

Telecommunications Technology Fundamentals



Date: Dec 28, 2001

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This chapter talks about the types of transmission lines and network connections, the electromagnetic spectrum, and what bandwidth is all about in this emerging broadband era. It looks at the differences between analog and digital signals, and it discusses multiplexing. Finally, this chapter describes the various standards bodies and their roles in shaping aspects of telecommunications.

Transmission Lines

Two prerequisites must be satisfied to have successful communication. The first prerequisite is understandability. The transmitter and receiver must speak the same language. It doesn't matter how big or how clean a pipe you have between the two endpoints. If they're not speaking the same language, you will not be able to understand the message. In the case of data communications, we've resolved these issues quite elegantly: We have software and hardware translation devices that can convert between the different languages that individual computing systems speak. In the realm of human communications, we're about to embark on that exciting journey as well. Through the use of advanced voice-processing systems, in the next five to seven years we should have the ability to do real-time foreign language translation as part of the network service.

The second prerequisite is the capability to detect errors as they occur and to have some procedure for resolving those errors. In the case of human communications, intelligent terminals at either end—human beings—can detect noise that may have affected a transmission and request a retransmission, thereby correcting for that error. In the case of data devices, similar logic has to be built in to end devices so that they can detect errors and request a retransmission in order to correct for the errors.

If these two prerequisites—understandability and error control—are met, then communication can occur. We communicate by using data devices over what is generically termed a *transmission line*. There are five main types of transmission lines—circuits, channels, lines, trunks, and virtual circuits—each of which has a specific meaning. The following sections describe each of these types of transmission lines in detail.

Circuits

A *circuit* is the physical path that runs between two or more points. It terminates on a *port* (that is, a point of electrical or optical interface), and that port can be in a host computer (that is, a switching device used to establish connections), on a multiplexer, or in another device, as discussed later in this chapter.

In and of itself, a circuit does not define the number of simultaneous conversations that can be carried; that is a function of the type of circuit it is. For example, a simple, traditional telephone circuit is designed to carry just one conversation over one physical pathway. However, converting that to a digital circuit gives you the ability to extract or derive multiple channels over that circuit, subsequently facilitating multiple simultaneous conversations. So, the circuit is the measure of the physical entity.

There are two types of circuits: two-wire circuits and four-wire circuits.

Two-Wire Circuits

A two-wire circuit has two insulated electrical conductors. One wire is used for transmission of the information. The other wire acts as the return path to complete the electrical circuit. Two-wire circuits are generally deployed in the analog local loop, which is the last mile between the subscriber and the subscriber's first point of access into the network. [Figure 2.1](#) shows an example of a two-wire circuit.

Two-Wire and Four-Wire Versus Two-Pair and Four-Pair

Don't confuse the terms two-wire circuit and four-wire circuit with the terms two-pair and four-pair. Two-pair and four-pair refer to the number of wires in the internal cabling plan. Two-wire and four-wire have to do with the number of electrical conductors associated with a transmission line.

Figure 2.1 A two-wire circuit

Figure 2.2 A four-wire circuit

Four-Wire Circuits

A four-wire circuit has two pairs of conductors. That is, it has two sets of one-way transmission paths: one path for each direction and a complementary path to complete the electrical circuit (see [Figure 2.2](#)). Four-wire circuits are used where there is distance between the termination points which requires that the signal be strengthened periodically. So, for example, four-wire circuits connect the various switches that make up the public switched telephone network (PSTN). Four-wire circuits are also used with leased lines, where a customer may be connecting locations of its own that are separated by distance. Also, all digital circuits are provisioned on a four-wire basis.

There are two types of four-wire circuits: physical four-wire and logical four-wire. In *physical four-wire* you can actually count four wires. In *logical four-wire*, physically there are only two wires, but you derive the four individual paths by splitting the frequency. Half of the frequency band carries the transmit signal, and the other half carries the receive signal. So you can't always tell just by looking what kind of circuit you're dealing with; the application dictates the type of circuit it is.

Using Two-Wire and Four-Wire Circuits

Whenever you release energy into space, it loses power as it's traveling over a distance. So, because networks were designed to carry communications over a distance, we need tools to augment signals that have been losing power as they have traveled across the network, which are called *attenuated signals*. These tools are called amplifiers and repeaters. An *amplifier* boosts an attenuated signal back up to its original power level so it can continue to make its way across the network. The PSTN traditionally used copper wires. Based on how quickly the signals flow through the copper wires, there's a certain distance requirement between amplifiers. The distance requirement between amplifiers is relatively short on copper wires—generally about 6,000 feet (1,800 meters). As networks were built, these distance considerations were kept in mind. (Repeaters are discussed later in this chapter, in the section "Digital Transmission.")

Network builders had to give some thought to another aspect of amplifiers: First-generation amplifiers were unidirectional. They could only amplify a signal moving in one direction, so any time you needed to provision a circuit that was going to be crossing a distance, you had to literally provision two circuits—one to amplify the information in the transmit direction and a second to amplify the information in the receive direction. Therefore, whenever a network was crossing a distance, it needed to use a four-wire circuit. But in building out the millions of local loops for subscribers, it was seen as being cost-effective to have to pull only two wires into every home rather than four. Therefore, the local loops were intentionally engineered to be very short; some 70% to 80% of the local loops worldwide are less than 2 miles (3.2 kilometers) long. Because the local loops are short, they don't need amplifiers, and therefore the subscriber access service can be provisioned over a two-wire circuit. However, the local loop is increasingly being digitalized, so as we migrate to an end-to-end digital environment, everything becomes four-wire. [Figure 2.3](#) shows an example of a segment of a network in which two- and four-wire circuits are traditionally used.

