is routed to the receiver through the voltage divider consisting of R_2 and R_3; R_2 is connected to a virtual ground. The other end of the receiver is returned to common through the low output impedance of the transmit amplifier.

The transmit signal is first amplified by the amplifier and further amplified by transistor Q_1. The voltage gain of this common-base state is determined by the input impedance of the telephone line and the impedance of R_1 in parallel with C_2. The antisidetone balance is achieved by adjusting the voltage divider (R_2 and R_3) to compensate for the gain of the common-base stage, leaving about the same potential at both

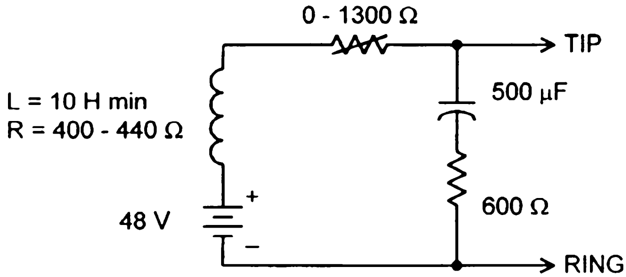

FIGURE 17.7.3 Loop simulator.

ends of the receiver. Capacitor C_2 is added to minimize any phase shift through this stage caused by the capacitance of the telephone line.

Transmission

Terminal equipment is part of a transmission and signaling circuit set up by a network provider for voice frequency transmission in the range of 300 to 3300 Hz. Four important characteristics a telephone must have in order to work properly in this circuit are dc resistance (for dc powering and loop supervision), ac impedance at 1000 Hz, signal power level at both the receiver and transmitter, and audio frequency response.

A simple loop simulator is shown in Fig. 17.7.3. A ring generator (86 V, 20 Hz, 400 Ω), present only during ringing, is not shown. DC power is provided by a nominal 48-V battery. Loop current is limited by the resistance of relay coils or current limiting circuits in the central office, and by the resistance of the loop itself. Maximum loop resistance is 1300 Ω, which is 15,000 ft of 26 AWG cable. With a 300-Ω telephone, loop current is about 20 to 80 mA. At 1000 Hz the transmission cable has a characteristic impedance of 600 Ω, which the telephone should match for maximum energy transfer and minimum echo.

Typical signal levels are shown in Fig. 17.7.4. Desirable sound power at the receiver was determined by subjective testing of people. The transmitter, while converting acoustic power into electrical energy, must also amplify the energy by about 20 dB to compensate for the 20-dB loss in converting the electrical energy back into acoustic power. The loop resistance provides an attenuation of 2.8 dB/m for 26 AWG cable. Today, virtually all central offices' trunks are digital, so there is no transmission loss between them.

Telephone receivers and transmitters are designed to achieve desired frequency characteristics. For example, telephone handset microphones have a rising response characteristic near 3 kHz to compensate for capacitive shunting loss in the loop and simulate the effects of acoustic diffraction about the human head that is present in face-to-face conversation.

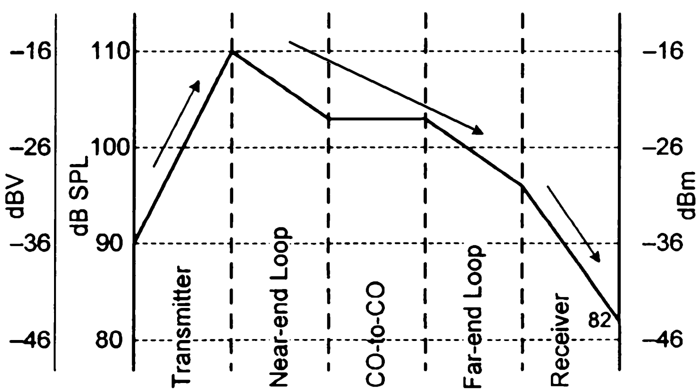

FIGURE 17.7.4 Transmission circuit signal levels.

Multifunction Electronic Telephones

Most new telephones are electronic. Conventional components, such as the bell ringer, the hybrid transformer, and the mechanical dial (rotary or push button) are replaced by active electronic devices that are incorporated into large-scale integrated (LSI) circuits. A typical dial-pulse electronic telephone set contains at tone ringer, an active network, an electronic dial keypad, a dial-pulsing transistor, and a low-power microphone. Several advantages result. The overall reliability of the telephone increases, automated manufacturing assembly is possible, telephone-set weight and size are reduced, and finally, overall transmission performance of the telephone is improved.

A diagram of a typical microcomputer-controlled multifunction electronic telephone is given in Fig. 17.7.5. Telephone features include last-number redial, repertory dialing, dial-tone detectors, speakerphone, integrated answer/record, hold, conference (for two-line telephones), and display of the dialed digits. Repertory dialing permits the user to store several telephone numbers in an auxiliary memory for automatic dialing. The dial circuit can produce pulse dialing or dual-tone multifrequency (DTMF) tones. The architecture often includes both general-purpose and custom LSI circuits, such as DTMF generator chips, clock (timer) chips, and display-driver chips, or it may contain one very large-scale integrated (VLSI) circuit. The microcomputer controls the operation of the various LSI circuits. The microcomputer receives information from the ringing detector, the dial keypad, function buttons, the electronic line switch, and the active network, and controls such items as the tone generator (ringer), the integrated answer/record system, the dial circuits, the display, and the speakerphone.

Electronic logic performs a variety of common switching functions, such as switching out the transmitter and lowering the gain to the receiver during dialing, functions performed by mechanical switches in traditional telephone sets. The line switch may also be electronic. The switch hook, rather than closing line circuit contacts when the handset is lifted, turns on a solid-state line switch. The user can also turn the telephone on and off electronically without having to lift the handset. Speakerphone, answer/record, and hold operations are common uses for an electronic line switch.

The media for message storage is either tape or solid-state electronic memory. Audio storage on tape uses standard tape deck recording and playback techniques. Storage in solid-state memory requires conversion to a digital format with the use of a CODEC. This digital data is stored and retrieved under control of a microprocessor. To further conserve relatively expensive memory, a DSP can be used to massage the data. Dead time is removed, and various compression algorithms are used to conserve memory. Here a trade-off is made between the amount of memory needed and the quality of the speech desired.

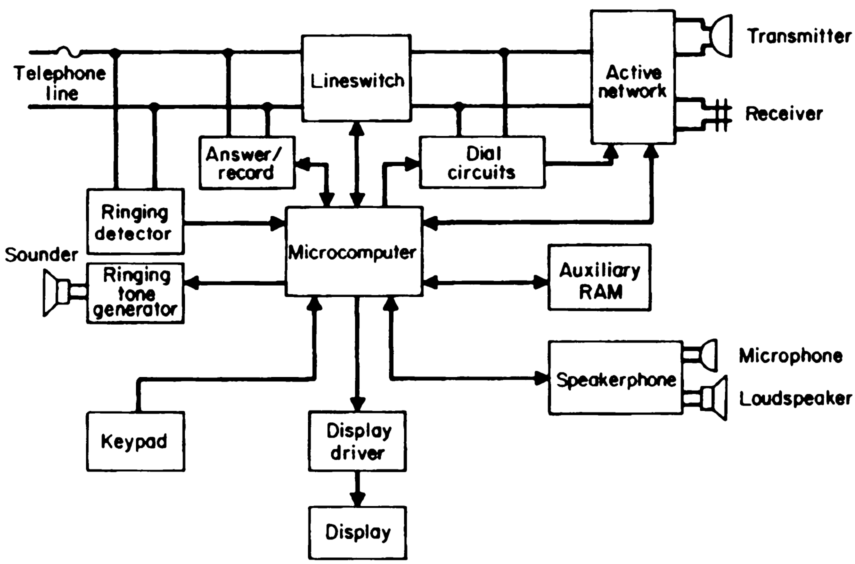

FIGURE 17.7.5 Multifunction electronic telephone.

