CHAPTER 1

FUNDAMENTALS OF THE TECHNOLOGY: CONCEPTS AND DEFINITIONS

What is the transmitter? It is an electrical ear which receives the shock of the dancing molecules, just as does the membrane of the human ear. . . . What is the receiver? It is an electric mouth which can utter human sounds.

John Mills, *The Magic of Communication: A Tell-You-How Story*, Information Department, American Telephone and Telegraph Company

Telecommunications is the transfer of information (*communications*) from a transmitter or sender to a receiver across a distance (*tele*). Some form of electromagnetic energy is employed to represent the data, usually through a physical medium, such as a copper wire or a glass fiber. A wireless medium, such as radio or infrared light, also may be employed. Additionally, a number of intermediate devices are typically involved in setting up a path for the information transfer and for maintaining adequate signal strength.

The information transfer must be established and maintained at acceptable levels in terms of certain key criteria such as speed of connection, speed of information transfer, speed of response, freedom from error, and, finally, cost. The information can be voice, data, video, image, or some combination of these—in other words, multimedia. The information can retain its original, or native, form during transmission. Alternatively, the transmission process can alter the data in some way in order to effect compatibility between the transmit and receive devices and with various intermediate network elements. For example, analog voice often is converted into a digital (data) bit stream for transmission over a digital network and is restored to analog form for the benefit of the analog-oriented human being on the receiving end. Additionally, the information can be compressed in order to improve the efficiency of information transfer and can even be encrypted for purposes of security.

Telecommunications and Data Communications Handbook, By Ray Horak

The electromagnetic energy employed to carry the data can be in the form of electric impulses or radiated energy in the form of either radio waves or light rays. The media employed can include metallic conductors (e.g., twisted pair or coaxial cable); free space, or airwaves (e.g., radio technologies such as microwave, satellite, or cellular or optical technologies such as free space optics); and glass or plastic fiber (fiber-optic cable). In a network of substantial size that spans a significant distance, a combination of transmission media typically is involved in the information transfer between transmitter and receiver. An intercontinental voice or data call might involve a combination of many media.

Additionally, a wide variety of intermediate devices might be employed to establish and maintain the connection and to support the information transfer. Such devices may include an appropriate combination of modems or codecs, controllers, concentrators, multiplexers, bridges, switches, routers, gateways, and so on.

This chapter examines a number of concepts and defines a fundamental set of elements that apply universally to communications networks. Distinctions are drawn between dedicated, switched, and virtual circuits, with two-wire and four-wire circuits defined and illustrated. The concept of bandwidth is explored in both analog and digital terms, with the advantages and disadvantages of each explained. The concept of multiplexers is discussed in detail, with variations on the theme detailed and illustrated. Finally, this chapter briefly explores the nature and evolution of various types of switches, including circuit, packet, frame, cell, and photonic switches.

1.1 FUNDAMENTAL DEFINITIONS

Developing a solid understanding of communications networking requires that one grasp a number of fundamental definitions, for telecommunications has a language of its own. The following terms, many of which are illustrated in Figure 1.1, are significant and are applied fairly universally across all voice, data, video, and other systems and network technologies. Some of the terms have multiple definitions that can be specific to a technology or application. As this book will use and illustrate these terms many times across a wide variety of technologies and applications, they soon will become part of your everyday vocabulary. (*Note:* This would be an excellent time to pause and warn your family and friends.)

- **Transmitter:** The *transmitter*, also known as the *sender* or *source*, is the device that originates the information transfer. Transmitters include voice telephones, data terminals, host computer systems, and video cameras.
- **Receiver:** The *receiver*, also known as the *sink*, is the *target* device, or *destination* device, that receives the information transfer. Receivers can include telephones, data terminals, host computers, and video monitors. Note that most devices are capable of both transmitter and receiver functions; exceptions include broadcast radio and TV devices.
- **Circuit:** A *circuit* is a communications path, over an established medium, between two or more points, from end to end, between transmitter and receiver. Circuit generally implies a *logical connection* over a *physical line*. Further, the

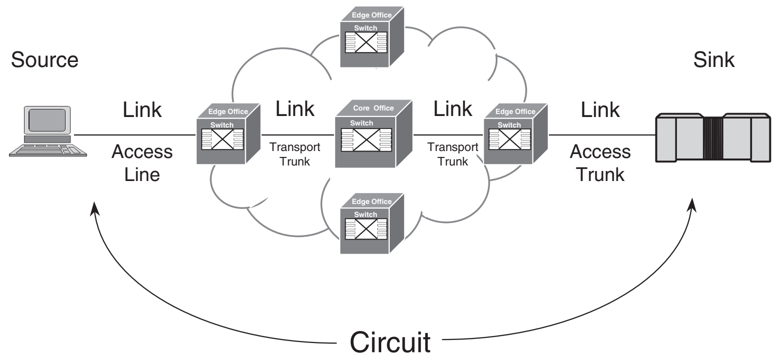

Figure 1.1 Simple circuit between transmitter and receiver across a network involving multiple links and switches.

term circuit often is used interchangeably with *path, link, line*, and *channel*, although such usage can be specific to the underlying technology, the overall context, and other factors. Circuits comprising copper twisted wire are either *two-wire* or *four-wire*, depending on the requirements of the specific application and the fundamental nature of the network. Circuits also may be for purposes of either access or transport. *Access* circuits are from the customer premises to the *edge* of the carrier network, while *transport* circuits are employed in the *core*, or *backbone*, of the network for purposes of long-haul transmission. Circuits may be *simplex* (one-way), *half-duplex* (two-way, but only one way at a time), or *full-duplex* (simultaneous two-way).

- **Link:** A *link* is a two-point segment of an end-to-end circuit (e.g., from terminal to switch or from switch to switch). Typically, a circuit comprises multiple links. Also, a circuit may consist of a single link, as often is the case between a host computer and a peripheral, such as a printer. Link sometimes is used interchangeably with line or circuit.
- **Line:** *Line* has several definitions, which may result in some confusion. In a Private Branch eXchange (PBX) environment, a *station line* refers to the connection between the PBX switch and the station user's terminal equipment, which usually is in the form of telephone, although it could be a computer workstation, a printer, a facsimile machine, or some other device. In rate and tariff terminology, line refers to a *local loop* connection from the telephone company Central Office (CO) switch to the user premises in support of Customer Premises Equipment (CPE) other than a switch. For example, such CPE may be in the form of a single-line residence or business set, a multiline set, or the common control unit of a key telephone system. In any case, line refers to a *voice-grade circuit*, in other words, a circuit serving a single physical location where it terminates in a relatively unsophisticated device. Further, a line has a single associated telephone number and generally is single channel in nature (i.e., supports a single transmission at a time). A line may be thought of as a tributary of a *trunk*. In telephone company (telco) parlance, line describes the user side or local loop side of the central office switch; in other words, the *line side* is the side of the network to which users connect to *access* the network.

The *trunk side* involves the high-capacity trunks that serve to interconnect the various telco switching centers in the core of the carrier network.

- **Trunk:** *Trunk* comes from the Latin *truncus*, meaning *torso*. The trunk is the main body apart from the head or appendages, much as the main channel of a river is apart from its tributaries. In the context of telecommunications, a trunk is a communications circuit, available to share among multiple users, on a pooled basis and with contention for trunk access managed by an intelligent switching device. Therefore, trunks interconnect *switches*. For example, *tie trunks* connect PBX switches in a private, leased-line network, *central office exchange trunks* connect PBXs to telephone company central office exchange switches, and *interoffice trunks* interconnect central office exchange switches. *Trunk groups* are groups of trunks serving the same special purpose, with examples including Direct Inward Dial (DID) and tie trunk groups. Trunks are directional in nature, with the options being one-way outgoing (originating), one-way incoming (terminating), or two-way (combination).
- **Channel:** In formal standards terms, a *channel* is a means of one-way connection between transmitter and receiver—therefore, a one-way *circuit* or signal *path*. In data processing terminology, particularly IBM, a channel is a high-speed two-way connection between mainframe and peripheral. In common usage, a channel is a *logical* connection over a *physical* circuit to support a single conversation. You can configure a physical circuit in such a way as to support one or many logical conversations. Multichannel circuits always are four-wire in nature—either physical or logical four-wire.
- **Switch:** A *switch* is a device that establishes, maintains, and changes logical connections over physical circuits. Common examples of switches include PBXs and Central Office Exchanges (COs or COEs), both of which are *circuit switches*. Circuit switches establish connections between circuits (or links) on demand and as available. While developed to support voice communications, circuit switches can support any form of information transfer (e.g., data and video communications). *Packet switch* is a generic term that actually includes packet, frame, and cell switches. Packet switching evolved in more sophisticated networks, primarily in support of computer-to-computer data and image transfer. In terms of physical placement, there are *edge switches* and *core switches*. Edge switches are positioned at the physical edge of a network; the user organization gains access to an edge switch via an access link. Core switches, also known as *tandem switches* and *backbone switches*, are high-capacity switches positioned in the physical core, or backbone, of a network and serving to interconnect edge switches. Although some switches are very intelligent in many respects, a pure switch makes connection decisions only at the link level. That is to say that a switch has a very limited view and cannot consider the network as a whole. Therefore, switches operate link by link, that is, hop by hop, generally under the control of a centralized set of logic that can coordinate their activities in order to establish end-to-end connectivity across a multilink circuit.
- **Router:** A *router* is a highly intelligent switch capable of making traffic routing decisions based on a view of the network as a whole. This is in contrast to simple switches, which see only an individual link and have no sense of the larger network. Routers are programmable devices that can be quite sophisticated. In

determining the route for a given communication, a router can be programmed to consider a number of factors including the addresses of the originating and destination devices, the least-cost route, the least-congested route, and the shortest route. Routers can be capable of connecting dissimilar networks, such as circuit-switched and packet networks, and accomplishing the conversion processes necessary to resolve any issues of incompatibility. Chapter 8 discusses routers in great detail.

- **Network:** A *network* is a fabric of elements that work together much as the fabric of a net to support the transfer of information. In the extreme sense, a network includes everything from the transmitters to the receivers, including all links, switches, and other intermediate devices that can be called upon to support a communication.
  - **Local Area Network (LAN):** A LAN is a local, that is, limited-distance, packet network designed for interconnecting computers, peripherals, storage devices, and other computing resources within a confined area. A LAN may serve an office, a single floor, an entire building, or perhaps a campus of many buildings but generally does not cross a public right-of-way. LANs generally are private networks. LANs can be interconnected, perhaps across a MAN or WAN.
  - **Metropolitan Area Network (MAN):** A MAN is a public network that serves a metropolitan area or perhaps a portion of a metropolitan area such as a city or a suburb. MANs tend to be data oriented and increasingly serve to interconnect LANs. A number of carriers now offer high speed metropolitan Ethernet services, for example.
  - **Wide Area Network (WAN):** A WAN is a network that covers a wide geographic area such as a state, province, region, or country. The *Public Switched Telephone Network* (PSTN) is a voice-oriented WAN that individuals use to connect voice calls. The Internet is a WAN, as are many other data-oriented public networks. WANs can serve to interconnect LANs and MANs. The WAN commonly is depicted as a *cloud* (Figure 1.1), which originated in sales presentations of the 1970s for data communications networks. The thought behind the cloud simply was that the specific internal workings of the networks could be many and various, change from time to time, and vary from place to place. The cloud served to obscure those internal workings from view. The cloud was the consummate conceptual sale—data simply popped in on one end of the network in one format and popped out on the other side of the network in another format. Interestingly, the data network that gave rise to the cloud never worked, but the cloud lives on. Actually, the cloud is entirely appropriate for depicting the Internet, as the specific internal workings are largely unpredictable for any given call.