Figure 2.3 Using two-wire and four-wire circuits

Channels

A *channel* defines a logical conversation path. It is the frequency band, time slot, or wavelength (also referred to as *lambda*) over which a single conversation flows. A channel is a child of the digital age because digital facilities enable multiple channels. The number of channels on a transmission line determines the number of simultaneous conversations that can be supported. Because we are becoming more digitalized all the time, you often hear people refer to the number of channels rather than the number of circuits.

Lines and Trunks

Lines and trunks are basically the same thing, but they're used in different situations. A *line* is a connection that is configured to support a normal calling load generated by one individual. A *trunk* is a circuit that is configured to support the calling loads generated by a group of users; it is the transmission facility that ties together switching systems. A *switching system* is a device that connects two transmission lines together. There are two major categories of switching systems:

- **CPE switches**—The most prevalent form of switch in the customer premises equipment (CPE) environment is the private branch exchange (PBX), which is called a private automatic branch exchange (PABX) in some parts of the world. A PBX is used to establish connections between two points. It establishes connections between telephones that are internal to the organization, and it establishes connections between internal extensions and the outside world (that is, the PSTN).
- **Network switches**—A hierarchy of network switches has evolved over time, and the appropriate switch is called into action, depending on which two points the switches are connecting together. For example, in [Figure 2.4](#) the CPE is on the left-hand side. Each individual single-line instrument represents a subscriber line. (Again, the fact that it's called a *line* means that it's a circuit configured to carry the calling load of just one user.) Above the single-line instrument is a business enterprise with a PBX. The connection from this PBX to the PSTN occurs over a trunk that is specifically configured to carry the calling load of multiple users. Beyond the PBX are multiple end users that are attached to that PBX. Each end user's connection would be referred to as a *station line*, again emphasizing that the line is carrying the calling load of one user.

Figure 2.4 Lines, trunks, and switches

The customer environment attaches to the PSTN, and the first point of access is the *local exchange*, which is also referred to as a *Class 5 office* (and in North America, as a *central office*). The traditional local exchange switch can handle one or more exchanges, with each exchange capable of handling up to 10,000 subscriber lines, numbered 0000 to 9999. The only kind of call that a local exchange can complete on its own, without touching any of the other switches in the network, is to another number in that same local exchange. Local exchanges are discussed in detail in Chapter 5, "The PSTN."

For a local exchange to call a neighbor that resides 10 miles (16 kilometers) away and who draws a dial tone from a different local exchange, the connection between those two different exchanges is accomplished through the second part of the hierarchy—a *tandem switch* (also called a *junction exchange*). The tandem switch is used to connect local exchanges throughout the metropolitan area. When it's time to make a toll call, one that is long-distance in nature, another switching center is called into action—the *toll center* (also called the *Class 4 office*, *transit switch*, or *trunk exchange*). The toll center is responsible for establishing and completing national, long-distance communications.

The top of the hierarchy is the *international gateway*, whose exchanges are specifically designed to connect calls between different countries.

A trunk supplies the connections between the numerous switches within the PSTN, between customer-owned switches such as the PBX, and between the PBXs and the PSTN. On the other hand, a line supports a single user in the form of a subscriber line in the PSTN or an extension provisioned from the PBX. (Chapter 5 describes in detail the entities involved in managing local, tandem, and toll exchanges.)

Virtual Circuits

Today, because of the great interest in and increased use of packet switching, most networks use virtual circuits. Unlike a physical circuit, which terminates on specific physical ports, a *virtual circuit* is a series of logical connections between sending and receiving devices (see [Figure 2.5](#)). The virtual circuit is a connection between two devices that acts as though it's a direct connection, but it may, in fact, be composed of a variety of different routes. These connections are defined by table entries inside the switch. A connection is established after both devices exchange agreement on communications parameters that are important to establishing and maintaining the connection and on providing the proper performance for the application they are supporting. The types of communication parameters that could be included are message size, the path to be taken, how to deal with acknowledgements in the event of errors, flow-control procedures, and error-control procedures. The term *virtual circuit* is largely used to describe connections between two hosts in a packet-switching network, where the two hosts can communicate as though they have a dedicated connection, although the packets may be taking very different routes to arrive at their destination.

Figure 2.5 A virtual circuit

There are two types of virtual circuits: permanent virtual circuits (PVCs) and switched virtual circuits (SVCs). The vast majority of implementations today involve PVCs. PVCs and SVCs are commonly used in packet-switching networks (for example, X.25, Frame Relay, ATM).

PVCs

A *PVC* is a virtual circuit that is permanently available; that is, the connection always exists between the two locations or two devices in question. A PVC is manually configured by a network management system, and it remains in place until the user reconfigures the network. Its use is analogous to the use of a dedicated private line because it provides an always-on condition between two locations or two devices.

SVCs

In contrast to PVCs, SVCs are set up on demand. They are provisioned dynamically by using signaling techniques. An SVC must be reestablished each time data is to be sent, and after the data has been sent, the SVC disappears. An SVC is therefore analogous to a dialup connection in the PSTN. The main benefit of an SVC is that you can use it to access the network from anyplace. The predominant application for SVCs is to accommodate people who are working at home, in a hotel, at an airport, or otherwise outside the physical location of the enterprise network.

Types of Network Connections

Three major types of networks connections can be made:

- **Switched network connections**—A switched connection is referred to as a *dialup* connection. This implies that you're using a series of network switches to establish the connection between the parties.
- **Leased-line network connections**—A leased line is also referred to as a private line. With a leased line, the same locations or the same devices are always connected, and transmission between those locations or devices always occurs on the same path.
- **Dedicated network connections**—In essence, a dedicated line works exactly like a leased line. It is always connected and it always uses the same path for transmission. However, the end user may own the transmission facility (rather than lease it) such that it is exclusive to that user.