Speakerphones must pick up much lower speech signals than a handset transmitter and must generate more acoustic energy than a handset receiver can, so it often requires more power than can be drawn over the telephone line. To prevent feedback from the loudspeaker to the microphone, the loudspeaker is muted when speech (or noise) is detected. Because the microphone must be very sensitive to pick up ordinary conversation it also picks up room reverberation, which causes speech to sound as if the speaker is in a tin can. New highly directional microphones can minimize unwanted echo and produce more natural speech.

Cordless Telephones

In a cordless telephone, the usual telephone functions are performed over a radio link, thereby eliminating the handset cord and providing the user with added mobility. A cordless telephone block diagram (Fig. 17.7.6) shows a portable handset unit, used for talking, listening, dialing, and ringing, and a fixed base unit, used for interfacing between the telephone line and radio link. More sophisticated applications include units with duplex intercoms and base units with integrated speakerphones or telephone answer/record devices.

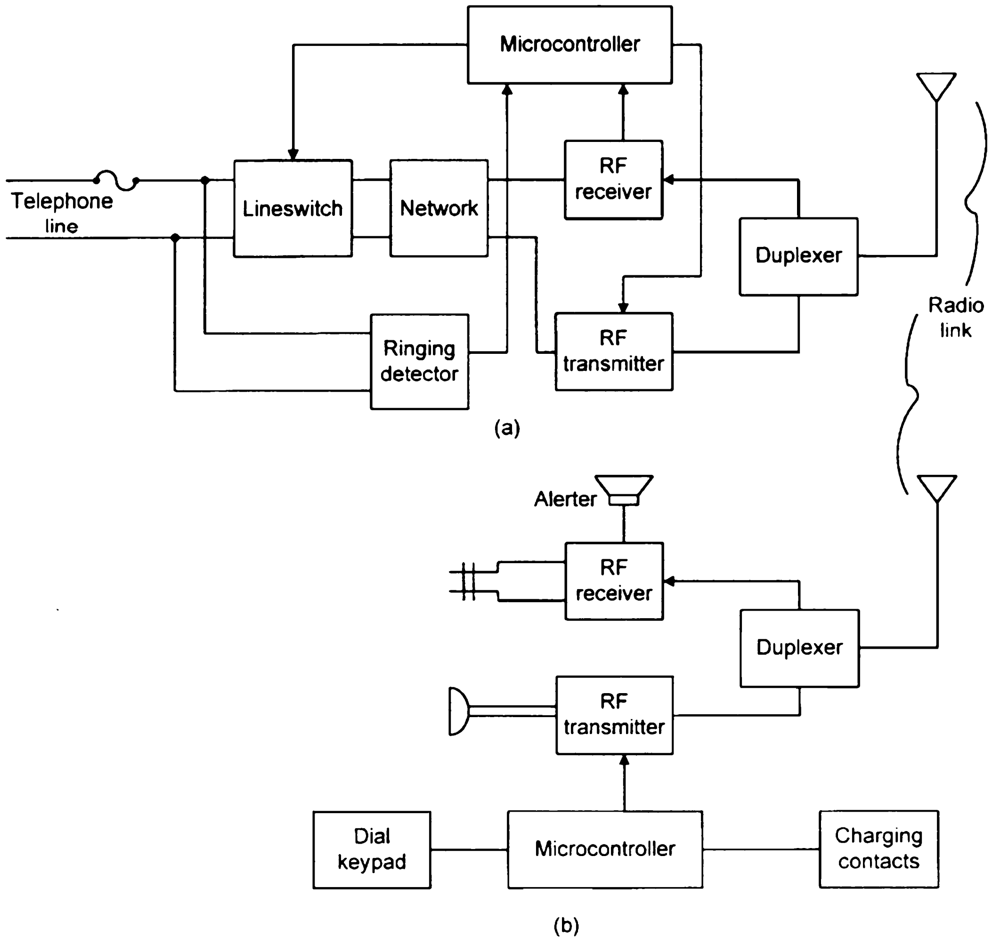

FIGURE 17.7.6 Cordless telephone: (*a*) base unit and (*b*) handset unit.

The normal listening and talking signals as well as ringing and DTMF signaling are transmitted over a radio-frequency link as a frequency-modulated carrier. This carrier is centered in one of ten 20-kHz-wide channels with the base-to-handset channels in the 46.6- to 47-MHz band and the handset-to-base channels in the 49.6- to 50-MHz band. Before 1983 the base-to-handset link used the 1.6- to 1.7-MHz band and there were only five FCC allocated channels. As cordless telephones became more popular, interference between people using the same channel became common. To minimize the probability of hearing conversations from nearby cordless telephones on the same channel, the FCC limits the maximum radiated field strength to 10,000 (V/m at 3 m, which allows satisfactory cordless telephone performance up to 1000 ft from the base station under ideal conditions. To minimize interference, some telephones scan the available channels and choose any vacant channel found.

Typical cordless telephones employ full duplex signaling between handset and base using *frequency shift keying (FSK)* modulation of the carrier. By embedding a digital code in all exchanges of information between handset and base, it is possible to virtually eliminate false operations and ringing. A security code is also embedded to prevent access of the base unit by a neighbor's handset.

Increasing user traffic in the 46/49 MHz frequency bands has caused renewed congestion. Therefore, the FCC has allowed use of one of the commercial bands with frequencies ranging from 902 to 928 MHz, using either analog or digital modulation schemes. Units using digital communication provide a higher degree of security because they do not allow simple FM demodulation by scanners or FM receivers, thereby preventing eavesdropping. This band also has less congestion and allows better RF propagation throughout the usable area, decreasing noise and interference.

Video Telephones

New video telephones provide motion video in color over the public switched telephone network. Advancements made in audio and video compression algorithms have made this possible. See Early et al. (1993). Previous video systems required a special transmission line, making them suitable only for business users. The video telephone first establishes a talking connection with another video telephone, then video mode is entered by pressing a "video" button on each telephone. In "video" mode compressed audio and video signals are transmitted between the two telephones.

The full-duplex DSP-based modem uses a four-dimensional, 16-state trellis code and supports data rates of 19.2 and 16.8 kb/s. The 19.2 kb/s mode transmits 6 bits per baud at 3200 baud, and the 16.8 kb/s mode transmits 6 bits per baud at 2800 baud. The modem will automatically drop back to the slower speed if the error rate indicates that the connection will not support the higher speed. The speech compression is achieved using a code-excited linear prediction (CELP) algorithm that has been enhanced by incorporating fractional-pitch prediction and constrained stochastic excitation. The speech encoder operates at 6.8 kb/s.

The video signal is preprocessed to produce three separate video frames. One frame contains the luminance information with a resolution of 128 pixels by 112 lines. The other two frames contain chrominance information with resolutions of 32 pixels by 28 lines. The frames are each segmented into 16 by 16 blocks for processing. A motion estimator compares the block being processed with blocks from the previous frame, and generates an error signal based on the best match. This signal is converted to the frequency domain using the discrete cosine transform. The transformed signal is quantized, encoded, and sent to the output buffer for transmission. The transmitted frame rate is 10 frames per second using an overlay scan instead of a TV raster scan. If there is a large amount of motion, the frame rate is reduced so that the output buffer does not overflow. A button on the telephone can be used to adjust the frame rate transmitted from the distance set. However, a higher frame rate decreases the resolution of the received image. A self-view mode is also provided to allow the near-end party to view the image that is being transmitted to the far-end party.