1.2 DEDICATED, SWITCHED, AND VIRTUAL CIRCUITS

Circuits can be provisioned on a dedicated, switched, or virtual basis, depending on the nature of the application and the requirements of the user organization. Ultimately, issues of availability and cost-effectiveness determine the specific selection.

1.2.1 Dedicated Circuits

Dedicated circuits are distinct physical circuits dedicated to directly connecting devices (e.g., PBXs and host computers) across a network (Figure 1.2). Dedicated circuits make use of access circuits, in the form of local loops, to access the service provider's *Point of Presence* (POP) at the edge of the carrier network. Rather than accessing a switch at the POP, a dedicated circuit terminates in the carrier's *wire center*, where cross connections are made from the short-haul access circuit to a long-haul transport circuit. Dedicated circuits serve a single-user organization only, rather than serving multiple users. They offer users the advantage of a high degree of availability as well as specified levels of capacity and quality. You can *condition* dedicated circuits to deliver specific levels of performance by adding amplification or other signal processing enhancements to the line to optimize its transmission characteristics, whereas you generally cannot do so with switched circuits, at least not from end to end. (*Note:* The local loop links used to access switched networks are dedicated circuits and therefore can be conditioned, but the links within the network cloud vary from call to call and therefore cannot be conditioned.) Additionally, the costs of dedicated circuits generally are calculated on a flat-rate, rather than a usage-sensitive, basis. That is to say you can use them continuously and to their full capacity for the same cost as if you never use them at all.

However, the reservation of a circuit for a specific customer has a negative effect on the network efficiency because that circuit is taken out of shared public use and, therefore, is unavailable for use in support of the traffic of other users. As a result, dedicated circuits tend to be rather expensive, with their costs being sensitive to distance and capacity. Additionally, the process of determining the correct number, capacity, and points of termination of such circuits can be a difficult and lengthy design and configuration process. Further, long lead times often are required for the carrier to configure (i.e., provision) or reconfigure such a circuit. Finally, as dedicated circuits are susceptible to disruption, backup circuits often are required to ensure effective communications in the event of either a catastrophic failure or serious performance degradation.

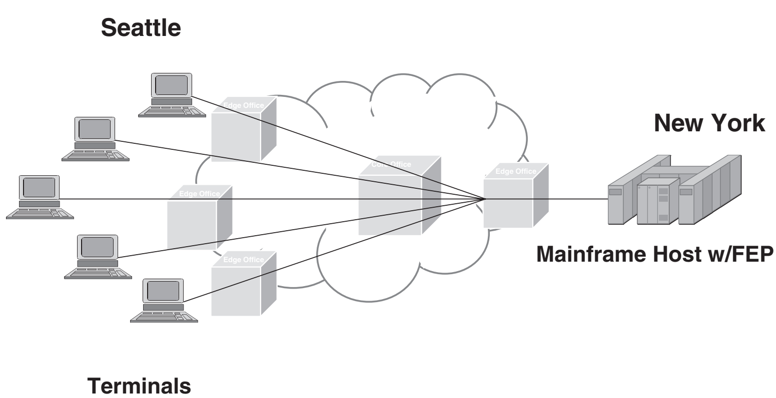

Figure 1.2 Dedicated circuits between Seattle data terminals and a New York mainframe through a Front-End Processor (FEP).

Traditionally, dedicated digital circuits have connected large data centers that communicate intensively. Similarly, many large end-user organizations with multiple locations have used dedicated circuits known as tie trunks to tie together multiple PBXs. In both cases, the advantages of assured availability, capacity, and quality in support of mission-critical, time-sensitive applications often outweigh considerations of configuration difficulty and risk of circuit failure. Dedicated circuits often are known as *nailed-up* circuits because, in days long past, the twisted-pair copper physical circuits were hung from nails driven in the walls of the carrier's wire centers.

1.2.2 Switched Circuits

Switched circuits are connected through the network on a flexible basis through one or more intermediate switching devices. Traditionally, the switches were in the form of the telephone company Central Office exchanges, as illustrated in Figure 1.3. Individual users seeking switched connections through the network connect to the edge switches via dedicated local loops, that is, access circuits, terminating at the premises. Through those local loops multiple users compete for limited core network resources on demand and as available, with each switch serving as a point of contention. This sharing of limited network resources clearly allows the network providers to realize significant operational efficiencies, which are reflected in lower network costs. The end users realize the additional advantages of flexibility and resiliency because the network generally can provide connection between any two physical locations through multiple alternate transmission paths.

In the domain of traditional circuit-switched voice networking, all local, regional, national, and international networks are interconnected. The cost of establishing switched circuits traditionally is sensitive to factors such as the distance between originating and terminating locations, duration of the connection, time of day (prime time vs. nonprime time), and day of the year (business day vs. weekend day or holiday). Yet, circuit-switched connections offer great advantage for calls of short

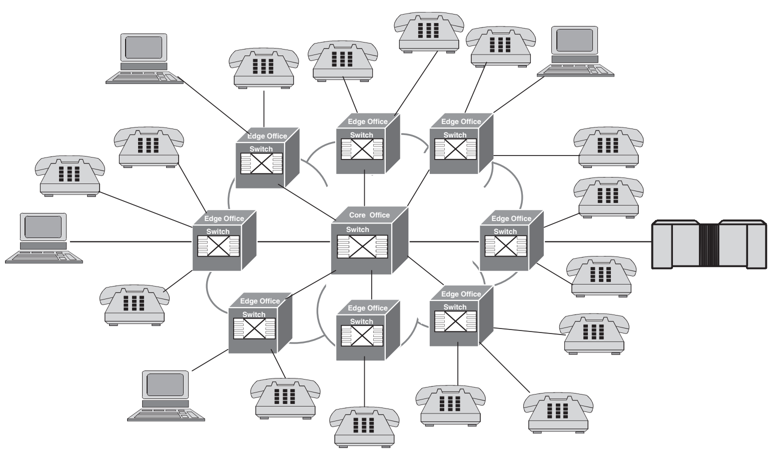

Figure 1.3 Circuit-switched connectivity between single-line telephone sets and between data terminals and a mainframe host computer through edge office and core office switches.

duration, connection between specific locations that communicate relatively infrequently, in cases where network redundancy is important, and at times when a high degree of flexibility is advantageous. Interconnectivity typically is much more selective in the data domain, with the exception of the Internet. Although the Internet is virtually ubiquitous, some governments place restrictions on access and content. The vast majority of voice calls and data calls are carried over switched circuits.

1.2.3 Virtual Circuits

Virtual circuits are logical, rather than physical, circuits. Virtual circuit connectivity is provided over high-capacity, multichannel physical circuits, such as fiber-optic transmission facilities. Virtual circuits are established through the network based on options and instructions defined in software routing tables. *Permanent Virtual Circuits* (PVCs) are permanently defined in routing tables, until such time as the carrier permanently redefines them. *Switched Virtual Circuits* (SVCs) are determined at the moment in time the communication is requested, with relatively sophisticated devices making highly informed decisions about the best path available in support of the specific requirements of the communication. In either case, a virtual circuit provides connectivity much as though it were a physical circuit, with all data traveling the same path. Such a physical circuit often can support a great number of logical circuits, or logical connections. In the high-capacity, fiber-optic backbone carrier networks, dedicated circuits are provided to users on a virtual basis, with the capacity and other performance characteristics of the circuit performing as though the circuit were dedicated.

Now it is worth pausing to further define and contrast the terms *transparent* and *virtual*. Transparent means that a network element (e.g., hardware or software) exists but appears to the user as though it does not. Without special test equipment, the end user may be totally unaware of its existence. Virtual means that the network element behaves as though it were something more than it actually is. So, a user can access a virtual circuit on a transparent basis.

It also is necessary to further define and distinguish between logical and physical circuits and channels. A *logical circuit* refers to the entire range of network elements (e.g., physical circuits, buffers, switches, and control devices) that support or manage communication between a transmitter and receiver. A single logical circuit can support many *logical channels*. In order to establish and support the information transfer, a *physical circuit*, or *physical path*, must be selected for the information transfer. A single physical circuit can support many logical circuits. The transmission facilities in the physical path may be in the form of copper wire (e.g., twisted-pair or coaxial cable), radio (e.g., microwave or satellite), or glass or plastic fiber (fiber optic) [1].

1.3 TWO-WIRE VERSUS FOUR-WIRE CIRCUITS

Telegraph and telephone circuits originally were metallic one-wire, which proved satisfactory even for two-way communications. Soon after the invention of the telephone, however, two-wire circuits were found to offer much better performance characteristics, due largely to their improved immunity from electromagnetic

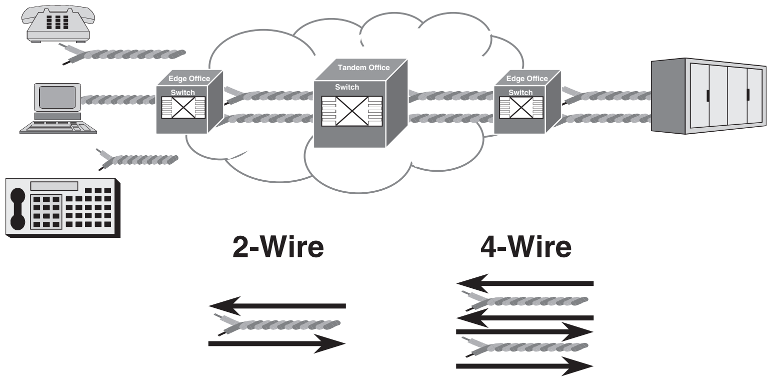

Figure 1.4 Applications for two- and four-wire circuits.

interference. Four-wire circuits offer still better performance, although at higher cost. Both two- and four-wire circuits are widely used.

1.3.1 Two-Wire Circuits

Two-wire circuits carry information signals in both directions over the same physical link or path. Typically, such a circuit is provisioned through the use of a single twisted-pair, copper wire connection. Within such a two-wire circuit, two wires are required to complete the electrical circuit, with the current in one wire opposite to the current in the other, and both wires carry the information signal. A common example is a local loop connection between a telephone company's CO switching center and an individual single-line or multiline telephone set, data terminal, or Key Telephone System (KTS), as depicted in Figure 1.4.

Two-wire circuits generally cover a short distance; the vast majority of two-wire local loops, for example, are less than 18,000 ft in length [2]. Longer loop lengths require some form of amplification in order to maintain signal strength. Additionally, such a circuit offers relatively little bandwidth, or capacity, and is single channel in nature (i.e., supports only a single conversation). Finally, two-wire circuits, generally speaking, are analog in nature; therefore, error performance (quality) is relatively poor. Two-wire circuits often are characterized as *voice grade*, that is, good enough for voice communications between humans, who are reasonably intelligent devices capable of adapting to errors in transmission over a circuit of relatively poor quality. A voice-grade circuit also will support low-speed data transmission through a modem, which has internal mechanisms for dealing with transmission errors. Two-wire circuits of lesser grade serve lesser applications, such as burglar alarms and fire alarms.

1.3.2 Four-Wire Circuits

According to the most basic definition, *four-wire circuits* carry information signals in both directions over separate physical links or paths and in support of simultaneous, two-way transmission. Traditionally, such a circuit was provisioned through the use of two copper pairs, one for transmission (*forward path*, or *upstream path*) and one for reception (*reverse path*, or *downstream path*); such a circuit is known as *physical four-wire*. However, current technology accommodates four-wire transmission over a single physical link or path and over a variety of transmission media, including twisted-pair, coaxial cable, or fiber-optic cable. In other words, the circuit may be physical two-wire (or even physical one-wire) and *logical four-wire*, performing as a four-wire circuit but employing fewer than four wires. In fact, a four-wire circuit can be established without the use of any wires at all, as in the case with a circuit established over microwave, satellite, or infrared transmission systems.