The Electromagnetic Spectrum and Bandwidth

This section talks about bandwidth and about where the various transmission media lie within the electromagnetic spectrum.

The Electromagnetic Spectrum

When electrons move, they create electromagnetic waves that can propagate through free space. This phenomenon was first predicted to exist by James Maxwell, in 1865, and it was first produced and observed by Heinrich Hertz in 1887. All modern communication depends on manipulating and controlling signals within the electromagnetic spectrum.

The electromagnetic spectrum ranges from extremely low-frequency radio waves of 30Hz, with wavelengths of nearly the earth's diameter, to high-frequency cosmic rays of more than 10 million trillion Hz, with wavelengths smaller than the nucleus of an atom. The electromagnetic spectrum is depicted as a logarithmic progression: The scale increases by multiples of 10, so the higher regions encompass a greater span of frequencies than do the lower regions.

Although the electromagnetic spectrum represents an enormous range of frequencies, not all the frequencies are suitable to purposes of human communications. At the very low end of the spectrum are signals that would be traveling at 30Hz (that is, at 30 cycles per second). One of the benefits of a very low frequency is that it can travel much farther than a high frequency before it loses power (that is, attenuates). So a 30Hz signal provides the benefit of being able to travel halfway around the world before it requires some form of amplification. For example, one defense agency uses 30Hz to communicate with its submarines by using telemetry (for example, a message that says "We're still here. We're still here" is sent, and the subs know that if they don't get that message, they better see what's going on). Again, the benefit of very-low-frequency signals is that they can travel a very long distance before they attenuate.

At the high end of the electromagnetic spectrum, signals travel over a band of 10 million trillion Hz (that is, 10^{22} Hz). This end of the spectrum has phenomenal bandwidth, but it has its own set of problems. The wave forms are so miniscule that they're highly distorted by any type of interference, particularly environmental interference such as precipitation. Furthermore, higher-frequency wave forms such as x-rays, gamma rays, and cosmic rays are not very good to human physiology and therefore aren't available for us to use for communication at this point.

Infrasound and the Animal World

The universe is full of *infrasound*—the frequencies below the range of human hearing. Earthquakes, wind, thunder, volcanoes, and ocean storms—massive movements of earth, air, fire, and water—generate infrasound. In the past, very-low-frequency sound has not been thought to play much of a role in animals' lives. However, we know now that sound at the lowest frequencies of elephant rumbles (14Hz to 35Hz) has remarkable properties. It is little affected by passage through forests and grasslands, and male and female elephants use it to find one another for reproduction. It seems that elephants communicate with one another by using calls that are too low-pitched for human beings to hear, and because of the properties of the infrasound range, these communications can take place over very long distances. Intense infrasonic calls have also been recorded from finback whales.

Because of the problems with very low and very high frequencies, we primarily use the middle of the electromagnetic spectrum for communication—the radio, microwave, infrared, and visible light portions of the spectrum. We do this by modulating the amplitudes, the frequencies, and the phases of the electromagnetic waves. Bandwidth is actually a measure of the difference between the lowest and highest frequencies being carried. Each of these communications bands offers differing amounts of bandwidth, based on the range of frequencies they cover. The higher up in the spectrum you go, the greater the range of frequencies involved.

Figure 2.6 shows the electromagnetic spectrum and where some of the various transmission media operate. Along the right-hand side is the terminology that the International Telecommunication Union (ITU) applies to the various bands: Extremely low, very low, low, medium, high, very high (VHF), ultrahigh (UHF), superhigh (SHF), extremely high (EHF), and tremendously high frequencies (THF) are all various forms of radio bands. And then we move into the light range, with infrared and visible light. You can see just by the placement of the various transmission media that not all are prepared to face the high-bandwidth future that demanding advanced applications (such as streaming media, e-learning, networked interactive games, interactive TV, telemedicine, metacomputing, and Web agents) will require.

Figure 2.6 The electromagnetic spectrum

The radio, microwave, infrared, and visible light portions of the spectrum can all be used for transmitting information by modulating various measurements related to electromagnetic waves (see Figure 2.7):

Figure 2.7 An electromagnetic wave

- **Frequency**—The number of oscillations per second of an electromagnetic wave is called its *frequency*.
- **Hertz**—Frequency is measured in *Hertz (Hz)*, in honor of Heinrich Hertz.
- **Wavelength**—The *wavelength* is the distance between two consecutive maxima or minima of the wave form.
- **Amplitude**—*Amplitude* is a measure of the height of the wave, which indicates the strength of the signal.
- **Phase**—*Phase* refers to the angle of the wave form at any given moment.
- **Bandwidth**—The range of frequencies (that is, the difference between the lowest and highest frequencies carried) that make up a signal is called *bandwidth*.

You can manipulate frequency, amplitude, and phase in order to distinguish between a one and a zero. Hence, you can represent digital information over the electromagnetic spectrum. One way to manipulate frequency is by sending ones at a high frequency and zeros at a low frequency. Devices that do this are called *frequency-modulated devices*. You can also modulate amplitude by sending ones at a high amplitude or voltage and zeros at a low amplitude. A complementary receiving device could then determine whether a one or a zero is being sent. As yet another example, because the phase of the wave form refers to shifting where the signal begins, you could have ones begin at 90 degrees and zeros begin at 270 degrees. The receiving device could discriminate between these two bit states (zero versus one) based on the phase of the wave as compared to a reference wave.