Bell and Tone Ringers

In traditional telephones an electromechanical bell ringer is used to alert the customer to an incoming call. The typical ringer has two bells of different pitch that produce a distinctive sound when struck by a clapper driven by a moving-armature motor. The ringer coil inductance in series with a capacitor resonates at 20 Hz to provide a low-impedance path for a 20-Hz ringing signal. The high inductance of the ringer coil prevents loading of the speech network or DTMF generator when the handset is off-hook.

Other ringer connections are used when customers are on party lines and must be rung selectively. Selective ringing schemes include the connection of the ringer between the tip or ring conductors and ground, or ringers tuned to different ringing frequencies (16 to 68 Hz).

Electronic tone ringers are used in most new telephones. A resistor-capacitor circuit is bridged to the telephone line to provide the proper input impedance (defined by FCC rules) for a ring-detect chip. Tone ringers can have equivalent effectiveness and acceptability to the customer when compared to bell ringers if acoustic spectral content and loudness are adequate. Typically, the tone ringer consists of a detector circuit, which distinguishes between valid ringing signals and transients on the telephone line, and a tone generator and amplifier circuit that drives an efficient electroacoustic transducer. The transducer may be a small ordinary loudspeaker, a modified telephone receiver, or a special piezoelectric-driven sounder (see Fig. 17.7.10).

Tone and Pulse Dialers

Dial-pulse signaling interrupts the telephone line current with a series of breaks. The number of breaks in a string represents the number being dialed; one break is a 1 and 10 breaks is a 0. These breaks occur at a nominal rate of 10 pulses per second, with 600 ms between pulse trains. The ratio of the time the line current is broken to the total cycle time (percent break) is nominally 61 percent. Dial pulse signaling can be used with all central offices.

The mechanical rotary dial in the traditional telephone uses a single cam to open and close the dial contacts. The dial is driven by a spring motor that is wound up by the user as each digit is dialed. The return motion is controlled by a speed governor mechanism to maintain the proper dial-pulsing rate.

DTMF signaling consists of sending simultaneously two audio frequencies of at least 50-ms duration representing a single number, separated by at least 45-ms intervals between numbers. On the standard 4-by-3 dial format, each column and each row is associated with a different frequency, as shown in Fig. 17.7.7. This method of signaling permits faster dialing for the user and more efficient use of the switching systems. Since the frequencies are in the audio band, they can be transmitted throughout the telephone network.

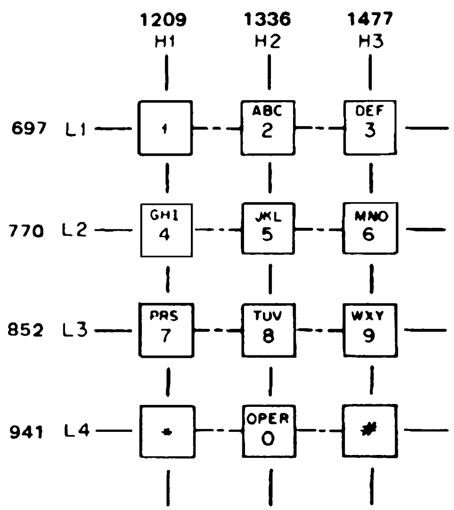

FIGURE 17.7.7 Basic arrangement of pushbuttons for dual-frequency dialing.

Pushbutton dials originally used for DTMF signaling were laid out in a rectangular format to accommodate the cranks and levers necessary to operate the mechanical switches. With modern electronic push-button dials any layout can be used, but the 4-by-3 format is still popular for its dialing speed. Pushbutton dials can perform the dial-pulse function electronically. These electronic "rotary dials" interrupt the line current with transistors or relays. Since the user can enter a number into the dial faster than the number can be pulsed out, a first-in, first-out memory is used to store the number as it is dialed. The dial-pulse timing is generated using an internal clock.

Several methods have been used to generate DTMF signals. Early methods used an inductor-capacitor oscillator with two independently tuned transformers. Different values of inductances are switched into the circuit to obtain different frequencies. Another method is a resistor-capacitor oscillator. Here a twin-tee notch filter in the feedback loop of a high-gain amplifier gives the desired frequency. Two amplifier-filter units are used, one for each frequency group. A modern method to generate DTMF signals uses CMOS integrated circuits to use digit synthesis techniques (Fig. 17.7.8). The keypad information is combined with a master clock to generate the desired frequency. This information is fed to a D/A converter, whose output is a stair-step waveform. The waveform is filtered and fed to a driver circuit, which provides the desired sine-wave frequency signals to the telephone line.

The latest method used to generate DTMF signals is in software. If a digital speech processor is available in the product, a subroutine can be written to generate the appropriate DTMF waveform. The output is a digital word that is periodically fed to a CODEC for conversion to an analog signal. This is fed to a buffer amplifier that drives the telephone line.

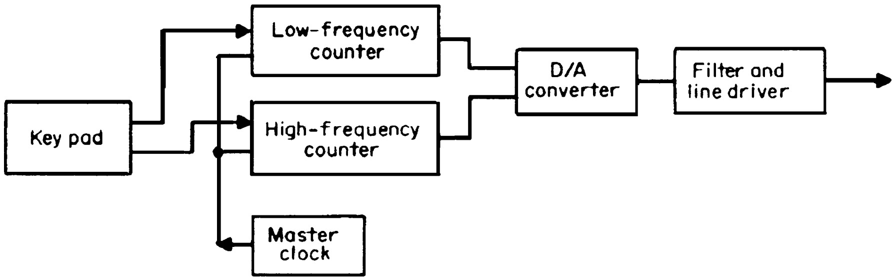

FIGURE 17.7.8 Digital-synthesis circuit.

To combine the versatility of both pulse and DTMF signaling, many telephones have the ability to switch between the two systems. This permits the user to use pulse dialing for making the telephone call and then switch to DTMF for end-to-end signaling for services such as bank-by-phone.

Microphones

The granular carbon microphone, often called a transmitter in telephony, dates back over 100 years to the birth of telephony. Sound striking the diaphragm imparts a pressure fluctuation on the carbon aggregate (Fig. 17.7.9*a*). Since granule contact force and dc resistance R_0 are inversely related, a modulation of the telephone loop current I_0 results. Carbon transmitters offer 20 to 25 dB of inherent signal-power gain and have a nonlinear input/output relationship that advantageously favors speech over low level background noise, but their low resistance consumes loop power. Electronic telephones require a microphone that consumes much less power.

The electret microphone is a small capacitive microphone that is widely used today. It has low sensitivity to mechanical vibrations, a low power requirement, and high reliability. An electret has no inherent gain, so requires a preamplifier. An effective bias voltage V_0 depends on a polymer diaphragm's trapped electret charge (Fig. 17.7.9*b*). The piezoelectric ceramic unimorph element (Fig. 17.7.10) is used as a microphone in cordless and cellular handsets. Its piezoelectric activity owes to an electrically polarized, synthetic (as opposed to natural crystal) ferroelectric ceramic. The flexural neutral axis of the structural composite is not at the ceramic's midplane; thus, vibration results in variation of the ceramic's diameter that induces a voltage across its thickness as defined through the piezoelectric constant d_{31}.