Although the absolute cost of four-wire circuits is higher than that of two-wire circuits, they offer considerably improved performance. As four-wire circuits can accommodate multiple, simultaneous communications in a full-duplex mode, all multichannel circuits are four-wire. By virtue of their multichannel capability, four-wire circuits also are capable of supporting out-of-band signaling and control, which offers the significant advantage of being nonintrusive, that is, nondisruptive. Additionally, such circuits typically offer much greater bandwidth, or capacity, and typically are digital, rather than analog, in nature. As a result, error performance generally improves. Long-haul circuits (traditionally defined as equal to or greater than 50 miles, or 80 km) usually are four-wire [3], as the carriers, or service providers, typically aggregate large volumes of traffic for transport over multichannel facilities. Figure 1.4 illustrates typical examples of cost-effective applications of four-wire circuits, specifically to interconnect PBX, CO, and tandem switches in a voice environment.

1.4 BANDWIDTH

Bandwidth is a measure of the capacity of a circuit or channel. More specifically, it refers to the total frequency on the available *carrier* for the transmission of data. There is a direct relationship between the bandwidth of a circuit or channel and both its frequency and the difference between the minimum and maximum frequencies supported. While the information signal (bandwidth usable for data transmission) does not occupy the total capacity of a circuit, it generally and ideally occupies most of it. The balance of the capacity of the circuit may be used for various *signaling and control (overhead)* purposes. In other words, the total *signaling rate* of the circuit typically is greater than the effective *transmission rate*. The more information you need to send in a given period of time, the more bandwidth you require.

1.4.1 Carrier

Carrier is a continuous signal on a circuit that is at a certain frequency or within a certain frequency range. The primary value of the carrier is in its support of the information-bearing signal (i.e., it carries the information signal), which the transmitter impresses on the carrier by varying the signal in some fashion and which the receiver must detect and interpret. The carrier also can support signaling and control information used to coordinate and manage various aspects of network operations.

1.4.2 Hertz

Hertz (Hz), named after Heinrich Rudolf Hertz, the physicist who discovered radio waves, is the measurement of frequency. Hertz also is the measurement of analog bandwidth, measured as the difference between the highest and lowest frequencies over a circuit or within a channel. Hertz refers to the number of electromagnetic waveforms transmitted per second (i.e., signals per second or cycles per second). Although some applications operate in very low capacity environments, measured in tens of or hundreds of hertz, the frequencies generally are much higher. Hence and by way of example, you can measure analog bandwidth in kilohertz (kHz, or thousands of hertz), megahertz (MHz, or millions of hertz), gigahertz (GHz, or billions of hertz), and terahertz (THz, or trillions of hertz).

1.4.3 Baud

Baud is an olde term that refers to the number of signal events (i.e., signal changes or signal transitions) occurring per second over an analog circuit. The baud rate can never be higher than the raw bandwidth of the channel, as measured in Hz. *Baud rate* and *bit rate* often and incorrectly are used interchangeably. The relationship between baud rate and bit rate depends on the sophistication of the modulation scheme used to manipulate the carrier. The bit rate and baud rate can be the same if each bit is represented by a signal transition. The bit rate typically is higher that the baud rate as a single signal transition can represent multiple bits. Chapter 6 explores the distinction between baud rate and bit rate in more detail.

1.4.4 Bits and Bytes per Second

Quite simply, *bps* (lowercase *b*) is the bit rate, or the number of *bits* transmitted over a circuit per second. It is the measurement of bandwidth over digital circuits and should not be confused with the speed of the electromagnetic signal, that is, the velocity of propagation. In other words, bps refers to the number of bits that pass a given point in a circuit, not the speed at which they travel over a distance. Over an analog circuit, you can manipulate the electromagnetic waveforms to support the transmission of multiple bits per baud. As a result, the bit rate (bps) can be a multiple of the baud rate, even without the application of special compression techniques. A thousand (1000) bps is a kilobit per second, or kbp*s*; a million (1,000,000) bps is a Megabit per second, or *Mbps*; a billion (1,000,000,000) bps is a Gigabit per second, or *Gbps*; and a trillion (1,000,000,000,000) bps is a terabit per second, or *Tbps*.

Bps (uppercase *B*) refers to the number of *bytes* transmitted over a circuit per second. Bps is used exclusively in the context of storage networking, as storage is byte oriented. Storage technologies such as *Fibre Channel* and *ESCON* (Enterprise Systems CONnection) measure the speed of information in bytes per second.

1.4.5 Narrowband, Wideband, and Broadband

Bandwidth levels or ranges fall into three categories: narrowband, wideband, and broadband. These terms are imprecise, as there are no widely accepted, formal standard definitions. Also, the definitions vary, depending on the technological context.

Narrowband: *Narrowband* refers to voice-grade bandwidth. In analog telecommunications terms, a narrowband channel has bandwidth of a nominal 4 kHz, which is the standard for analog voice. In digital terms, a narrowband channel is 64 kbps, which is the fundamental standard for uncompressed, digitized voice. Narrowband sometimes is used to describe a channel or circuit of less than voice-grade bandwidth. Narrowband also is used to describe some number of 64-kbps channels ($N \times 64$ kbps). *Narrowband Integrated Services Digital Network* (N-ISDN), for example, comprises two information-bearing channels of 64 kbps each plus a signaling and control channel of 16 kbps, for a total of 144 kbps. So, narrowband essentially is used to describe a circuit or channel offering relatively little bandwidth. I use the term narrowband to describe bandwidth up to a maximum of 24 channels. In digital terms, this means bandwidth up to the T1 signaling rate of 1.544 Mbps, which supports 24 channels at 64 kbps, at least according to North American standards. The equivalent international standard is E1, which has a signaling rate of 2.048 Mbps and supports 30 channels at 64 kbps. (*Note:* You will find detailed information on narrowband ISDN, T1, and E1 in subsequent chapters.)

Wideband: *Wideband* is used to distinguish a circuit or channel with capacity greater than narrowband. Wideband sometimes is used to describe a circuit or channel that has bandwidth wider than normal for operation. In the radio domain, wideband refers to a radio channel covering a relatively wide range of frequencies. *Ultra-Wideband* (UWB), for example, is defined as a radio system with occupied bandwidth (i.e., the difference between the highest and lowest frequencies in the radio channel) greater than 25 percent of the center frequency. Wideband sometimes is used interchangeably with broadband, as if the terminology were not already confusing enough.

Broadband: *Broadband* is an imprecise, evolving term referring to a circuit or channel providing a relatively large amount of bandwidth. I generally use the term to describe capacity equal to or greater than the nominal T1 rate of 1.544 Mbps, which is the basis for Broadband ISDN (B-ISDN), according to North American standards. European and international standards define B-ISDN at the E1 rate of 2.048 Mbps. Broadband has an entirely different definition in the context of LANs. You will learn about LANs in Chapter 8.

1.5 ANALOG VERSUS DIGITAL

Along one dimension, communications fall into two categories, analog and digital. In the analog form of electronic communications, information is represented as a continuous electromagnetic waveform. Digital communications involves modulating (i.e., changing) the analog waveform in order to represent information in binary form (1s and 0s) through a series of blips or pulses of discrete values, as measured at precise points in time or intervals of time.

1.5.1 Analog Sine Waves: Starting Point

Analog is best explained by examining the transmission of a natural form of information, such as sound or human speech, over an electrified copper wire. In its native form, human speech is an oscillatory disturbance in the air that varies in terms of its volume or power (amplitude) and its pitch or tone (frequency). In this native acoustical mode, the variations in amplitude cause the physical matter in the air to vibrate with greater or lesser intensity and the variations in frequency cause the physical matter in the air to vibrate with greater or lesser frequency. So, the physical matter in the space between the speaker's mouth (transmitter) and the listener's ear (receiver) serves to conduct the signal. That same physical matter, however, also serves to *attenuate* (weaken) the signal. The longer the distance is between mouth and ear, the more profound the effect. As a result, it is difficult, if not impossible, to communicate acoustically over distances of any significance, especially between rooms separated by doors and walls or between floors separated by floors and ceilings. In order to overcome these obvious limitations, native voice acoustical signals are converted into electromagnetic signals and sent over networks, with the compression waves falling onto a microphone in a transmitter embedded in a handset or speakerphone. The microphone converts the acoustical signals into *analogous* (approximate) variations in the continuous electrical waveforms over an electrical circuit, hence the term analog. Those waveforms maintain their various shapes across the wire until they fall on the speaker embedded in the receiver. The speaker converts them back into their original acoustical form of variations in air pressure, which can be received by the human ear and understood by the human brain.

A similar but more complicated conversion process is used to transmit video over networks. In its native form, video is a series of still images, each comprising reflected light waves. Transmitted in rapid succession, the series of still images creates the illusion of fluidity of motion. The transmitter (i.e., video camera) creates analogous variations in electrical or radio waveforms, which it sends in rapid succession over a network to a receiver (i.e., monitor), which re-creates an approximation (analog) of the original information.

Information that is analog in its native form (voice and other forms of audio and image and video) can vary continuously in terms of intensity (volume or brightness) and frequency (tone or color). Transmission of the native information stream over an electrified analog network involves the translation of those variations into amplitude and frequency variations of the carrier signal. In other words, the carrier signal is *modulated* (varied) in order to create an analog of the original information stream.

The electromagnetic sinusoidal waveform, or *sine wave*, as illustrated in Figure 1.5, can be varied in amplitude at a fixed frequency using *Amplitude Modulation* (AM). Alternatively, the frequency of the sine wave can be varied at constant amplitude using *Frequency Modulation* (FM). Additionally, both frequency and amplitude can be modulated simultaneously to create an analog of the native signal, which generally varies simultaneously along both parameters. Finally, the position of the sine wave can be manipulated (actually, can appear to be manipulated), adding the third technique of *Phase Modulation* [PM, also known as *Phase Shift Keying* (PSK)]. Chapter 6 discusses these modulation techniques in considerable detail.

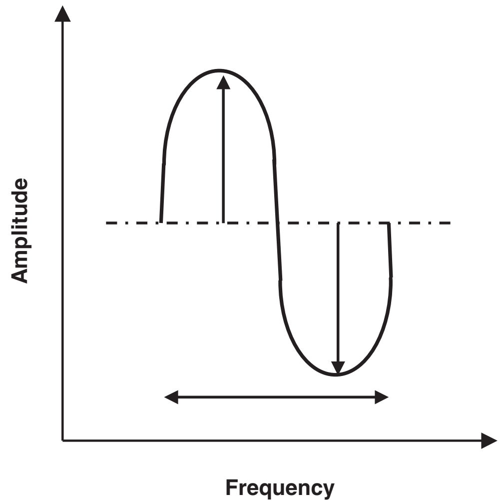

Figure 1.5 Sinusoidal waveform illustrating amplitude and frequency.

Bandwidth, in the analog world, is measured in Hertz (Hz). The available bandwidth for a particular signal is the difference between the highest and lowest frequencies supported by a channel or circuit. For example, a circuit can support a 3.0-kHz voice channel through the use of a *bandpass* (i.e., band-limiting) filter supporting transmission at frequencies between approximately 300 and 3300 Hz. Similarly, a circuit can support a 3.0-kHz channel at frequencies between 7000 and 10,000 Hz. *Passband* refers to the upper and lower cutoff frequencies at which the bandpass filters operate [2].