Twisted-pair, which was the original foundation of the telecommunications network, has a maximum usable bandwidth of about 1MHz. Coax, on the other hand, has greater capacity, offering a total of 1GHz of frequency spectrum. The radio range, particularly microwave, is the workhorse of the radio spectrum. It gives us 100GHz to operate with. In comparison, fiber optics operates over a band of more than 200THz (terahertz). So, as we see increasingly more bandwidth-hungry applications, we'll need to use fiber optics to carry the amount of traffic those applications generate. Twisted-pair will see little use with the future application set. [Figure 2.8](#) plots various telecommunications devices on the electromagnetic spectrum.

[Figure 2.8](#) Telecommunications devices and the electromagnetic spectrum

Bandwidth

As mentioned earlier, *bandwidth* is the range of frequencies that make up a signal. There are three major classes of bandwidth that we refer to in telecommunications networks: narrowband, wideband, and broadband.

Narrowband

Narrowband means that you can accommodate up to 64Kbps, which is also known as the DS-0 (Digital Signal level 0) channel. This is the fundamental increment on which digital networks were built. Initially, this metric of 64Kbps was derived based on our understanding of what it would take to carry voice in a digital manner through the network. If we combine these 64Kbps channels together, we can achieve wideband transmission rates.

Wideband

Wideband is defined as being $n \times 64\text{Kbps}$, up to approximately 45Mbps. A range of services are provisioned to support wideband capabilities, including T-carrier, E-carrier, and J-carrier services. These are the services on which the first generation of digital hierarchy was built.

T-1 offers 1.544Mbps, and because the T-carrier system is a North American standard, T-1 is used in the United States. It is also used in some overseas territories, such as South Korea and Hong Kong. E-1, which provides a total of 2.048Mbps, is specified by the ITU. It is the international standard used throughout Europe, Africa, most of Asia-Pacific, the Middle East, and Latin America. J-carrier is the Japanese standard, and J-1 offers 1.544Mbps.

Not every office or application requires the total capacity of T-1, E-1, or J-1, so you can subscribe to *fractional services*, which means you subscribe to bundles of channels that offer less than the full rate. Fractional services are normally provided in bundles of 4, so you can subscribe to 4 channels, 8 channels, 12 channels, and so on. Fractional services are also referred to as $n \times$ 56Kbps/64Kbps in the T-carrier system and $n \times$ 64Kbps under E-carrier. High-bandwidth facilities include T-3, E-3, and J-3. T-3 offers 45Mbps, E-3 offers 34Mbps, and J-3 supports 32Mbps. (T-, E-, and J-carrier services are discussed in more detail in Chapter 5.)

Broadband

The future hierarchy, of course, rests on broadband capacities, and *broadband* can be defined in different ways, depending on what part of the industry you're talking about. Technically speaking, the ITU has defined broadband as being anything over 2Mbps. But this definition was created in the 1970s, when 2Mbps seemed like a remarkable capacity.

The Impact of Fiber Optics on Bandwidth

So far this chapter has used a lot of bits-per-second measurements. It can be difficult to grasp what these measurements really mean. So, here's a real-world example. Today, fiber optics very easily accommodates 10Gbps (that is, 10 billion bits per second). But what does that really mean? At 10Gbps you'd be able to transmit all 32 volumes of the *Encyclopedia Britannica* in 1/10 second—the blink of an eye. That is an incredible speed. Not many people have a computer capable of capturing 10Gbps.

Keep in mind that underlying all the various changes in telecommunications technologies and infrastructures, a larger shift is also occurring—the shift from the electronic to the optical, or photonic, era. To extract and make use of the inherent capacity that fiber optics affords, we will need an entire new generation of devices that are optical at heart. Otherwise, we'll need to stop a signal, convert it back into an electrical form to process it through the network node, and then convert it back into optics to pass it along, and this will not allow us to exercise the high data rates that we're beginning to envision.

Given today's environment, for wireline facilities, it may be more appropriate to think of broadband as starting where the optical network infrastructure starts. Synchronous Digital Hierarchy (SDH) and Synchronous Optical Network (SONET) are part of the second generation of digital hierarchy, which is based on fiber optics as the physical infrastructure. (SDH and SONET are discussed in detail in Chapter 5.) The starting rate (that is, the lowest data rate supported) on SDH/SONET is roughly 51Mbps. So, for the wireline technologies—those used in the core or backbone network—51Mbps is considered the starting point for broadband. In the wireless realm, though, if we could get 2Mbps to a handheld today, we'd be extremely happy and would be willing to call it broadband. So, remember that the definition of broadband really depends on the situation. But we can pretty easily say that broadband is always a multichannel facility that affords higher capacities than the traditional voice channel, and in the local loop, 2Mbps is a major improvement.

Analog and Digital Transmission

There are a number of differences between analog and digital transmission, and it is important to understand how conversions between analog and digital occur. Let's look first at the older form of transmission, analog.

Analog Transmission

An analog wave form (or signal) is characterized by being continuously variable along amplitude and frequency. In the case of telephony, for instance, when you speak into a handset, there are changes in the air pressure around your mouth. Those changes in air pressure fall onto the handset, where they are amplified and then converted into current, or voltage fluctuations. Those fluctuations in current are an analog of the actual voice pattern—hence the use of the term *analog* to describe these signals (see [Figure 2.9](#)).