Receivers

The receiver converts the electrical voice signal back into acoustic sound pressure. Either one of the following designs is often used. The electromagnetic (moving-iron) receiver uses voice coil currents to modulate the dc flux,

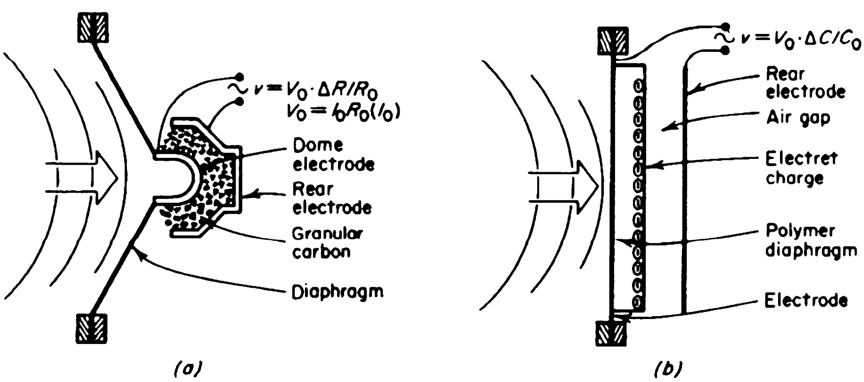

FIGURE 17.7.9 Microphones: (*a*) Carbon and (*b*) electret.

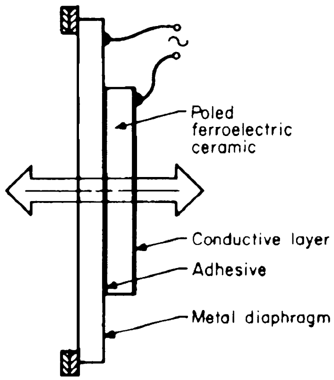

FIGURE 17.7.10 Piezoelectric ceramic unimorph.

which produces variable force on the iron armature (Fig. 17.7.11*a*). The electrodynamic (moving-coil) receiver uses coil current perpendicular to the dc magnetic field to generate an axial force on the movable coil (Fig. 17.7.11*b*). The constant (dc) reluctance of the coil air gap results in less distortion than in the electromagnetic receiver. The moving-coil unit has a lower, more nearly real impedance compared with the moving-iron design. Both of these permanent magnet devices can be used as a microphone, since their input/output is reversible.

Certain hearing aids can inductively couple to the leakage flux from receivers that generate an external magnetic filed. Some receiver types (e.g., piezoelectric) must have an induction coil added to the design to provide an adequate magnetic field. Most telephones are required to provide a magnetic field (see Public Law 100-394, 1988) in order to provide the hearing impaired with access to the telephone network.

Handsets

The handset holds the microphone and receiver. It may also contain a line switch, dial keypad, and other circuits to make it a complete one-piece telephone. The handset positions the transmitter in proper location with respect to the mouth when the receiver is held on the ear. Standard dimensions for the relative mouth and ear locations have been defined (Fig. 17.7.12) for a head that is statistically modal for the population.

The handset should provide an acoustic seal to the ear, provide proper acoustic coupling for its transmitter and receiver, and be heavy enough to operate the telephone switch hook when placed on it. Handsets for hearing impaired users may also contain an amplifier and a volume control.

Protection

The user must be protected against contact with ringing voltages, lightning surges, and test signals applied to the telephone line. Telephone service personnel are trained to work on live telephone lines, but users are not. Lightning surges that are induced onto the telephone might be 1000 V peak but are of limited energy (Fig. 17.7.13). (See Carroll and Miller, 1980). Test signals applied from the central office can be up to 202 V dc.

Telephone cables, often strung on the same poles as power cables, are subject to power crosses if the power cable breaks (as in a storm) and falls on the telephone cables. The telephone company installs

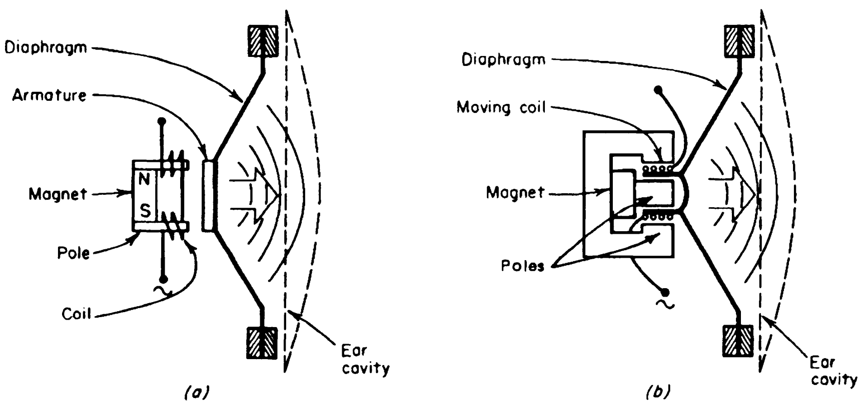

FIGURE 17.7.11 Receivers: (*a*) central armature magnetic (moving armature) and (*b*) dynamic (moving coil).

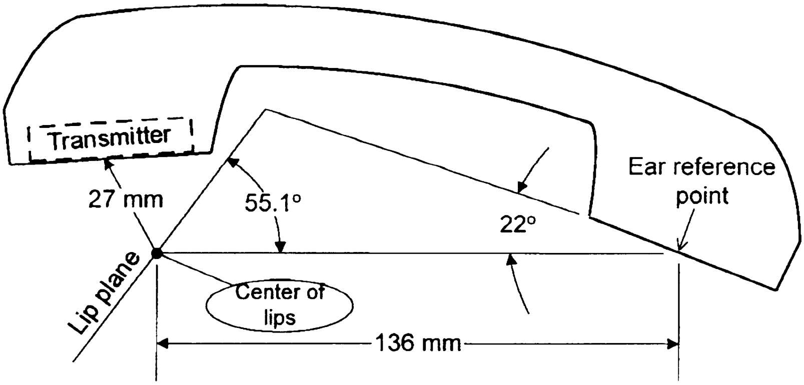

FIGURE 17.7.12 Handset modal dimensions.

primary protectors at building entrances to shunt voltages from power crosses and direct lightning strikes to ground.

Extraneous signals can cause very high acoustic output from the receiver. Either a varistor, V_3 in Fig. 17.7.1, placed directly across the receiver terminals, or the telephone's circuit design is used to limit maximum acoustic output to 125 dBA.

Finally, the telephone network itself needs protection from excessive signal power, fault voltages, and other disturbances caused by the terminal equipment. Requirements for telephone network protection are contained in FCC Rules and Regulations, Part 68.