1.5.1.1 Voice The signaling rate of a voice-grade channel is nominally (approximately) 4000 Hz, or 4 kHz. The bandwidth in the range 0–300 Hz generally is ignored, suppressed by the equipment's lack of ability to deal with it at those low frequencies. The voice band is approximately 3.0 kHz wide, running at 300–3300 Hz. Signaling and control functions take place in the band 3300–3700 Hz. The lower band of 0–300 Hz and the upper band of 3700–4000 Hz have value, as they are used for maintaining separation between information channels, each of which is supported over a separate carrier frequency range, when analog voice channels are multiplexed using Frequency Division Multiplexing (FDM), which is described later in this chapter. While human speech can transmit and human hearing can receive a much wider range of frequencies, 3.0 kHz is considered sufficient for voice communications and certainly is more cost effective for the service providers than attempting to support full-fidelity voice. [*Note:* At 300 Hz, the cutoff frequency is high enough to reject AC (Alternating Current) electrical hum at 60 Hz in North American networks and 50 Hz in European networks. At 3300 Hz, the frequency is high enough to include all of the important harmonics that make human voice recognizable. Voice bandwidth and frequency range vary, with some filters operating at 200–3500 Hz, some at 300–3400 Hz, and so on.] Band-limiting filters employed in carrier networks constrain the amount of bandwidth provided for a voice application, which certainly conserves bandwidth. Capping the bandwidth at 3300 Hz also prevents

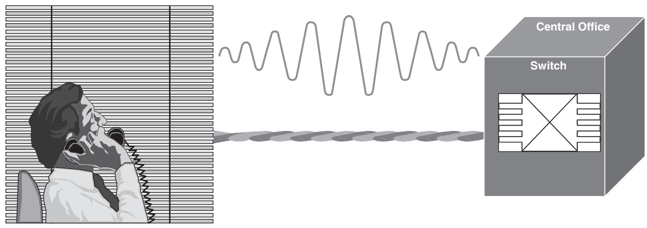

Figure 1.6 Analog voice transmission over a two-wire local loop.

aliasing, a phenomenon that occurs when different continuous signals overlap and become indistinguishable (i.e., becoming *aliases* of one another) when encoded into digital format for transmission over digital facilities [3, 4]. Figure 1.6 illustrates an analog local loop supporting voice communications.

1.5.1.2 Video An analog cable TV [Community Antenna TeleVision (CATV)] video channel has a width of approximately 6,000,000 Hz, or 6 MHz. Approximately 4.5 MHz is used for transmission of the video signal and the balance is used for *guard bands* to separate the various adjacent channels riding the common, analog coaxial cable system.

1.5.2 Digital Bit Streams: Ones and Zeros

While the natural world is analog in nature, the decidedly unnatural world of contemporary computers is digital in nature. Computers process, store, and communicate information in binary form. That is to say that a unique combination of *1*s and *0*s has a specific meaning in a computer coding scheme, which is much like an alphabet. A *bit* (*bi*nary digi*t*) is an individual *1* or *0*. The output of a computer is in the form of a *digital bit stream*.

Digital communication originates in telegraphy, in which the varying length (in time) of making and breaking an electrical circuit results in a series of *dots* (short pulses) and *dashes* (long pulses) that, in a particular combination, communicate a character or series of characters. Early mechanical computers used a similar concept for input and output. Contemporary computer systems communicate in binary mode through variations in electrical voltage.

Digital signaling, in an electrical network, involves a signal that varies in voltage to represent one of two discrete and well-defined states. Two of the simplest approaches are *unipolar* signaling, which makes use of a positive (+) voltage and a *null*, or zero (0), voltage, and *bipolar* signaling, which makes use of a positive (+) or a negative (−) voltage. The transmitter creates the signal at a specific carrier frequency and for a specific duration (*bit time*), and the receiver monitors the signal to determine its state (+ or −). Various data transmission protocols employ different physical signal states, such as voltage level, voltage transition, or the direction of the transition. Because of the discrete nature of each bit transmitted, the bit form is often referred to as a *square wave*. Digital devices (Figure 1.7) benefit greatly from communications over digital transmission facilities, which are not only faster but also relatively free from noise impairments.

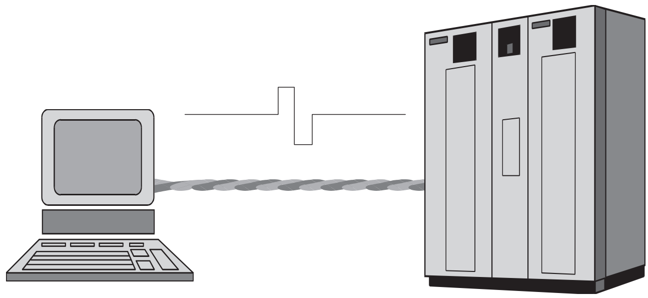

Figure 1.7 Digital communications between a terminal and host.

Digital signaling in an optical network can involve either the pulsing on and off of a light source or a discrete variation in the intensity of the light signal. Digital transmission over radio systems (e.g., microwave, cellular, or satellite) can be accomplished by discretely varying the amplitude, frequency, or phase of the signal.

Bandwidth, in the digital world, is measured in bits per second. The amount of bandwidth required depends on the amount of raw data to be sent, the desired speed of transmission of that set of data, and issues of transmission cost. Compression of data files prior to transmission is fairly routine, as it improves the efficiency of transmission, reduces the transmission time, and thereby reduces transmission costs.

1.5.3 Analog versus Digital Transmission

Transmission systems are either analog or digital in nature. In an analog transmission system, all components operate in analog (continuous-waveform) mode. Similarly, a digital transmission system must be digital from end to end. A network may consist of both analog (A) and digital (D) transmission systems, with A-to-D and D-to-A conversions required to resolve the obvious issues of incompatibility. Analog and digital signal formats each have advantages and appropriate applications.

1.5.3.1 Analog Advantages Analog transmission offers advantages in the transmission of analog information, in which case it is more bandwidth-conservative than is digital transmission. As analog transmission facilities formed the foundation for telecommunications networking, they were widely deployed and remain virtually ubiquitous.

1.5.3.1.1 Analog Data Analog has an advantage with respect to the transmission of information that is analog in its native form, such as voice and video. The process of transmission of such information is relatively straightforward in an analog format, as the continuous flow of the native signal is easily and fully represented in the continuous flow of the carrier signal. Although filters may constrain the amplitude and frequency levels of the transmitted signal, the essence of the native signal and its infinite variations is faithfully represented through an analog system and over an analog network. Conversion of analog data to a digital bit stream requires special conversion equipment. Such equipment adds cost, contributes additional points of failure, and can negatively affect the quality of the signal through the conversion process itself. The end result is very much an approximation of the original data. The examination of T-carrier in Chapter 7 includes discussion of the impacts of analog and digital conversions.

1.5.3.1.2 Bandwidth A raw analog information stream, if fully and faithfully transmitted, consumes far less bandwidth in analog form than in digital form. This is particularly evident in CATV transmission, where 50 or more analog channels routinely are provided over a single coaxial cable system. Without the application of relatively sophisticated compression techniques, fewer digital channels could be supported.

1.5.3.1.3 Availability Finally, analog transmission systems are in place worldwide. All aspects of the standards are well understood and easily implemented, and the interconnection of analog systems is routine. As voice traditionally comprised the majority of network traffic and as the vast majority of voice terminals still are analog devices, voice communication largely continues to depend on analog networks at the local loop level. Conversion to fully digital networks would require prohibitively expensive, wholesale conversion of such terminal equipment and local loops.

1.5.3.2 Digital Advantages Digital transmission certainly is advantageous for the transmission of digital information. Additionally, digital data can be compressed effectively and easily. Security of the data can be more readily ensured, and the error performance of digital networks is much improved over their analog counterparts. Finally, the cost effectiveness of such networks is improved by virtue of the greater bandwidth they provide, especially since they can be more easily upgraded and more effectively managed.

1.5.3.2.1 Digital Data Just as it often is better to transmit analog information in an analog format, it is better to transmit digital information in a digital format. Digital transmission certainly has the advantage when transmitting binary computer data. The (modem) equipment required to convert the information to an analog format and send the digital bit streams over an analog network represents additional cost, is susceptible to failure, and can induce errors into the datastream.

1.5.3.2.2 Compression Digital data can be compressed relatively easily, thereby increasing the efficiency of transmission. As a result, substantial volumes of computer data can be transmitted using relatively little raw bandwidth, with the receiving device decompressing the data to reconstitute it in its original form. Transmission of analog voice and video benefits from digital conversion if the analog signals are sampled at appropriate intervals, converted into byte format, and compressed. Several subsequent chapters discuss this process in detail.

1.5.3.2.3 Security Digital systems offer much improved security. While analog systems can offer some measure of security through the scrambling, or intertwining, of several frequencies, you can fairly easily defeat that technique. Digital information, conversely, can be *encrypted* to create the appearance of a single, pseudorandom bit stream. Thereby, the true meaning of individual bits, sets of bits, and the total bit stream cannot be determined without the key to unlock the encryption algorithm employed.

1.5.3.2.4 Error Performance Digital transmission offers much improved error performance (data integrity) in comparison with analog. This is due to the nature of the devices that serve to boost the signal at periodic intervals in the transmission system to overcome the effects of attenuation. Additionally, digital networks deal more effectively with *noise*, which always is present in transmission networks.

Attenuation: Electromagnetic signals tend to weaken, or *attenuate*, over a distance; this is particularly true of electrical signals carried over twisted-pair copper wire, due to factors including the level of resistance (or impedance) in the wire and the tendency of the signal to radiate, or spread out, from the wire. It also is true of microwave radio and other terrestrial radio systems, due to the physical matter in the air and the tendency of the signal to spread out, or disperse. Attenuation is sensitive to carrier frequency, with higher frequency signals attenuating more than lower frequency signals.

Noise: Signals also tend to pick up *noise* as they transverse the network. Again, this is particularly true of twisted-pair copper wire systems. Such wires tend to act as antennas and, therefore, absorb noise from outside sources of *ElectroMagnetic Interference* (EMI) and *Radio Frequency Interference* (RFI). The quality of the signal degenerates as it is distorted by the noise, and the integrity of the data transmission suffers as a result.

1.5.3.2.5 Cost The cost of the computer components required in the digital conversion and transmission process has dropped to a considerable extent, while the ruggedness and reliability of those components have increased over the years.

1.5.3.2.6 Upgradeability Since digital networks comprise computer components, they can relatively easily be upgraded, within design limits. Such upgrades might increase bandwidth, improve error performance, and enhance functionality. Certain upgrades often can be effected through software downloads over the network, thereby eliminating the need for a "truck roll," that is, the need to dispatch a technician.

1.5.3.2.7 Management Generally speaking, digital networks can be managed much more easily and effectively because they comprise computerized *Network Elements* (NEs). Such components can be endowed by their creators with the abilities to determine their status (i.e., on or off), sense their own levels of performance relative to programmed thresholds, isolate and diagnose failures, initiate alarms to upstream management systems, respond to queries, and respond to commands to correct the failure condition. Further, the cost of so enabling these devices is dropping rapidly.

1.6 LOADING COILS, AMPLIFIERS, AND REPEATERS

As noted earlier in this chapter, electromagnetic energy attenuates over a distance, whether the energy passes through a conductor or the air. Therefore, you must place some sort of device at regular spatial intervals in a network to overcome this phenomenon by boosting the signal strength. These boosting units receive a weakened incoming signal and transmit a stronger outgoing signal, which propagates across the network, weakening until it reaches another boosting unit, and so on. Analog networks make use of devices known as *loading coils* and *amplifiers*, which were originally known as relays. Digital networks employ *repeaters*.