Figure 2.9 Analog transmission

When it comes to an analog circuit—what we also refer to as a voice-grade line—we need to also define the frequency band in which it operates. The human voice, for example, can typically generate frequencies from 100Hz to 10,000Hz, for a bandwidth of 9,900Hz. But the ear does not require a vast range of frequencies to elicit meaning from ordinary speech; the vast majority of sounds we make that constitute intelligible speech fall

between 250Hz and 3,400Hz. So, the phone company typically allotted a total bandwidth of 4,000Hz for voice transmission. Remember that the total frequency spectrum of twisted-pair is 1MHz. To provision a voice-grade analog circuit, bandwidth-limiting filters are put on that circuit to filter out all frequencies above 4,000Hz. That's why analog circuits can conduct only fairly low-speed data communications. The maximum data rate over an analog facility is 33.6Kbps when there are analog loops at either end.

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How 56Kbps Modems Break the 33.6Kbps Barrier

With 56Kbps modems, only one end of the loop can be analog. The other end of the connection has to be digital. So, in other words, if you're using a 56Kbps modem to access your Internet service provider (ISP), you have an analog connection from your home to the local exchange. But the ISP has a digital subscriber line (DSL) or a digital termination facility from its location to its exchange.

Analog facilities have limited bandwidth, which means they cannot support high-speed data. Another characteristic of analog is that noise is accumulated as the signal traverses the network. As the signal moves across the distance, it loses power and becomes impaired by factors such as moisture in the cable, dirt on a contact, and critters chewing on the cable somewhere in the network. By the time the signal arrives at the amplifier, it is not only attenuated, it is also impaired and noisy. One of the problems with a basic amplifier is that it is a dumb device. All it knows how to do is to add power, so it takes a weak and impaired signal, adds power to it, and brings it back up to its original power level. But along with an increased signal, the amplifier passes along an increased noise level. So in an analog network, each time a signal goes through an amplifier, it accumulates noise. After you mix together coffee and cream, you can no longer separate them. The same concept applies in analog networks: After you mix the signal and the noise, you can no longer separate the two, and, as a result, you end up with very high error rates.

Digital Transmission

Digital transmission is quite different from analog transmission. For one thing, the signal is much simpler. Rather than being a continuously variable wave form, it is a series of discrete pulses, representing one bits and zero bits (see [Figure 2.10](#)). Each computer uses a coding scheme that defines what combinations of ones and zeros constitute all the characters in a character set (that is, lowercase letters, uppercase letters, punctuation marks, digits, keyboard control functions).

Figure 2.10 Digital transmission

How the ones and zeros are physically carried through the network depends on whether the network is electrical or optical. In electrical networks, one bits are represented as high voltage, and zero bits are represented as null, or low voltage. In optical networks, one bits are represented by the presence of light, and zero bits are represented by the absence of light. The ones and zeros—the on/off conditions—are carried through the network, and the receiving device repackages the ones and zeros to determine what character is being represented. Because a digital signal is easier to reproduce than an analog signal, we can treat it with a little less care in the network. Rather than use dumb amplifiers, digital networks use *regenerative repeaters*, also referred to as *signal regenerators*. As a strong, clean, digital pulse travels over a distance, it loses power, similar to an analog signal. The digital pulse, like an analog signal, is eroded by impairments in the network. But the weakened and impaired signal enters the regenerative repeater, where the repeater examines the signal to determine what was supposed to be a one and what was supposed to be a zero. The repeater regenerates a new signal to pass on to the next point in the network, in essence eliminating noise and thus vastly improving the error rate.

Analog Versus Digital Transmission

Table 2.1 summarizes the characteristics of analog and digital networks.

Table 2.1 Characteristics of Analog and Digital Networks

Feature	Analog Characteristics	Digital Characteristics
Signal	Continuously variable, in both amplitude and frequency	Discrete signal, represented as either changes in voltage or changes in light levels
Traffic measurement	Hz (for example, a telephone channel is 4KHz)	Bits per second (for example, a T-1 line carries 1.544Mbps, and an E-1 line transports 2.048Mbps)
Bandwidth	Low bandwidth (4KHz), which means low data transmission rates (up to 33.6Kbps) because of limited channel bandwidth	High bandwidth that can support high-speed data and emerging applications that involve video and multimedia
Network capacity	Low; one conversation per telephone channel	High; multiplexers enable multiple conversations to share a communications channel and hence to achieve greater transmission efficiencies
Network manageability	Poor; a lot of labor is needed for network maintenance and control because dumb analog devices do not provide management information streams that allow the device to be remotely managed	Good; smart devices produce alerts, alarms, traffic statistics, and performance measurements, and technicians at a network control center (NCC) or network operations center (NOC) can remotely monitor and manage the various network elements
Power requirement	High because the signal contains a wide range of frequencies and amplitudes	Low because only two discrete signals—the one and the zero—need to be transmitted
Security	Poor; when you tap into an analog circuit, you hear the voice stream in its native form, and it is	Good; encryption can be used

Feature	Analog Characteristics	Digital Characteristics
	difficult to detect an intrusion	
Error rates	High; 10^{-5} bits (that is, 1 in 100,000 bits) is guaranteed to have an error	Low; with twisted-pair, 10^{-7} (that, is 1 in 10 million bits per second) will have an error, with satellite, 10^{-9} (that is, 1 in 1 billion per second) will have an error, and with fiber, 10^{-11} (that is only 1 in 10 trillion bits per second) will have an error

Conversion: Codecs and Modems

The fact is that today we don't have all-digital or all-analog networks; we have a mix of the two. Therefore, at various points in a network, it is necessary to convert between the two signal types. The devices that handle these conversions are codecs and modems (see [Figure 2.11](#)).

Figure 2.11 Codecs and modems

A *codec* (which is a contraction of *coder-decoder*) converts analog signals into digital signals. There are different codecs for different purposes. For the PSTN, for example, there are codecs that minimize the number of bits per second required to carry voice digitally through the PSTN. In cellular networks, because of the constraints and available spectrum, a codec needs to compress the voice further, to get the most efficient use of the spectrum. Codecs applied to video communication also require very specific compression techniques to be able to move those high-bandwidth signals over what may be somewhat limited channels today.