TELEPHONES WITH ADDED FUNCTIONS

Herbert M. Zydney, R. M. Sachs

Key Telephone Sets

Key telephones are designed for users who need access to more than one central-office (CO) line or PBX extension. In almost all cases, this is accomplished by the addition or illuminated keys (hence the term "key"

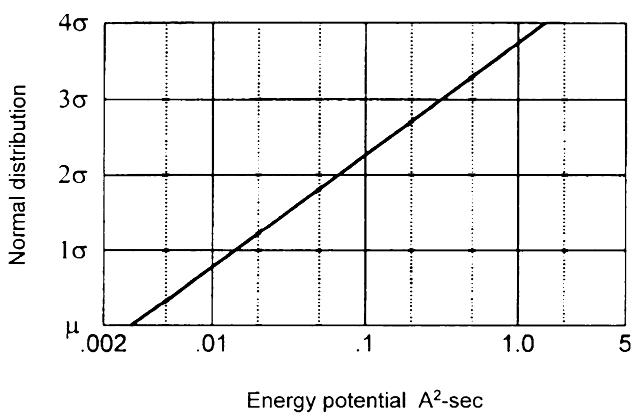

FIGURE 17.7.13 Induced lightning energy distribution.

telephone) to a telephone instrument. These are arranged to correspond to the line or extension and are generally illuminated to identify their status, either directly or by adjacent LEDs or LCDs. The keys can be operated independently to select the desired line. To permit switching between two calls, a hold key is provided so that the active call can be maintained in a "hold" state before operating the key associated with a different call. To allow the user to be alerted, a common audible ringer is provided, which sounds when a new call occurs on any line. Where the telephone is used as a part of a self-contained system, a key is also included so that internal, or "intercom," calls can be made between individual key telephone sets. As technology has evolved, more features have been included in key telephones that improve the efficiency of telephone system operation. Because these newer systems are software controlled, a number of the features are not fully resident in the telephone itself, but depend on a distributed architecture to fully implement them.

Examples include:

(a) *Memory Dialing*: Prestored telephone numbers can be dialed at either the touch of one button or by an abbreviated code from the dial pad.
(b) *Speaker/Microphone Services*: A loudspeaker powered at the station can support hands-free dialing or, when connected to the telephone channel, permit hands-free intercom services or speaker phone operations.
(c) *Display-Based Features*: The most advanced key telephone sets offer a display of either numeric or alphanumeric characters, sometimes augmented by graphic symbols. In conjunction with system software support, these permit users to identify who is calling them, determine the status of other telephones in the same system, and retrieve messages such as the intercom extensions of unanswered calls.

The technology used to implement key telephones presently in use is quite varied. Four categories are worth singling out:

Electromechanical Key Telephone Sets. These early-style telephones generally use electromechanical keys with a relatively large number of electrical contacts for most functions. Individual CO lines and control leads are brought to each telephone, and the actual switching of lines occurs within the telephone set. Additional wires provide control of illumination and ringing. They rely on external line-holding relay circuits and power for their operation, although standard telephone operation is possible on a stand-alone basis. A common hold button activates the external relay holding bridge for all telephones and is released when any other telephone activates its line key. Many key stations are arranged to offer visual and audible signals for status identification using varying rates of interruption for different states. For ease of installation, dedicated wires are usually grouped into 25-pair (50-conductor) cables with a standard connector. Some of the larger telephones, with dozens of line keys, can have four or more such connectors.

Key Telephone Sets with Internal Control. By adding electronic circuitry within the key telephone, many of the functions accomplished by the external relay circuitry can occur internally. Every CO line is brought to each telephone. Internally, ringing signals can be sensed and holding bridges can be applied. Complementary electronics in other telephones can sense the holding bridges and provide the necessary visual signals for multiple line states and access. Because of limitations associated with the power and sensitivity of CO lines, these key sets generally are limited to two, or at the most, four lines. External power, often rectified in the set, is required for the larger systems; smaller systems can operate from the power provided over CO loops.

Electronic Key Telephone Sets. Electronic sets (see Silverio et al., 1985) are different from the first two classes because the CO lines terminate in a centralized switch (often called a key service unit or control unit) rather than in the telephone itself. One, or sometimes two, voice paths are terminated in the telephone itself. A separate digital data link between the telephone and the control unit is used to exchange information, including what keys have been pressed and what visual and audible status information is to be presented to the user. There is little standardization in the functional definition of the conductor pairs. The voice pair generally operates at 600 Ω or 900 Ω, although the actual transmission levels may differ from standard loop levels. The digital data link may be provided on either one or two pairs and can operate from less than 1 to over 200 kb/s. Power is derived in a number of ways: the simplest is to dedicate one pair to power; alternatively, the power may be impressed on a center tap of paired transformers carrying either voice or data and removed and filtered in the telephones; lastly, the digital signals can be shaped so that they have no dc component, and the power may then be sent along with the data. Voltage of +24, +36, and –48 V are commonly used. The basis of operations within the telephone is to scan input stimuli such as keys in real time, format messages, and then send them serially to the key service unit. Messages from the key service unit are demultiplexed and are used either to drive audible ringers or to flash lights. Advanced systems embed a cyclic redundancy check with each message to minimize false operations. Somewhat more complex message structures are involved where displays are to be updated. This approach minimizes the requirement for changes within the telephone to customize it to the user's needs. For example, software in the key service unit is responsible for determining the meaning of most keys on the telephone sets. If it is changed, only the set labels are varied. The circuitry for operating the set is implemented in a number of ways. Small 4- and 8-bit microprocessors are common, although the smallest sets use custom VLSI for their operation. Wiring is reduced to as few as two pairs, although up to four pairs are also used where a second voice path is required. Depending on the speed of transmission, twisted wire pairs without bridge taps are usually required, which makes these sets of limited use in residential locations unless the locations are rewired.

Digital Key Telephones. The most technologically advanced key telephones send and receive voice signals in digitally encoded format. See Murato et al. (1986). The standard network encoding of 64 kb/s in mu-law format is most common. The telephone contains a compatible codec that converts from analog to digital formats. As with the electronic key telephones, many designs have nonstandard interfaces. The most basic digital key telephones combine the digital data stream and the encoded voice stream on two pairs. More advanced telephones add an additional channel for binary data at speeds from 9.2 to 64 kb/s. As an option, or at times built-in, this channel appears at a standard EIA RS 232 interface, which can directly interface with compatible terminals or desktop computers. The key service unit can switch this data channel independently of the voice channel. The conductor formats vary. Some systems use two pairs, each carrying the combined voice and other signals in different directions at speeds up to about 200 kb/s. Other systems use just a single pair at speeds approximating 500 kb/s. These operate by sending information to the telephone preceded by a flag bit; the telephone synchronizes to this data stream and, at its conclusion, replies with a return message. This format is defined as time-compression multiplex in transmission terminology; informally, it is referred to as "ping-pong" because the signals go back and forth constantly. In recent years, international standards bodies have defined a new network standard, ISDN (integrated services digital network). The protocols formulated for this service network are broad enough to embrace the needs of digital key telephones. The implementation costs and flexibility of this new standard have become low enough that these protocols are now appearing in digital key telephone sets.

PC-Emulation of Key Telephone Sets

With the increasing number of desktops that have both telephones and PCs, the two are being integrated in a number of forms. In the simplest arrangement, logical connections are made between the PC and the telephone. Graphics software recreates the telephone on the face of the screen, allowing keyboard commands or mouse-and-click operations to replace buttons. Graphic displays replace the illumination of traditional telephones. A more advanced configuration builds circuitry into the PC, eliminating the requirement for a physical instrument entirely, if the PC has audio capability. User benefits include simpler operation for complex features, built-in help screens, and better integration with data bases. For example, if the PC is able to receive the incoming telephone number of the calling party, the PC screen can automatically display information about the incoming caller.

Speakerphone

A speakerphone is basically the transmission portion of a telephone set in which the handset has been replaced by a microphone and loudspeaker, typically located within a few feet of the user. This arrangement provides the user freedom to move about with hands free during a telephone conversation, as well as some reduction in fatigue on long calls. It also facilitates small-group participation on a telephone call and can be of benefit in special cases involving hearing loss and physical handicaps.