> *On long lines the current often becomes so reduced by leakage, and from other causes, that it is insufficient to work an electro-magnet, either to mark paper, or give audible sound. It is therefore usual on such lines to interpose an instrument called a relay. A current weakened by distance, although unable to effectively work the receiving instrument, may have enough force to cause a light armature to be attracted by a small magnet. This movement may be made to bring a local battery into circuit so as to strengthen the current, and such an arrangement constitutes a relay.*
>
> —*Wonders of the Universe*, The Werner Company. 1899 [5]

1.6.1 Loading Coils

In order to ensure that local loops perform properly, they must be designed in such a way that signal strength is maintained at acceptable levels. The obvious solution to the problem of signal attenuation is to limit the length of the local loop. Therefore, the size of a *Carrier Serving Area* (CSA), that is, the geographical area served by a CO, is generally limited to a radius of about 18,000 ft. Beyond that distance, analog voice-grade signals of 4 kHz sent through Unshielded Twisted-Pair (UTP) copper cables attenuate to such an extent that they become unusable. Now, the attenuation increases substantially across all frequencies in the band because of the high *capacitance* created between the two tightly spaced conductors comprising a cable pair. The performance of long copper local loops can be improved through the use of *induction coils* known as *load coils* or *loading coils*. A load coil is a toroidal (i.e., ring-shaped or donut-shaped) device made up of a powdered iron core or sometimes a soft iron wire core around which copper wire is wound. The coil is then spliced into the local loop at some point where it can be properly sheltered in a weatherproof splice case, underground vault, or some similarly protected environment.

The coil functions as a *lumped* inductor, which is to say that at a specific point in the circuit the process of inductance takes place to compensate for the distributed capacitance. In effect, the load coil tunes the copper circuit, optimizing it for mid-voice-band performance. The load coil also functions as a low-pass filter, increasing loss above the cutoff frequency, which is 4 kHz in this case. The coil can reduce mid-voice-band attenuation by as much as 80 percent. Load coils commonly are placed on local loops that exceed approximately 18,000 ft (5.5 km) in length. The first load coil is placed approximately 3000 ft (0.9 km) from the CO and at intervals of 6000 ft (1.8 km) or so thereafter. Load coils are passive devices, that is, not electrically powered, and generally are limited to use in analog loops. The presence of load coils renders local loops unusable for ISDN, T-carrier, and other loops operating at high data rates, as the load coils filter out the high frequencies that accompany those higher data rates. The existence of so many loaded loops within the existing telephone plant was a significant deployment deterrent when ISDN was introduced some 20–25 years ago, as ISDN requires frequencies above 4 kHz. Neither are load coils acceptable for use on Asymmetric Digital Subscriber Line (ADSL) and other broadband local loops, as the frequency ranges far exceed 4 kHz. Where such services are to be deployed, the local loops must be properly *conditioned*, which entails removing the load coils and other impediments.

The presence of a load coil also has the effects of increasing the impedance of the circuit and reducing the velocity of propagation, that is, speed of signal propagation, to 10,000–12,000 miles per second. This speed penalty is not of particular significance in short voice-grade local loops. If the loops are long, however, load coils can create unacceptable problems with *echo*, or signal reflection. At any point in a circuit where an electromagnetic wave meets a discontinuity, a portion of the wave is reflected back in the direction of the transmitter. Such discontinuities can be caused by impedance mismatches, mismatches between line and balancing networks, irregular spacing of loading coils, and a host of other anomalies that are beyond the scope of this book. Now, echo is not a problem for human-to-human conversations as long as the echo return is not longer that 30–40 milliseconds (ms), which means that a circuit that propagates electromagnetic waves at near the speed of light (186,000 miles per second) can be 3000–4000 miles long without creating an echo problem. However, a circuit of 1000 miles with a signal traveling at 10,000 miles per second returns an echo in 100 ms, which makes conversations nearly impossible. In contemporary networks, high-speed carrier circuits are conditioned, with loading coils removed, so the problem largely disappears except in international communications or where satellite circuits are involved. Contemporary networks also are designed with echo cancelers or echo suppressors to deal with this problem, although it still occurs on occasion when echo suppressors fail.

1.6.2 Amplifiers (Analog)

The active boosting devices in an analog network are known as *amplifiers*. Amplifiers are unsophisticated devices that simply boost, or amplify, the weak incoming signal, much as does an amplifier in a radio receiver or TV set. In addition to attenuating, the signal accumulates noise as it transverses the network; the amplifier boosts the noise along with the signal. This effect is compounded through every step of the transmission system and through each cascading amplifier, thereby creating the potential for significant accumulated noise at the receiving end of the transmission. The resulting *Signal-to-Noise Ratio* (SNR) can produce unacceptable results. Amplifiers are spaced every 18,000 feet or so in a typical analog voice-grade twisted-pair local loop, for example. The exact spacing is sensitive to a number of factors, including the transmission medium and the carrier frequency, which affects raw bandwidth, transmission speed, and attenuation level. According to William Shockley, coinventor of the transistor [6]:

> *If you take a bale of hay and tie it to the tail of a mule and then strike a match and set the bale of hay on fire, and if you then compare the energy expended shortly thereafter by the mule with the energy expended by yourself in the striking of the match, you will understand the concept of amplification.*

The impact of amplification on voice communications generally is tolerable, as humans are relatively intelligent receivers who can filter out the noise or at least adjust to it. In the event of a truly garbled transmission, the human-to-human error detection and correction process simply involves a request for retransmission—the *Huh?* protocol. Should the quality of the connection be totally unacceptable, you can terminate and reestablish the connection. Computer systems, however, are not so forgiving, and garbled data are of decidedly negative value.

There are several exceptions to this broad characterization of amplifiers. *Erbium-Doped Fiber Amplifiers* (EDFAs), used in high-speed fiber-optic systems, amplify light signals falling in a narrow optical frequency range, performing much more cost effectively than optical repeaters. *Raman amplification*, another amplification technique used in fiber-optic systems, makes use of pump lasers that send a high-energy light signal in the reverse direction (i.e., the direction opposite the signal transmission). This technique not only increases the strength of the signal but also serves to improve its clarity. Note that the immunity of fiber-optic systems to ambient noise makes the use of such amplification techniques quite acceptable. A detailed discussion of EDFAs and Raman amplification can be found in Chapter 9.

1.6.3 Repeaters (Digital)

Digital systems generally replace periodic amplifiers with *regenerative repeaters* that regenerate the signal, rather than simply amplifying it. In an electrically based system, for example, the repeater essentially guesses the binary value (1 or 0) of the weak incoming signal based on its relative voltage level and regenerates a strong signal of the same value without the noise. This process considerably enhances the signal quality. Repeaters are spaced at approximately the same intervals as amplifiers, which is approximately 18,000 feet in voice-grade twisted-pair circuits.

The performance advantage of digital networks can be illustrated by comparing the error rates of amplifiers and regenerative repeaters. A twisted-pair, analog network, for example, yields an error rate on the order of 10^{-5} [3]. In other words, digital data sent across an analog network through modems will suffer one errored bit for every 100,000 bits transmitted (Figure 1.8). The very same twisted-pair network, if digitized and equipped with repeaters, will yield an expected error rate of 10^{-7}, or one errored bit in every 10,000,000, which is an improvement of two

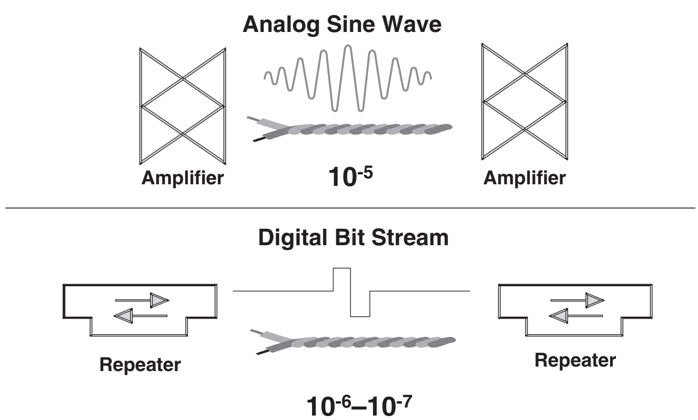

Figure 1.8 Comparative error performance of analog versus digital transmission over twisted pair.

orders of magnitude. Digital fiber-optic systems, currently considered to be the ultimate in transmission systems, commonly yield error rates in the range between 10^{-11} and 10^{-14}, or an error rate as low as 1 bit for every 100,000,000,000,000 transmitted, which is virtually perfect [4].

1.7 CONVERSION PROCESS: MODEMS AND CODECS

Regardless of the relative merits of analog and digital transmission, both technologies are in place. Local loops, which connect the user premises to the COE, generally are analog, at least in residential and small-business applications. Medium- and large-size businesses typically make use of digital local loops in the form of either T-carrier or ISDN over twisted pair. Cellular radio networks in the United States originally were analog, although most now are digital. High-capacity, backbone carrier transmission generally is digital. Analog-to-digital and D-to-A conversions take place routinely in contemporary networks.

1.7.1 Digital to Analog: Modems

As local loops often are analog, computer communications across such circuits are not possible without the assistance of a device to accomplish the D-to-A conversion. Where a digital circuit is available, it generally is considerably more expensive.

The device that accomplishes the D-to-A conversion process is known as a *modem*. Modems *mo*dulate and *dem*odulate the analog carrier wave in order to represent digital bit streams across the analog local loop, reconstructing the digital signal on the receiving end through a process of A-to-D conversion (Figure 1.9). A variety of techniques, explained in Chapter 6, are used to accomplish this process.

1.7.2 Analog to Digital: Codecs

The opposite conversion process is necessary to send analog information across a digital circuit. Certainly, this occurs often in carrier networks, where huge volumes of analog voice are digitized and sent across high-capacity digital circuits. This requirement also exists where high-capacity digital circuits connect premises-based, PBX voice systems to COEs or to other PBXs, assuming that the PBXs or COs have not already performed the conversion. As video also is analog in its native form, a similar process must be employed to send such information across a digital circuit.

The device that accomplishes the A-to-D conversion is called a *codec*. Codecs *co*de an analog input into a digital (data) format on the transmit side of the con-

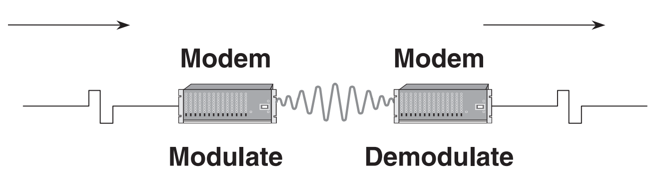

Figure 1.9 Modem: D-to-A and A-to-D conversion.

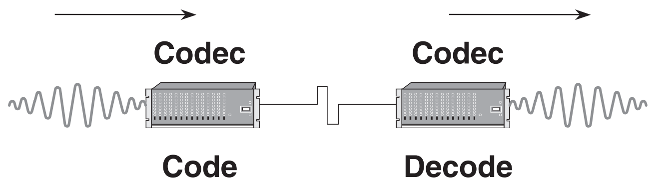

Figure 1.10 Codec: A-to-D and D-to-A conversion.

nection, reversing the process, or *dec*oding the information, on the receive side, to reconstitute an approximation of the original analog signal (Figure 1.10).

Encoding is the process of converting an analog information stream (e.g., voice or video) into a digital data stream. The voice or video signal is sampled at frequent intervals and each sample is expressed in terms of a binary value, usually a four- or eight-bit byte (i.e., data word). The reverse process of *decoding* takes place on the receiving end, resulting in recomposition of the information in its original form, or at least a reasonable approximation thereof.

1.8 MULTIPLEXERS (MUXES)

The term *multiplex* has its roots in the Latin words *multi* (many) and *plex* (fold). Multiplexers (muxes) act as both concentrators and contention devices that enable multiple relatively low speed terminal devices to share a single high-capacity circuit (physical path) between two points in a network. The benefit of multiplexers is simply that they enable carriers and end users to take advantage of the economies of scale. Just as a multilane highway can carry large volumes of traffic in multiple lanes at high speeds and at relatively low incremental cost per lane, a high-capacity circuit can carry multiple conversations in multiple channels at relatively low incremental cost per channel.