A *modem* (which is a contraction of *modulator-demodulator*) is used to infuse digital data onto transmission facilities. Some modems are designed specifically to work with analog voice-grade lines. There are also modems that are designed to work specifically with digital facilities (for example, ISDN modems, ADSL modems). A modem manipulates the variables of the electromagnetic wave to differentiate between the ones and zeros.

Although it is possible to convert between analog and digital networks, in general, conversions are a weak link in a network. A conversion is a point at which network troubles can occur, an opportunity for errors and distortions to be introduced. Therefore, ideally, we want to move toward an end-to-end digital and end-to-end optical environment. This means that nowhere between the transmitter and the receiver do signal conversions need to be done.

Multiplexing

Multiplexers, often called *muxes*, are extremely important to telecommunications. Their main reason for being is to reduce network costs by minimizing the number of communications links needed between two points. As with all other computing systems, multiplexers have evolved. Each new generation has additional intelligence, and additional intelligence brings more benefits. The types of benefits that have accrued, for example, include the following:

- The capability to do data compression so that you can encode certain characters with fewer bits than normally required and free up that additional capacity for the movement of other information.
- The capability to do error detection and correction between the two points that are being connected to

ensure that data integrity and accuracy are being maintained.

- The capability to manage transmission resources on a dynamic basis, with such things as priority levels. If you have only one 64Kbps channel left, who gets it? Or what happens when the link between San Francisco and Hong Kong goes down? How else can you reroute traffic to get the high-priority information where it needs to go? Multiplexers help solve such problems.

The more intelligent the multiplexer, the more actively and intelligently it can work on your behalf to dynamically make use of the transmission resources you have.

When you're working with network design and telecommunications, you need to consider line cost versus device cost. You can provide extremely high levels of service by ensuring that everybody always has a live and available communications link. But you must pay for those services on an ongoing basis, and their costs become extremely high. You can offset the costs associated with providing large numbers of lines by instead using devices such as multiplexers that help you make more intelligent use of a smaller group of lines.

Figure 2.12 illustrates a network without multiplexers. Let's say this network is for Bob's department stores. The CPU is at Location A, a data center that's in New York that manages all the credit authorization functions for all the Bob's stores. Location B, the San Francisco area, has five different Bob's stores in different locations. Many customers will want to make purchases using their Bob's credit cards, so we need to have a communications link back to the New York credit authorization center so that the proper approvals and validations can be made. Given that it's a sales transaction, the most likely choice of communications link is the use of a leased line from each of the locations in San Francisco back to the main headquarters in New York.

Figure 2.12 A network without multiplexers

Remember that the use of leased lines is a very expensive type of network connection. Because this network resource has been reserved for one company's usage only, nobody else has access to that bandwidth, and providers can't make use of it in the evenings or the weekends to carry residential traffic, so the company pays a premium. Even though it is the most expensive approach to networking, the vast majority of data networking today still takes place using leased lines, because they make the network manager feel very much in control of the network's destiny. With leased lines, the bandwidth is not affected by sudden shifts of traffic elsewhere in the network, the company can apply its own sophisticated network management tools, and the network manager feels a sense of security in knowing who the user communities are at either end of that link. But leased lines have another negative attribute: They are mileage sensitive, so the longer the communications link, the higher the cost. And in a network that doesn't efficiently make use of that communications link all day long, leased lines become overkill and an expensive proposition.

The astute network manager at Bob's tries to think about ways to make the network less expensive. One solution, shown in Figure 2.13, is to put in multiplexers. Multiplexers always come in pairs, so if you have one at one end, you must have one at the other end. They are also symmetrical, so if there are five outputs available in San Francisco, there must also be five inputs in the New York location. The key savings in this scenario comes from using only one leased line between New York and California. In San Francisco, short leased lines, referred to as *tail circuits*, run from the centrally placed multiplexer to each of the individual locations. Thus, five locations are sharing one high-cost leased line, rather than each having its own leased line. Intelligence embedded in the multiplexers allows the network manager to manage access to that bandwidth and to allocate network services to the endpoints.

Figure 2.13 A network with multiplexers

Various techniques—including Frequency Division Multiplexing (FDM), Time Division Multiplexing (TDM), Statistical Time Division Multiplexing (STDM), intelligent multiplexing, inverse multiplexing, and Wavelength Division Multiplexing (WDM)/Dense Wavelength Division Multiplexing (DWDM)—enable multiple channels to coexist on one link. The following sections examine each of these techniques.

FDM

FDM is an environment in which the entire frequency band available on the communications link is divided into smaller individual bands or channels (see Figure 2.14). Each user is assigned to a different frequency. The signals all travel in parallel over the same communications link, but they are divided by frequency—that is, each signal rides on a different portion of the frequency spectrum. Frequency, which is an analog parameter, implies

that the type of link you see with FDM is usually an analog facility. A disadvantage of frequency division muxes is that they can be difficult to reconfigure in an environment in which there's a great deal of dynamic change. For instance, to increase the capacity of Channel 1 in [Figure 2.14](#), you would also have to tweak Channels 2, 3, and 4 to accommodate that change.

Figure 2.14 FDM

If an enterprise has a high degree of moves, additions, and changes, FDM would be an expensive system to maintain because it would require the additional expertise of frequency engineering and reconfiguration. Given the environment today, we don't make great use of FDM, but it is still used extensively in cable TV and in radio. In cable TV, multiple channels of programming all coexist on the coax coming into your home, and they are separated based on the frequency band in which they travel. When you enter a channel number on your set-top box or cable-ready TV, you're essentially indicating to the network what portion of the frequency band it's traveling on.