The lengthened transmitting and receiving acoustical paths in the speakerphone arrangement, compared with those of a conventional telephone, introduce loss, which can be on the order of 20 dB or more in each path. This requires that gain be added to both the transmit and receive channels over that provided in a conventional telephone. The amount of gain that can be added in each channel to compensate for the loss in the acoustical paths is limited by two problems. A "singing" problem may occur if an acoustic signal picked up by the microphone is

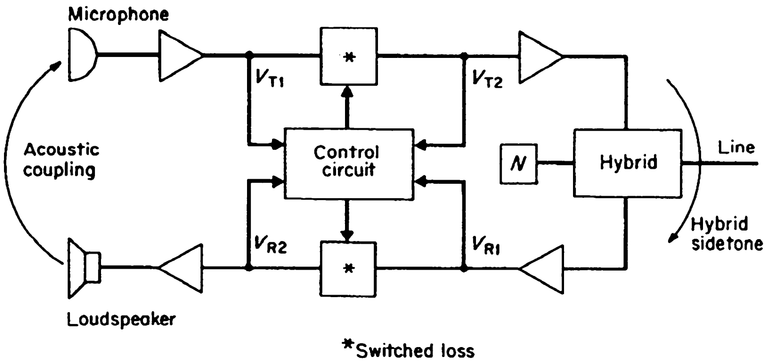

FIGURE 17.7.14 Block diagram of a voice-switched speakerphone. (*Copyright American Telephone and Telegraph Co. Used by permission*)

fed back to the loudspeaker via the sidetone path and returns to the microphone through acoustic coupling in the room. Singing can occur when too much gain is added in this loop. Even before reaching this condition, however, the return of room echoes to the distant talker can become highly objectionable. Echoes occur when coupling from loudspeaker to microphone causes the incoming speech to be returned to the distant party with delay.

A solution to these problems can be found in *voice switching*, in which only one direction of transmission is fully active at a time. With voice switching, a switched-loss or switched-gain element is provided in both the transmit and receive channels, which operate in a complementary fashion. In this manner, full gain is realized in the chosen direction of transmission, while margin is provided against singing and distant talker echo. Voice switching, however, results in one-way-at-a-time communication. Also, there can be a problem of clipping of a portion of the speech, since control of the voice-switching operation is derived from the speech energy itself. A functional diagram of the essential elements for a voice-switched speakerphone is shown in Fig. 17.7.14, in which a measure of the speech energy is provided to the control circuit from four distinct locations in the transmission paths. Signals V_{TI} and V_{RI} that are a measure of the speech energy in the transmit and receive paths, respectively, are compared to determine the direction of transmission to be enabled. Signal V_{TZ} is used by the control circuit to guard against switching falsely into receive because of transmit energy arriving in the receive path through the hybrid sidetone circuit. Similarly, the control circuit uses V_{R2} to guard against switching falsely into transmit because of receive energy arriving in the transmit path from acoustic coupling. Many speakerphone designs do not use all four control signals directly, but equivalent functions are generally provided by other means.

Another solution to echo control and reduction that enables full duplex performance of speaker-phones is acoustic echo cancellation (AEC). With this technique, the signal driving the speakerphone's loudspeaker is compared to the signal generated by the microphone. The speakerphone builds an acoustic model to remove the acoustic echo from the microphone signal before it is transmitted over the telephone network. The use of AEC in speakerphones has been enabled by the availability of lower cost, high function digital signal processors (DSPs), and advances in adaptive signal processing needed to handle dynamic changes of the acoustic environment as people move about and room conditions change. Once the AEC is adapted, the need for switched loss is diminished and the speakerphone can attain a full-open, or full duplex, condition. This allows for very natural and fluid voice communications.

An AEC is typically placed between the control circuits of a speakerphone and the signals to and from its transducers. This effectively eliminates echoes before they pass the control circuit. Typically there is a signaling connection between the AEC and the control circuit to communicate the status of the AEC. This allows the control circuit to switch less loss when the AEC has adapted, and to prevent the AEC from adapting during periods of double-talking. Hybrid echo cancellers (see Sondhi and Presti, 1966) is typically placed between the hybrid and the control circuit of the speakerphone.

While voice switching and acoustic echo cancellation is effective in eliminating the problems of singing and distant-talker echo, it does not relieve the higher transmitted levels of room ambient noise and reverberant (barrel effect) speech caused by increased gain in the transmit channel. Both of these problems can be reduced by one or more of the following methods: (1) install the speakerphone in a quiet, nonreverberant room; (2) reduce the transmit gain and the actual talker-to-microphone distance; (3) reduce the transmit gain and the effective talker-to-microphone distance with the use of a directional microphone; and (4) roll off the low-frequency end of the transmit response slightly to reduce the pickup of low-frequency noise and reverberant room modes, at the expense of a slight reduction in voice naturalness.

DATA TERMINALS

W. J. Lawless

Data Terminal Equipment (DTE)

Data terminals are used in a variety of data communication systems. Terminals can be classified into two categories: general-purpose terminals and special-purpose terminals. General-purpose terminals contain many elements found in a modern day personal computer (PC) and thus it is very common to use a PC to emulate a data terminal. In some applications, where cost is particularly important, a teletypewriter data terminal containing only the basic terminal elements is used as a general-purpose data terminal.

Special-purpose data terminals have been designed for a number of applications. In some cases the functionality is limited as in the case of a send-only or a receive-only terminal. Likewise, special purpose terminals are used in applications requiring special functionality such as in handheld applications and point-of-sale applications.

In either case the data terminal is used to communicate messages. Some common applications include inquiry-response, data collection, record update, remote batch, and message switching.

Data terminals are typically made up of combinations of the following modules: keyboard, CRT display, printer, storage device, and controller. These modules can be organized into the categories of operator *input-output*, terminal *control*, and *storage.*

Keyboards are available in a number of formats. In some cases, two formats are provided on the same keyboard, side by side or integrated, e.g., typewriter with a numeric pad. Future designs call for multifunction keyboards whose designations or functions can be easily changed by the user, because in many cases differently trained operators will be using the same terminal to enter different types of data for different applications. There is also a trend toward special-application keyboards, such as the cash register keyboard used by a fast food chain in which the name of each item on the menu appears on a proximity-switch plastic overlay.

CRT displays allow the operator to enter, view, and edit information; editing information with a CRT display is typically much faster than if a mechanical printer were used. The most popular CRT displays exhibit 24 lines of up to 80 characters each. Other size variations include 12 lines of up to 80 characters and a full page display (approximately 66 lines of up to 80 characters). Other features found on CRT displays include blinking, half-intensity, upper- and lowercase character sets, foreign-character sets, variable character sizes, graphic (line-drawing) character sets, and multicolor displays.

Printers for data terminals generally are classified into three types—dot-matrix, inkjet, and laser printers. Dot-matrix printers are lowest in cost but are also lowest in print speed and print quality. Laser printers are highest in cost but provide the highest print speed and quality. Inkjet devices are intermediate in cost, speed, and quality.

Storage devices for data terminals include electronic (RAM, ROM), magnetic (floppy disk, hard disk), and optical (CD-ROM). Optical storage is emerging as the most popular medium for storage and retrieval of large amounts of information, particularly in applications such as online encyclopedias, image retrieval, and large databases.