> *The modern saying, "Time is Money," is indeed most of all true when applied to telegraphic signalling; and many endeavours have been made, not only to transmit signals with celerity, but also to transmit more than one communication at the same time along the same wire. This has been successfully done in the duplex system—by which a message is sent from either end of the same wire simultaneously; in the diplex system—in which two messages can be sent simultaneously in one direction; and in the quadruplex system, which combines the two former methods, and by which it is possible to convey four signals along the same wire at the same moment. This last method was invented by Mr. Edison.*
>
> —*Wonders of the Universe* [5]

Contemporary multiplexers rely on four-wire circuits, which enable multiple logical channels to derive from a single physical circuit and permit high-speed transmission simultaneously in both directions. In this manner, multiple communications (either unidirectional or bidirectional) can be supported. Multiplexing is used commonly across all transmission media, including twisted pair, coaxial and fiber-optic cables, and microwave, satellite, and other radio systems.

Traditional multiplexing comes in several varieties, presented in the following sections in chronological order of development and evolution. Included are Frequency Division Multiplexing (FDM), Time Division Multiplexing (TDM) and Statistical Time Division Multiplexing (STDM). Wavelength Division Multiplexing (WDM), a relatively recent development, is used in fiber-optic cable systems.

1.8.1 Frequency Division Multiplexing

Frequency Division Multiplexing (FDM) takes advantage of the fact that a single twisted pair copper circuit, for example, can support much more than the 4 kHz guaranteed for an individual voice conversation. Even in the early days of vacuum tube technology, a set of two copper pairs (a four-wire circuit, with two wires supporting transmission in each direction) could support up to 96 kHz, thereby enabling the support of up to 24 individual voice channels separated by frequency guard bands [3]. In terms of a commonly understood analogy, a single four-wire electrical circuit can support multiple frequency channels through frequency separation much as the airwaves can support multiple radio stations and TV channels.

Through an FDM (Figure 1.11), conversation 1 might be supported over frequencies 0–4000 Hz, conversation 2 over frequencies 4000–8000 Hz, conversation 3 over frequencies 8000–12,000 Hz, and so on. Small slices of frequency within each channel are designated as subchannels, or *guard bands*, which separate the carrier channels used for information transmission. The guard bands serve to minimize the likelihood of interference between conversations riding in adjacent logical information channels over the same physical circuit. This prevents *crosstalk*, in which parties using adjacent channels hear each other unless the filters or frequency converters drift from their proper settings. Of course, the individual channels are not separated spatially; rather, they are overlaid, with all sharing the same physical space on the wires.

Frequency division multiplexers typically are not particularly intelligent. Specific devices or groups of devices often are tuned to using designated frequency bands for communications. As noted in Figure 1.11, the bandwidth associated with those

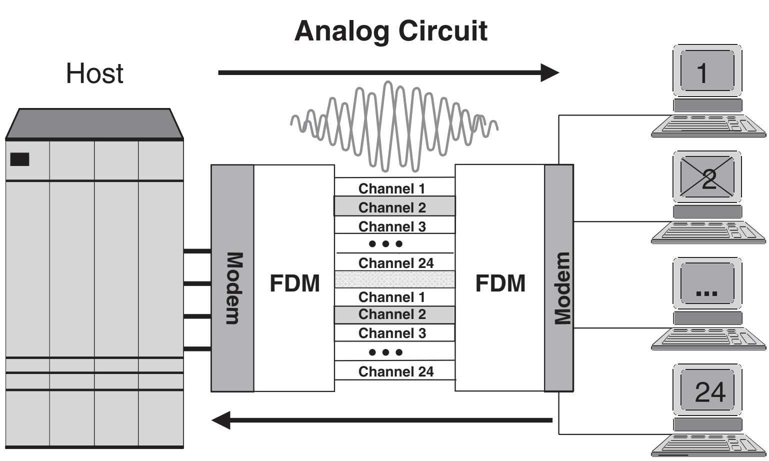

Figure 1.11 FDM in a data communications application.

X'ed-out devices is unused if the communication is inactive for some reason, even though other devices could make effective use of it.

FDM served its purpose well, at the time, for long-haul voice transmission. Data communications over FDM, however, requires sets of low-speed modems, one for each channel, with one set at each end of the facility. FDM currently is used in broadband LANs, which support multiple simultaneous transmissions. FDM also is used in cellular radio networks and in certain digitized voice applications. As noted, however (all else being equal), digital generally is better, especially when data traffic is involved, and data rules in this world.

1.8.2 Time Division Multiplexing

Time Division Multiplexing (TDM) offers all of the advantages of digital transmission, namely improved bandwidth utilization, enhanced error performance, improved security, and upgradability.

At the transmitting end of the connection, the time division multiplexer scans the buffers associated with the ports to which individual devices are attached. Each device port is allocated a channel in the form of a *time slot* for transmission of data. Device 1 transmits through port 1 and over time slot 1, device 2 transmits through port 2 and over time slot 2, and so on, in a serial fashion, as illustrated in Figure 1.12. The transmitting time division multiplexer typically accepts an 8-bit sample of data from each buffered port and byte interleaves those data samples into a *frame* of data. (*Note: Byte* generally refers to an eight-bit value, or *data word*, although some bytes comprise more or fewer bits. The term *octet* is more precise and frequently is used in telecommunications standards to describe an 8-bit unit of data.) As the mux completes a scan of the ports and transmits a set of such data samples, it prepends each frame with some number (1 in the case of T1) of *framing bits*. The

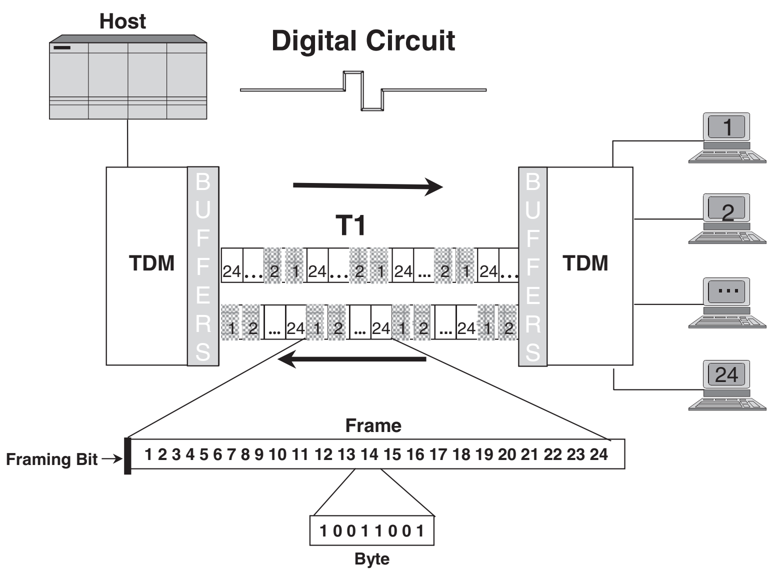

Figure 1.12 TDM in a data communications application.

framing bits are sent in a repeating pattern that delineates one frame from another and are used by the multiplexers and other intermediate devices for purposes of *synchronization*. Sophisticated multiplexers also use the framing bits for various signaling and control purposes.

At the receiving end, the process is reversed. Each channel in each frame is identified, the individual transmissions are demultiplexed, and each is forwarded over the port to which the intended receiving terminal device is attached. Clearly, the muxes must be carefully *synchronized* in time in order for the receiving mux to determine the proper separation of frames and channels of data. The framing bits typically provide the mechanism not only for synchronization between the muxes but also for the attached terminals.

The primary constraint of basic TDM is that of static configuration. In other words, channel 1 is always reserved for port 1, over which terminal 1 always transmits. Terminals that are idle, turned off, unplugged, or otherwise out of action are still allocated valuable bandwidth, which has a negative effect on the cost effectiveness of the facility, as illustrated in Figure 1.12. As their bandwidth allocation capabilities are so limited, basic time division multiplexers are generally considered obsolete.

1.8.3 Statistical Time Division Multiplexing

Statistical Time Division Multiplexing (STDM) is much improved over TDM because the muxes are intelligent. STDMs, or *Stat muxe*s, offer the advantage of dynamic allocation of available channels and raw bandwidth. In other words, STDM can allocate bandwidth, in the form of time slots, in consideration of the transmission requirements of individual devices serving specific applications (Figure 1.13). An STDM also can oversubscribe a trunk, supporting aggregate port speeds that may be in the range of 3–10 times the trunk speed by buffering data during periods of high activity. Further, an intelligent STDM can dynamically adapt to the changing

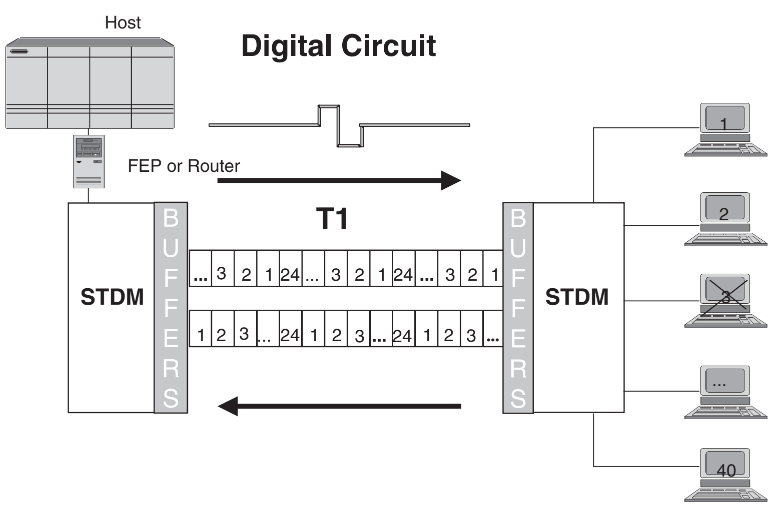

Figure 1.13 STDM in a data communications application.

nature and associated requirements of the load placed on it and in consideration of the available capacity of the circuit.

Stat muxes can recognize active versus inactive devices as well as priority levels. Further, they can invoke *flow control* options that can cause a transmitting terminal to cease transmission temporarily, in the event that the mux's internal *buffer*, or temporary memory, is full, thereby preventing a fast transmitter from overwhelming a slower receiver. Flow control also can restrain low-priority transmissions in favor of higher priority transmissions. Additionally, an STDM may offer the advantages of data compression, error detection and correction, and reporting of traffic statistics.

An STDM typically divides a high-speed, four-wire digital circuit into multiple time slots to carry multiple voice conversations or data transmissions. Channelized T1 (North America), for example, commonly provides 24 time slots to carry 24 conversations, each of a maximum of 64 kbps. Channelized E1 (European and international) commonly provides 30 time slots in support of 30 conversations.

Additionally, the individual channels can be grouped to yield higher transmission rates (*superrate*) for an individual bandwidth-intensive communication such as a videoconference. The individual channels also can be subdivided into lower speed (*subrate*) channels to accommodate many more, less bandwidth-intensive communications, such as low-speed data. Also, many muxes allocate bandwidth on a priority basis, providing delay-sensitive traffic (e.g., real-time voice or video) with top priority to ensure that the presentation of the data at the receiving end is of high quality.