TDM

The second muxing technique to be delivered to the marketplace was TDM. There are various levels of TDM. In the plain-vanilla TDM model, as shown in [Figure 2.15](#), a dedicated time slot is provided for each port or point of interface on the system. Each device in a predetermined sequence is allotted a time slot during which it can transmit. That time slot would enable one character of data, or 8 bits of digitized voice, to be placed on the communications link. The allocated time slots have to be framed in order for the individual channels to be separated out. A problem with a standard time-division mux is that there is a one-to-one correlation between each port and time slot, so if the device attached to Port 2 is out for the day, nobody else can make use of Time Slot 2. Hence, there is a tendency to waste bandwidth when vacant slots occur because of idle stations. However, this type of TDM is more efficient than standard FDM because more subchannels can be derived.

Figure 2.15 TDM

FDM and TDM can be combined. For example, you could use FDM to carve out individual channels and then within each of those channels apply TDM to carry multiple conversations on each channel. In fact, this is the way that some digital cellular systems work (for example, Global Systems for Mobile Communications [GSM]). Digital cellular systems are discussed in Chapter 14, "Wireless Communications."

STDM

STDM was introduced to overcome the limitation of standard TDM, in which stations cannot use each other's time slots. Statistical time-division multiplexers, sometimes called *statistical muxes* or *stat muxes*, dynamically allocate the time slots among the active terminals, which means that you can actually have more terminals than you have time slots (see [Figure 2.16](#)).

Figure 2.16 STDM

A stat mux is a smarter mux and it has more memory than other muxes, so if all the time slots are busy, excess data goes into a buffer. If the buffer fills up, the additional access data gets lost, so it's important to think about how much traffic to put through the stat mux to ensure that performance variables are maintained. By dynamically allocating the time slots, you get the most efficient use of bandwidth. Additionally, because these are smarter muxes, they have the additional intelligence mentioned earlier in terms of compression and error-control features. Because of the dynamic allocation of time slots, a stat mux is able to carry two to five times more traffic than a traditional time-division mux. But, again, as you load the stat mux with traffic, you run the risk of delays and data loss occurring.

Stat muxes are extremely important because they are the basis on which packet-switching technologies (for example, X.25, IP, Frame Relay, ATM) are built. The main benefit of a stat mux is the efficient use of bandwidth, which leads to transmission efficiencies.

Intelligent Multiplexing

An intelligent multiplexer is often referred to as a *concentrator*, particularly in the telecom world. Rather than

being a device used in pairs, it is used as a singular device, a line-sharing device whose purpose is to concentrate large numbers of low-speed lines to be carried over a high-speed line to a further point in the network. A good example of a concentrator is in a device called the *digital loop carrier* (DLC), which is also referred to as a *remote concentrator* or *remote terminal*. In [Figure 2.17](#), twisted-pairs go from the local exchange to the neighborhood. Before the advent of DLCs, you needed a twisted-pair for each household. If the demand increased beyond the number of pairs you had available out of that local exchange, you were out of luck until a new local exchange was added.

Figure 2.17 Intelligent multiplexing: Concentrators

With digital technology, you can make better use of the existing pairs. Instead of using each pair individually per subscriber from the local exchange to the subscriber, you can put a DLC in the center. You use a series of either fiber-optic pairs or microwave beams to connect the local exchange to this intermediate DLC, and those facilities then carry multiplexed traffic. When you get to the DLC, you break out the individual twisted-pairs to the household. This allows you to eliminate much of what used to be an analog plant leading up to the local exchange. It also allows you to provide service to customers who are outside the distance specifications between a subscriber and the local exchange. So, in effect, that DLC can be used to reduce the loop length.

Traditional DLCs are not interoperable with some of the new DSL offerings, including ADSL and SDSL. For example, about 30% to 40% of the U.S. population is serviced through DLCs. And in general, globally, the more rural or remote a city or neighborhood, the more likely that it is serviced via a DLC. For those people to be able to subscribe to the new high-bandwidth DSL services, the carrier will have to replace the DLC with a newer generation of device. Lucent's xDSL Access Gateway, for example, is such a device; it offers a multiservice access system that provides Lite and full-rate ADSL, Integrated Services Digital Network (ISDN), Asynchronous Transfer Mode (ATM), and plain old telephone service (POTS)/analog over twisted-pair lines or fiber. Or, the carrier may simply determine that the market area doesn't promise enough revenue and leave cable modems or broadband wireless as the only available broadband access technique. But the nature of a concentrator is that it enables you to aggregate numerous low-speed residential lines and to multiplex them onto high-bandwidth facilities to pass off to the local exchange.

Inverse Multiplexing

The inverse multiplexer arrived on the scene in the 1990s. It does the opposite of what the multiplexers described so far do. Rather than combine lots of low-bit-rate streams to ride over a high-bit-rate pipe, an inverse multiplexer breaks down a high-bandwidth signal into a group of smaller-data-rate signals that can be dispersed over a range of channels to be carried over the network. A primary application for inverse multiplexers is to support of high-bandwidth applications such as videoconferencing.

In [Figure 2.18](#), a videoconference is to occur at 1.5Mbps. For a good-quality,

or PRI ISDN. (PRI ISDN is discussed in Chapter 3, "Transmission Media: Characteristics and Applications.") The channels are carried across the network separately. At the destination point, a complementary inverse mux again reaggregates, resynchronizes, and reproduces that high-bandwidth signal so that it can be projected on the destination video monitor.