Controllers interconnect the channel interface and the various terminal components and make them interact to perform the terminals' specific functions. They also perform other functions, such as recognizing and acting on protocol commands, e.g., polling and selecting sequences in selective calling systems; code translation; and error detection and correction. Controller designs typically use microprocessors to increase their functions, including programmability, which greatly increases terminal versatility.

Standardized codes, protocols, and interfaces provide a uniform framework for the transmission and reception of data by data terminals. Codes and some character-oriented protocols and interfaces are discussed in Chap. 1.

IBM's binary synchronous communication (bisynch) protocol has been implemented widely. It accommodates half-duplex transmission and is character-oriented. Bit-oriented protocols that provide both half- and full-duplex transmission are also in widespread use. The American National Standards Institute's (ANSI) Advanced Data Communications Control Procedures (ADCCP), the International Organization for Standardization's High Level Data Link Control (HDLC), and Synchronous Data Link Control (SDLC) are the current standards for bit-oriented protocols. Although there are differences, they are generally compatible, and the trend is for other bit-oriented protocols to establish compatibility with them.

Display Terminals (Text and Graphics)

Terminals for the entry and electronic display of textual and or graphic information generally are used for communicating between people and computers and, increasingly, between people. Typical application areas include computer program preparation, data entry, inquiry-response, inventory control, word processing, financial transactions, computer-aided design, and electronic mail. A terminal is divided functionally into control, display, user input, and communication-line-interface portions (see Fig. 17.7.15). The first three portions are detailed in the following sections. Interface to a communication line is typically through either an analog or digital modem.

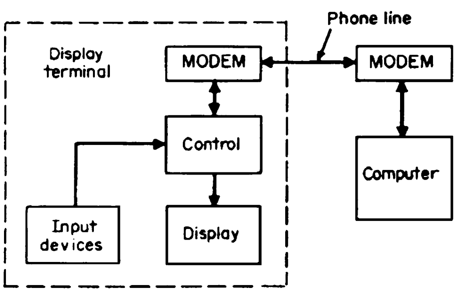

FIGURE 17.7.15 Display-terminal organization.

Terminals have differing amounts of control or decision-making logic built into them. A popular classification scheme is based on terminal control capabilities. A nonintelligent ("dumb") terminal is a basic I/O and control device. Although it may have some information buffering, it must rely on the computer to which it is connected for most processing of any external information and editing of output displays. A "smart" terminal has both information entry and editing capabilities, and although it too is generally connected to a computer, it can perform information processing locally. The terminal usually contains a microcomputer that is programmed by the terminal manufacturer to meet the general and special needs of a user. An "intelligent" terminal has a microcomputer that can be programmed by a user to meet user-specific needs (as mentioned earlier, it is very common to use a PC as a smart or intelligent terminal). Terminals can be connected to a supporting computer either directly or through a cluster controller that supervises and services the data-communication needs of a number of terminals.

The most common terminal display device is a cathode-ray tube (CRT) similar to that used in a home TV receiver. In a home TV receiver, the refreshing is done by the broadcast station, which sends 30 complete images per second. In a terminal, the refreshing must be provided by the control logic from information stored electronically in the terminal. This storage may be in a separate electronic memory or memory that is part of the microcomputer. In either case the microcomputer can address, enter, or change the information.

Flat panel displays are an alternative to the CRT. Since these devices require much less space and power, they are ideal for laptop and notebook terminals and computers. The most common technology is liquid-crystal displays (LCD). For LCD, liquid-crystal molecules in most displays align themselves parallel to an electric field and lie flat when no field is present. In another type of display, the crystals tilt in different directions in response to a field. Depending on the device construction and the type of liquid crystals used, some of those orientations allow light to pass; others block it out. The result is either a dark image on a light background or the reverse.

Most display terminals have a typewriterlike keyboard and a few extra buttons as input devices for the entry of textual and control information. Text is entered at a position on the display indicated by a special symbol called a *cursor*. The control moves this cursor along much as a typewriter head moves along as text is entered via the keyboard. Different manufacturers use different cursors (underlines, blinking squares, video-inverted characters).

To allow editing of existing images or flexible entry of additions, auxiliary control devices are provided to change the cursor position. The simplest is a set of five buttons, four of which move the cursor up, down, right, or left one character position from the current position. The fifth indicates that the cursor is to be "homed" to the upper left-hand corner of the image, where textual entry usually starts. Another popular position-controlling device is a joystick, a lever mounted on gimbals. Movement of the top of the lever is measured by potentiometers attached to the gimbals. The control senses the potentiometer outputs and moves the cursor correspondingly.

Another form of control for the cursor is the mouse. Movement of the mouse on the desktop provides a direct relationship with the movement of the cursor on the screen. A precision mouse will typically use either a magnetic or optical tablet that maps the position of the mouse on the tablet to a particular spot on the screen. A less-accurate form of the mouse uses friction between the desk surface and a ball contact on the bottom of the mouse to indicate the *relative* motion of the cursor. This form of mouse typically uses two optically sensed shafts mounted perpendicular to each other to detect movement of the ball contact. A mouse can have one to three buttons on top.

Additional input methods include track balls, pressure-sensitive pads, and even eye-position trackers. These find utility in specific areas such as artwork generation, computer image processing, industrial applications, training, and games.

Terminals intended for display of complex graphic information, e.g., for computer-aided design, generally have either tablet-stylus devices, light pens, or mice for input of new information or indication of existing information. One popular tablet-stylus has a surface area under which magneto-strictive waves are alternately propagated horizontally and vertically. A stylus with a coil pickup in the tip senses the passing of a wave under the stylus position. Electronic circuitry measures the time between launching of a wave and its sensing by the stylus and computes the position of the stylus from that time and the known velocity of the wave.

A light pen senses when light is within its field of view. In a raster-scan display, the time from the start of a displayed image until the light-pen signal is received indicates where the pen is in the raster-scan pattern and, with suitable scaling factors, gives the position of the pen over the image. In a directed-beam display, pen-position locating is more complicated. The centering of a special tracking pattern under the pen is sensed, and through a feedback arrangement controlled by the terminal computer the pattern is moved to keep it centered. The position of the pattern is then the same as the pen position.

Multimedia terminals are now becoming more commonplace. These terminals combine text, image, and voice capabilities. In the office environment, tighter coupling between data, e-mail, fax, and voice will take place. Similarly, with more of the workforce working at home, either full-time or part-time, multimedia services in the home will be required.

Data Transmission on Analog Circuits

The devices used for DTE generate digital signals. These signals are not compatible with the voice circuits of the public telephone network, partly because of frequency range but more importantly because these circuits are likely to produce a small frequency offset in the transmitted signal. The offset causes drastic changes in the received waveform. This effect, while quite tolerable in voice communications, destroys the integrity of individual pulses of a baseband signal. Compatible transmission is obtained by modulating a carrier frequency within the channel passband by the baseband signal in a modem (modulator-demodulator).

For a band-limited system, Nyquist showed that the maximum rate for sending noninterfering pulses is two pulses (usually called symbols) per second per hertz of bandwidth. The bit rate depends on how these pulses are encoded. For example, a two-level system transmits 1 b with each pulse. A four-level system transmits 2 b per pulse, an eight-level system transmits 3 b, and so forth. Unfortunately, we cannot go to an arbitrarily large number of levels because, assuming that the total power is limited, the levels will become so closely spaced that the random disturbances in the transmission medium will make one level indistinguishable from the next. Shannon's fundamental result states that there is a maximum rate, called the *channel capacity,* up to which one can send information reliably over a given channel. This capacity is determined by the random disturbances on the channel. If these random disturbances can be characterized as white Gaussian noise, the channel capacity C is given by

$$C = W \log_2 (1 + S/N)$$

where W is the bandwidth of the channel and S and N are the average signal and noise powers, respectively.