1.8.4 Wavelength Division Multiplexing

Wavelength Division Multiplexers (WDMs) enable a single fiber-optic transmission system to support multiple high-speed channels through the transmission of multiple *wavelengths* of light, or *lambdas.* (*Note*: Physicists denote wavelength with the Greek letter λ (lambda), with wavelength being inversely proportional to *frequency*.) Much as multiple electrical frequencies can support multiple, simultaneous conversations in an FDM transmission system, multiple wavelengths can coexist on a single fiber of the appropriate type. A number of carriers now routinely deploy *Dense Wavelength Division Multiplexing* (DWDM) on fiber-optic systems, with eight or more lambdas introduced into the optical fiber through the use of *tunable lasers* firing through *windows*, or wavelength ranges. Each lambda might run at 2.5 Gbps, for a total yield of 20 Gbps per fiber strand. If each of 48 lambdas on a standard DWDM grid were to run at 10 Gbps, for example, the total yield would be 480 Gbps per fiber strand. At last count, the International Telecommunications Union—Telecommunications Standardization Sector (ITU-T) had defined 160 wavelengths at spacings of 100 GHz and manufacturers currently offer DWDM systems that multiplex as many as 80 lambdas. As some carriers deploy hundreds of fiber strands along a given path, the bandwidth potential is incredible.

1.8.5 Inverse Multiplexers

Inverse multiplexers perform the inverse process of traditional muxes. In other words, they accommodate a single, high-bandwidth data stream by transmitting it over multiple, lower bandwidth channels or circuits. The transmitting mux segments

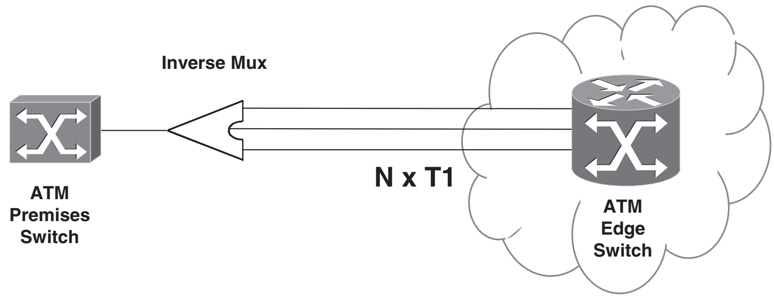

Figure 1.14 Inverse multiplexing over ATM.

the data stream and spreads it across the circuits on a consistent and coordinated basis, and the receiving mux reconstitutes the composite data stream. Clearly, the two devices must synchronize carefully with each other and with the transmission characteristics of the individual paths and channels in order to minimize errors and delays. An individual communication might spread over multiple switched circuits, dedicated circuits, or channels on multichannel circuits. A broadcast quality videoconference, for example, requires a full T1 (1.544 Mbps). Assuming that a full T1 is currently unavailable between two locations, the mux might split the signal across portions of multiple T1s and recombine at the receiving end.

Inverse Multiplexing over ATM (IMA) fans out an *Asynchronous Transfer Mode* (ATM) cell stream across multiple circuits between the user premises and the edge of the carrier network. Where significant levels of ATM traffic are destined for the WAN, a single circuit of appropriate bandwidth may either be unavailable or too costly. In such a circumstance, multiple physical T1 circuits can be used as a single, logical ATM pipe. The IMA-compliant ATM concentrator at the user premises spreads the ATM cells across the T1 circuits in a round-robin fashion, and the ATM switch at the edge of the carrier network scans the T1 circuits in the same fashion in order to reconstitute the cell stream (Figure 1.14). There is a similar Implementation Agreement (IA) for Frame Relay (FR), and Multilink Point-to-Point Protocol (MPPP) serves much the same purpose in the Internet domain.

1.8.6 Data over Voice and Voice over Data

A large number of manufacturers now offer muxes that enable data to be sent over voice lines and voice to be sent over data lines. A digital data circuit, for example, also can accommodate voice (for which it was not intended) through the use of a special mux that digitizes the voice signal and transmits it; the reverse process takes place at the receiving end. The voice and data conversations share the same circuit sequentially, rather than simultaneously. Bandwidth is allocated as appropriate, with priority provided to the delay-sensitive voice traffic.

This approach enables the user to take advantage of excess capacity on a dedicated data circuit. It also can support both voice and data communications over a single circuit-switched analog circuit. There is, of course, an investment required in the multiplexing equipment, although such equipment is quite affordable.

A number of manufacturers have developed muxes and routers that enable voice to share excess capacity on a Frame Relay network. While the quality of *Voice over*

Frame Relay (VoFR) can suffer from delay due to network congestion, the voice conversation essentially is free, as the justification for the Frame Relay network is in support of data communications requirements. *Voice over IP* (VoIP) can be supported over the Internet, although the quality can be poor due to the many uncertainties inherent in the Internet. VoIP quality is much better over high-speed *Internet Protocol* (IP) networks optimized for voice, rather than for the data traffic for which the protocol was originally developed. While the availability of these networks remains somewhat limited at this point, their potential is quite significant. VoIP promises to support toll-quality voice at much lower cost, due to the inherent efficiencies of packet switching as compared to circuit switching. These networks also are designed to support voice, data, video, and image traffic over the same infrastructure, with priority granted to real-time voice and video; such an integrated network offers inherent efficiencies and a single point of service management. You will find considerable detail about VoFR and VoIP in subsequent chapters.

1.9 SWITCHES AND SWITCHING: THE BASICS . . . AND THEN SOME

> *A click of metal against metal as an expert hand thrusts a plug in to a switchboard's face or as mechanisms shift automatically, at the command of electric impulses. Then . . . the simple question "Number, please?" or its mechanical equivalent, both alike in meaning: "Make way for the messages of the people!"*
>
> *Telephone Almanac*, American Telephone and Telegraph Company, 1934

Switches serve to establish connectivity between terminal devices (transmitters and receivers) on a flexible basis. They effectively serve as contention devices, managing contention between multiple transmit devices for access to shared circuits. In this manner, the usage and cost of expensive circuits can be optimized, based on standard traffic engineering principles. Without switches, each device would require a direct, dedicated circuit to every other device in a *full mesh topology*. Such an approach clearly is extremely resource intensive, highly impractical, and even completely impossible, as early experience proved. This discussion of switches and switching is presented in chronological order of development, beginning with circuit switching and its evolution and progressing through packet, frame, and cell switching. I also introduce the concepts of soft switching and optical switching.

1.9.1 Circuit Switching: Optimized for Voice

Circuit switches establish connections between physical circuits on a temporary, continuous, and exclusive basis. That is to say, on demand and as available, a circuit switch establishes connections through the switching matrix and provides continuous and exclusive access between the physical circuits for the duration of the conversation. From circuit to circuit and through the switching matrix, the switch establishes a transmission path, dedicating the prescribed level of bandwidth to that one transmission. Many switches may be involved in supporting an end-to-end circuit between terminal devices over a long distance. Contemporary circuit switches provide continuous access to a logical TDM channel within the shared internal bus of the switch for each call. A residential or small-business user typically gains access to a port on the switch over a single-channel, voice-grade analog circuit. A larger business user more commonly gains access through a high-capacity, multichannel digital circuit such as T1 copper local loop or perhaps an optical fiber. The high-capacity, multichannel circuits that interconnect the circuit switches in the core of the carrier network are known as *InterMachine Trunks* (IMTs), which generally are in the form of fiber-optic circuits. While circuit switches originally were developed for voice communications, much data traffic also is switched in this fashion. Typical examples of circuit switches include PBXs and COs.

1.9.1.1 Manual Switchboards These switches involve an operator who manually establishes the desired connection at the verbal request of the transmitting party. The original switch was in the form of a *switchboard*, which literally was a series of small, manual mechanical *jackknife switch*es mounted on a *board*. The operator manually switched the blade of the jackknife switch from one contact to another to establish a unique physical and electrical connection. While the term switchboard is used even today, the technology soon gave way to that of the *cordboard*, another manual switching technology that requires the operator to establish connections on a *plug-and-jack* basis, with the plugs on *cord*s and the jacks mounted on a board (Figure 1.15). Again, the operator establishes a unique physical and electrical connection that remains in place for the duration of the call. When either party disconnects, the operator is alerted and manually disconnects the circuit, which then becomes available for use in support of another call. The size of such switches, the complexity of interconnecting long-distance calls across multiple switches, and the labor intensity of this approach all contributed to their functional obsolescence many years ago. Note, however, that many thousands of such switches remain in service, largely in developing countries.

The term *tip and ring* came from the cordboard plugs that establish the connection between the copper wire pairs. One wire connects electrically through the *tip* of the plug, while the other wire in the pair connects through the *ring*, or seat, of the plug. The New Haven District Telephone Company in New Haven, Connecticut, installed the first such switch on January 28, 1878, at a cost of $28.50. The system connected 21 subscribers referred to by name, rather than telephone number. As

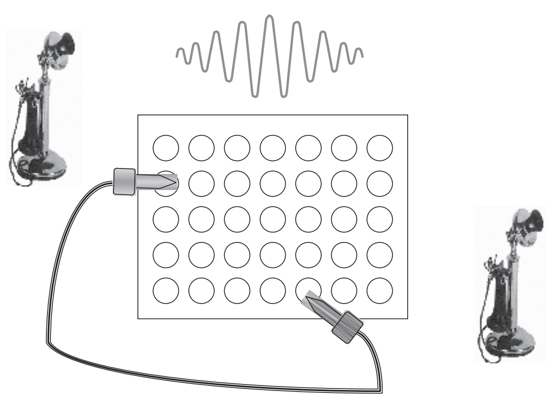

Figure 1.15 Manual switching: cordboard. (Courtesy of AG Communications Systems Corporation.)

was typical in those days, the operators were young boys, most of them experienced as messengers and telegraph clerks. The boys soon proved too boisterous and unreliable and were replaced with women [7].

1.9.1.2 Step-by-Step Switches Step-by-Step (S × S) switches are electromechanical in nature. Almon B. Strowger, a Kansas City undertaker frustrated with the behavior of the local telephone company operator, invented and patented the first such switch in 1891. According to legend, the operator was directing Mr. Strowger's calls to a competing undertaker, who also happened to be the operator's husband. Building on earlier Bell system work, Strowger invented a system that served 99 subscribers. The telephones that worked with that first automatic switch had two buttons. In order to reach subscriber 99, for example, the caller slowly and deliberately pressed the first button nine times and then the second button nine times. As the caller pressed a button, it would complete an electrical circuit, and as the caller released the button, it would break the circuit. Making and breaking the circuit would cause a mechanical rotary wiper on the switch to rotate from one contact to another. That patent served as the foundation for the company he founded, Automatic Electric Company, which later became the manufacturing subsidiary of General Telephone and Electric (GTE), which is now part of Verizon [7].

More contemporary S × S switches consist of a large number of *line finders* to which groups of individual subscribers are assigned for dial tone. The transmitting party dials a series of numbers, perhaps with a rotary dial telephone terminal, which causes the making and breaking of an electrical circuit. As displayed in Figure 1.16, those electrical pulses cause successive mechanical *line selectors* to step across contacts (one contact per electrical dial pulse) to set up the conversation path as the complete number is dialed [8]. *Note*: A touchtone telephone also can be used if a tone converter is installed on the line at the central office termination.

The S × S switches quickly gained favor, as they largely eliminated human operators, who were expensive and prone to make errors. So many were installed and proved so durable that as late as 1986 approximately 38 percent of all U.S. circuit switches were based on S × S. As they clearly are slow, expensive, large, maintenance intensive, and capacity limited [3], S × S switches are rarely found in the networks of developed countries. Large numbers of S × S switches do remain in service, however, mostly in developing countries.

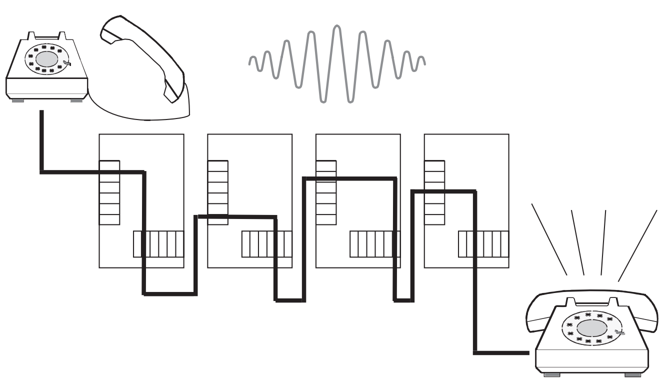

Figure 1.16 Step-by-step switching. (*Source:* AG Communications Systems Corporation.)

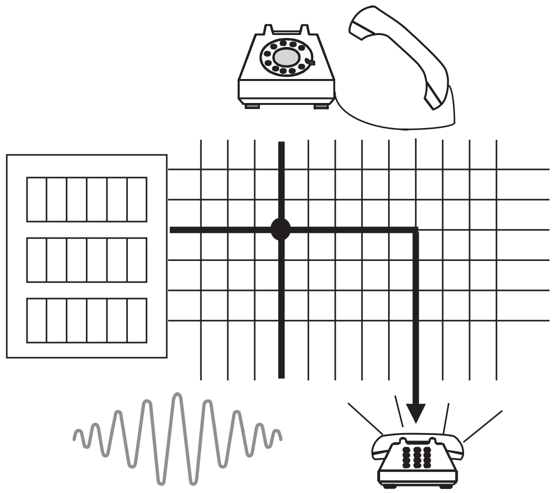

Figure 1.17 XBar switching. (*Source:* AG Communications Systems Corporation.)

1.9.1.3 Crossbar Switches The first *common control* switches, CrossBar(XBar) switches are electromagnetic in nature. While the original concept was developed at Bell Telephone Laboratories, the Ericsson company in Sweden accomplished much of the early practical development work. The first such switch installed in the United States was a central office exchange in Brooklyn, New York (1938) [6]. XBar switches quickly became predominant.

In an XBar switch (Figure 1.17), a request for dial tone is recognized by a *marker*, which directs a *sender* to store the dialed digits. A *translator* is then directed to route the call, reserving a path through a *switching matrix* [3]. Once the call connects, these various components become available to serve other calls. Compared to the S × S switch, the XBar has relatively few moving parts. XBar switches offer the advantages of increased intelligence, common control, greater speed of connection, smaller physical footprint, lower maintenance, and greater capacity. XBar switches were considered state of the art for nearly 30 years.

1.9.1.4 Electronic Common Control Switches Electronic Common Contral (ECC) switches reflect the marriage of computer technology and telephony. While the first ECC switches were analog, contemporary switches are fully digital in nature. Voice conversations are digitized and switched over high-speed digital circuits, with all processes accomplished through programmed logic. ECC switches are microprocessor controlled. The first ECC switch was the Electronic Switching System (ESS), developed by AT&T Bell Telephone Laboratories (Bell Labs) with the assistance of Western Electric. It was based on the transistor, invented at Bell Labs in 1948, and involved a development effort that began in earnest in the early 1950s. The first ESS CO began service in Succasunna, New Jersey, on May 30, 1965, connecting 200 subscribers. By 1974, there were 475 such offices in service, serving 5.6 million subscribers. The development effort was estimated to involve 4000 man-years and a total cost of US$500 million [7].

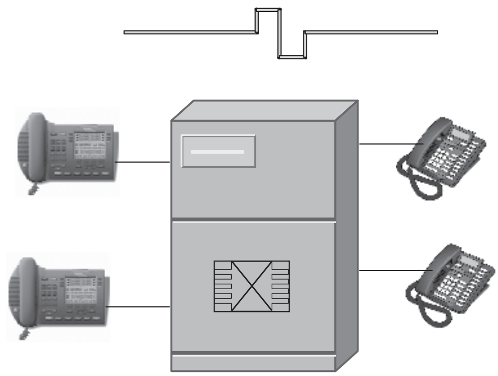

Figure 1.18 ECC switching.

ECC switches (Figure 1.18), as compared to the previous generations of switching technology, offer the advantages of further increased intelligence, greater speed of call setup and overall call processing, and a still smaller footprint. Additionally, they offer lower maintenance costs and can be monitored and managed from a remote location. Many contemporary ECC switches are unmanned, in favor of control from a centralized *Network Operations Center* (NOC). ECC switches offer greater capacity and on a scalable basis; that is, capacity can be increased through the addition of various system modules, or cabinets, with a reasonably graceful relationship maintained between increases in capacity and associated costs. As specialized, software-controlled computer systems, the functionality and feature content of ECC switches often can be upgraded through additional software and/or firmware. Such switches generally possess the ability to switch data and video as well as voice. Finally, they interface with various application processors to further increase the range of services provided. Example application processors include voice processors (e.g., voicemail) and fax servers (fax mail).

1.9.2 Packet Switching: Optimized for Data

First deployed in 1971, *packet switching* grew out of the U.S. Advanced Research Project Agency (ARPA) network. Commonly referred to as *ARPANET*, the network was established to support interactive, asynchronous computer-to-computer communications between the defense and university communities.

Rather than employing circuit switching, which is far too inefficient and expensive for intensive, interactive computer communications, ARPANET and its successors such as the Internet make use of packet switching. Packet switching involves the transmission of data formed into *packets* and sent across a shared network. Each packet, or datagram, is individually addressed in order that the packet switches can route each packet over the most appropriate and available circuit and each packet can survive independently. Each packet may represent an individual set of data or a larger set of data can be fragmented into multiple packets, each of which works its way through the network independently. A packet also can comprise multiple smaller, related sets of data gathered together for transport. Packets offered to the network by large numbers of users make use of the same universe of switches and transmission facilities. Such a highly shared network offers dramatically lower costs of data transmission in comparison to circuit switching.

Traditional packet-switched networks are based on mature and stable technologies, are widely available both domestically and internationally, and are low in cost. *X.25*, an international standard packet-switching interface (Chapter 7), is an excellent example of an early packet-switched network. X.25 offered great advantage in terms of its ability to support the interconnection of virtually any computer system through its ability to accomplish protocol conversion. This highly desirable feature caused X.25-based packet networks to be characterized as the first *Value-Added Networks* (*VANs*). X.25 remains heavily used, especially in developing countries. Disadvantages of X.25 include the fact that it can support only relatively low speed data transmission. Also, as the switches assume a 1960s-vintage analog network environment of twisted pair, each X.25 switch must examine each individual packet for errors created in transmission. In the event that an error is discovered, the switch must resolve the problem through a request for retransmission. These factors, in combination, result in unpredictable, variable levels of packet *latency* (delay). Therefore, packet switching traditionally has been considered to be unsuitable for stream-oriented communications such as real-time voice and video.

Most people, however, immediately think of *Transmission Control Protocol/Internet Protocol* (TCP/IP) when they think of packet switching. TCP/IP is not only fundamental to the Internet but also serves as the basis for a number of developing high-speed packet networks optimized for voice, currently challenging the traditional circuit-switched PSTN for dominance. These networks, which also are challenging Frame Relay and ATM, are based on the IP suite and various fiber-optic transmission technologies. (*Note*: IP is more prevalent in access networks, while ATM is more prevalent in carrier backbones.) Chapter 12 includes detailed discussion of the IP protocol suite, and Chapter 14 addresses IP-based competitive networks.

1.9.3 Frame Switching: Optimized for LAN Internetworking

A relative newcomer, *Frame Relay* was first offered commercially in 1992 by Wiltel (United States). Essentially a high-speed, streamlined version of X.25, Frame Relay was developed specifically as a protocol for access to a packet network in support of LAN-to-LAN networking. But Frame Relay can be used in support of the transmission of virtually any computer data stream in its native form, with individual *frames* varying in length up to 4096 bytes. Frame Relay became the overwhelming choice for data networking during most of the 1990s and currently is widely available as a domestic service offering in developed nations and on an international basis between them. As Frame Relay, like packet switching, is oriented toward data transmission over a highly shared network, frame latency is variable and unpredictable in duration. While increasingly satisfactory techniques have surfaced for support of voice and video, Frame Relay was not designed with those applications in mind. Detailed discussion of Frame Relay will be provided in Chapter 10.

1.9.4 Cell Switching: Optimized for Everything

Cell switching encompasses both *Switched Multimegabit Data Service* (SMDS) and ATM. Data are organized into *cells* of fixed length (53 octets), shipped across high-speed facilities, and switched through high-speed, specialized switches. SMDS proved very effective for data networking in a metropolitan area, although it was available only in limited serving areas and its popularity was short-lived. At this point, SMDS is but a historical footnote in the evolution of data networking.

ATM was much more successful than SMDS. ATM's value is in its ability to support any type of data stream (e.g., voice, data, video, image, and multimedia), whether real time or non–real time, compressed or uncompressed, and providing each with the appropriate level of Quality of Service (QoS) on a guaranteed basis. ATM is unique in this respect. Chapter 10 includes a detailed discussion of ATM.

1.9.5 Softswitches: Optimized for Flexibility

S*oftswitches* were developed in the late 1990s as a replacement for the *hard-coded* proprietary CO circuit switches used in the traditional PSTN. As packet-based IP, Frame Relay, and ATM networks began to develop in the 1990s to address the explosive growth in data traffic, it became clear that COs were functionally obsolete in such networks. Further, service providers had a strong interest in developing and delivering a wide variety of data and multimedia services in very short periods of time. Softswitches addressed those issues by offloading call processing functions (i.e., signaling and call control) to industry standard server hardware, essentially decomposing the call control logic from the switching platform. This allows the call control logic to reside at some geographically centralized location from which it can control multiple switching platforms, as well as the separate devices that provide for interconnection of circuit and packet networks such as IP and Frame Relay. In addition to controlling this protocol conversion function, softswitches can support multiple QoS and Grade of Service (GoS) mechanisms and levels. Softswitches are software programmable elements that support open *Application Programming Interfaces* (APIs) in consideration of the pressure to shorten the *time to market* for new services. Softswitches tend to be much more flexible and much less expensive and have much smaller footprints than traditional COs.

1.9.6 Photonic Switches: Optimized for Optics

Currently in the early stages of commercial application, *photonic switches* are yet another dimension in the evolution of switch technology. Photonic switches operate at the pure optical level, switching individual lambdas in a DWDM transmission system. Photonic switches do so without the requirement for optoelectric conversion imposed by more traditional electronic common control switching devices with optical interfaces but electronic switching fabrics, thereby offering clear advantages in terms of simplicity, speed, and cost. They are, however, limited to switching at the lambda level. (See Chapter 9.)

1.10 SIGNALING AND CONTROL

Signaling and control comprise a set of functions that must take place within any network to ensure that it operates smoothly. In this context, various elements within the network must identify themselves, communicate their status, and pass instructions. Fundamental examples include on-hook and off-hook indication, dial tone provision, call routing control, busy indication, and billing instructions. Further examples include dialed digits, route availability, routing preference, carrier preference, and originating number or circuit [3].

In more sophisticated, contemporary networks, the responsibility for overall signaling and control functions resides within a separate *Common Channel Signaling* (CCS) and control network. Such a sophisticated CCS network involves highly intelligent devices capable of monitoring and managing large numbers of lower order devices in the communications network that it controls. From a centralized *Network Control Center* (NCC), the network can be monitored, and faults or performance failures can be identified, diagnosed, and isolated. Finally, the lower order devices in the communications network often can be addressed and commanded to correct the condition. Contemporary signaling and control systems include *Signaling System 7* (SS7) in TDM networks and *H.323* and *Session Initiation Protocol* (SIP) in VoIP networks. Chapter 5 explores signaling and control in detail.

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