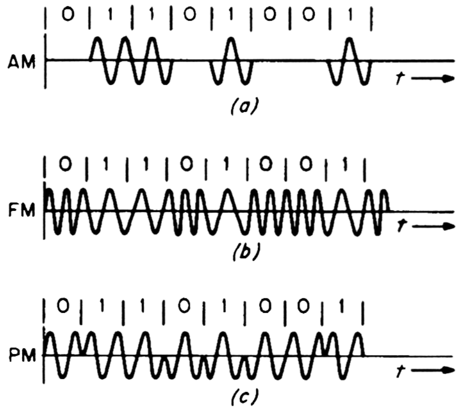

FIGURE 17.7.16 Binary (*a*) amplitude-, (*b*) frequency-, and (*c*) phase-modulated carrier waves.

With today's more elaborate commercial modulation and coding techniques it is possible to transmit data at approximately 70 percent of the capacity of the channel. With the most complex modulation and coding techniques, it is possible to get very close to the capacity of the channel, but this is achieved at the expense of considerable processing complexity and delay.

Data transmission employs all three modulation methods (amplitude, frequency, and phase) plus combinations of them. A description of these methods is given in Sec. 12.

An on-off amplitude-modulated (AM) signal is shown in Fig. 17.7.16*a*. This modulation scheme is little used for voice-band modems because of its inferior performance compared to other modulation schemes.

An example of a binary frequency-modulated (FM) carrier wave, sometimes called frequency-shift keying (FSK), is shown in Fig. 17.7.16*b*). While FM or FSK requires somewhat greater bandwidth than AM for the same symbol rate, it gives much better performance in the presence of impulse noise and gain change. It is used extensively in low- and medium-speed (voice-band) telegraph and data systems.

A binary phase-modulated (PM) carrier wave is shown in Fig. 17.7.16*c*, where a phase change of 180° is depicted. However, this modulation method is usually employed in either four-phase or eight-phase systems. In a four-phase-change system, the binary bits are formed in pairs, called *dibits*. The dibits determine the phase change from one signal element to the next. The four phases are spaced 90° apart. In effect, this method employs a four-state signal, and such a system is inherently capable of greater transmission speeds for the same bandwidth, as is obvious from the Nyquist-Shannon criteria stated above. With improvement of voice channels, the four-phase (quaternary-phase-modulated) scheme is being employed increasingly for medium-speed (voice-band) data transmission systems to give higher transmission speeds than FM for the same bandwidth. The system is useful only in synchronous transmission.

In present-day modems FSK is the preferred modulation technique for bit rates below 1800 b/s. At 2400 b/s, the commonly used modulation technique is PSK using four phases, and at 4800 b/s it is PSK using eight phases. The latter requires the use of an adaptive equalizer, an automatically adjustable filter that minimizes the distortion of the transmitted pulses resulting from the imperfect amplitude and phase characteristics of the channel. At 9600 b/s, the preferred modulation technique is quadrature amplitude modulation, a combination of amplitude and phase modulation.

Above 9600 b/s (e.g., 19,200 and 28,800 b/s) the preferred modulation technique is quadrature amplitude modulation combined with a coding technique called trellis-coded modulation (TCM). TCM provides much improved performance at the expense of more complex implementation. CCITT Recommendation V.34 specifies this type of modulation for 19.2 kb/s.

Data Transmission on Digital Circuits

The techniques outlined in the previous section apply for transmission over the analog telephone network, sometimes referred to as public switched telephone network (PSTN). While local access to PSTN is over analog facilities (typically, twisted copper wires), the interexchange and long-distance telephone network consists principally of digital facilities. Analog signals are converted via pulse-code-modulation into digital signals for transmission over these digital facilities. If digital access to the long-distance digital facilities is provided, then end-to-end digital transmission is possible. Data signals then need not go through the digital to analog conversion outlined in the previous section, but can remain digital end-to-end (from source to destination).

An early example of an end-to-end digital system was AT&T's Digital Data System, which was deployed during the mid-1970s. Here, digital access lines together with Central Office digital multiplexers enabled transmission at rates of 2.4, 4.8, 9.6, and 56 kb/s.

Switched network digital systems are provided today via ISDN, which provide circuits at 64 and 128 kb/s.

Higher bandwidth circuits and services are now being offered by various entities. Modern systems use packet-switching techniques rather than circuit-switching to enable very efficient use of the digital transmission facilities. These packet-switched systems typically use frame relay and asynchronous transfer mode (ATM) protocols.

In addition to the advances in the long-distance marketplace, changes are also taking place in the local access area. Regional Bell operating companies as well as alternate access providers are beginning to deploy local distribution systems using optical fiber and coax cable combinations. While the initial thrust for these systems is to provide multichannel video to the home, the broadband facilities also provide an excellent vehicle for broadband multimedia data services to the home.

Local Area Networks

As with wide area networks, a local network is a communication network that interconnects a variety of devices and provides a means for information exchange among those devices. See Stallings and Van Slyke (1994). There are several key distinctions between local networks and wide area networks:

1. The scope of the local network is small, typically a single building or a cluster of buildings. This difference in geographic scope leads to different technical solutions.
2. It is usually the case that the local network is owned by the same organization that owns the attached devices.
3. The internal data rates of local networks are much greater than those of wide area networks.

The key elements of a local area network are the following:

- Topology: bus or ring
- Transmission medium: twisted pair, coaxial cable, or optical fiber

TABLE 17.7.1 LAN Technology Elements

Element	Options	Restrictions	Comments
Topology	Bus	Not with optical fiber	No active elements
	Ring	Not CSMA/CS or broadband	Supports fiber, high availability with star wiring
Transmission medium	Unshielded twisted pair	—	Inexpensive; prewired; noise vulnerability
	Shielded twisted pair	—	Relatively inexpensive
	Baseband coaxial cable	—	—
	Broadband coaxial cable	Not with ring	High capacity; multiple channels; rugged
	Optical fiber	Not with bus	Very high capacity; security
Layout	Linear	—	Minimal cable
	Star	Best limited to twisted pair	Ease of wiring; availability
Medium access control	CSMA/CD	Bus, not good for broadband or optical fiber	Simple
	Token passing	Bus or ring, best for broadband	High throughout, deterministic

Source: Stallings, W. and Van Slyke, R. "Business Data Communications," Macmillan College Publishing Company, 1994.

- Layout: linear or star
- Medium access control: CSMA/CD or token passing

Together, these elements determine not only the cost and capability of the LAN but also the type of data that may be transmitted, the speed and efficiency of communications, and even the kinds of applications that can be supported. Table 17.7.1 provides an overview of these elements.

Data Communication Equipment (DCE) Trends

Most modern businesses of significant size depend heavily on their data-communications networks. Large businesses usually have a staff of specialists whose job it is to manage the network. To assist in this network-management function, DCE manufacturers have provided various testing capabilities in their products. Sophisticated DCEs now are capable of automatically monitoring their own "health" and reporting it to the centralized location where the network management staff resides. These new capabilities are frequently implemented through the use of microprocessors. Some DCEs can also establish whether a trouble is in the modem itself or the interconnecting channel. If it is the channel that is in trouble, equipment is available that automatically sets up dialed connections to be used as backup for the original channel. Another capability is to send in a trouble report automatically when a malfunction is detected.