Figure 2.18 Inverse multiplexing

Inverse multiplexing therefore allows you to experience a bit of elastic bandwidth. You can allocate existing capacity to a high-bandwidth application without having to subscribe to a separate link just for that purpose.

WDM/DWDM

WDM was specifically developed for use with fiber optics. In the past, we could use only a fraction of the available bandwidth of a fiber-optic system. This was mainly because we had to convert the optical pulses into electrical signals to regenerate them as they moved through the fiber network. And because repeaters were originally electronic, data rates were limited to about 2.5Gbps. In 1994, something very important happened: optical amplifiers called *erbium-doped fiber amplifiers* (EDFAs) were introduced. Erbium is a chemical that's injected into the fiber. As a light pulse passes through the erbium, the light is amplified and continues on its merry way, without having to be stopped and processed as an electrical signal. The introduction of EDFAs immediately opened up the opportunity to make use of fiber-optic systems operating at 10Gbps.

EDFAs also paved the way to developing wavelength division multiplexers. Before the advent of WDM, we were using only one wavelength of light within each fiber, whereas the visible light spectrum engages a large number of different wavelengths. WDM takes advantage of the fact that multiple colors or frequencies of light can be transmitted simultaneously down a single optical fiber. The data rate that's supported by each of the wavelengths depends on the type of light source. Today, we have light sources that operate at a rate of OC-48, which is shorthand for 2.5Gbps. We have light sources that operate at OC-192, which is equivalent to 10Gbps. And there are systems in trial that operate at OC-768, offering 40Gbps per wavelength. In the future, we'll go beyond that. Part of the evolution of WDM is that every year we double the number of bits per second that can be carried on a wavelength, and every year we double the number of wavelengths that can be carried over a single fiber. But we have just begun. Soon light sources should be able to pulse in the terabits per second range, and in five years, light sources should pulse in the petabits per second (1,000Tbps) range.

One thing to clarify about the first use of WDM is that unlike with the other types of multiplexing, where the goal is to aggregate smaller channels into one larger channel, WDM is meant to furnish separate channels for each service, at the full data rate. Increasingly, enterprises are making use of new high-capacity switches and routers that are equipped with 2.5Gbps interfaces, so there's a great deal of desire within the user community to be able to plug in to a channel of sufficient size to carry that high-bandwidth signal end-to-end, without having to break it down into smaller increments only to build them back out at the destination.

WDM furnishes a separate channel for each service at the full rate; you cannot aggregate smaller channels into one large channel. Systems that support more than 16 wavelengths are referred to as DWDM (see [Figure 2.19](#)). Systems at the OC-48

Figure 2.19 WDM and DWDM

Standards Organizations

Telecommunications is a major industry sector for two main reasons: First, it generates vast amounts of revenue every year, and second, it touches every aspect of life and business. Networking is an international phenomenon, and recommendations have to be made on how systems and networks should interoperate. Standardization within the industry is intended to perform three basic functions:

- Facilitate interconnection between different users.
- Facilitate the portability of equipment within different regions and applications, with the intent of increasing market size, resulting in reduced costs for all.
- Ensure equipment interoperability, so that different vendors' products work with each other.

Standards bodies that make such recommendations have traditionally been active in Europe, North America, and Japan. In recent years, the Internet Society (ISOC) has also become an increasingly important organization.

Standards Organization Time Line

The following time line shows some of the important dates in the history of standards organizations:

	1865	The ITU was established by 20 European states at the first International Telegraph Convention.
	1923	The CCIF was established in Paris, for the study of long-distance telephony.
	1925	The CCITT (now the ITU-T) was established for the technical study of telephony problems. The CCIF and CCITT both became part of the ITU.

	1927	The CCIR was formed in Washington, with the objective of concentrating on technical issues surrounding radio communications.
	1947	The ITU was recognized as an agency of the United Nations, specializing in telecommunications.
	1959	The CCIF and the CCITT were combined and became known simply as the CCITT, now called the ITU-T.

The ITU is a body of the United Nations that includes members from around the world. Three specific groups within the ITU are relevant to telecommunications. The ITU-T, the telecommunications standardization sector, develops recommendations for wireline networks. ITU-R, the radio communications standardization sector, deals with the wireless arena. The ITU-D sector works on standards for developing nations. The ITU standards are followed throughout most of the world, including Africa, most of Asia-Pacific, Europe, Latin America, and the Middle East.

The second major standards body is the American National Standards Institute (ANSI), whose recommendations are followed throughout North America, as well as in some Asian countries.

The third major body is the Telecommunications Technology Committee, which is followed in Japan.

ISOC has an agreement to work with the ITU to ensure that developments do not take place separately in evolving the PSTN versus the Internet. These organizations are working toward converging networks on the same sets of requirements and functions.

In order to make room for the many—often conflicting—interests, the international standards-making organizations concentrate on producing what are known as *base standards*. These base standards contain allowable variants or options, which are defined by the implementer. By adopting any of the variants, or options, the implementer is compliant with the standard, but there is no guarantee that the equipment will interoperate with the equipment of other vendors. This problem of interoperability and internetworking is addressed by regional and national standards bodies, often involving trade organizations and users groups. These groups adapt the international base standards as functional standards, which contain a limited amount of agreed-upon allowable options. These groups also develop test specifications and methods, and independent test houses then perform the necessary conformance testing and certify products that meet the requirements. (See [Figures 2.20](#) and [2.21](#).)

Figure 2.20 Standards-making groups

Figure 2.21 The standards-making process

Table 2.2 lists some of the key standards organizations throughout the world.

Table 2.2 Standards Organizations

Region	Standards Organizations
International	ITU, ITU-T, ITU-R, ITU-D, IEC, ISO

Region	Standards Organizations
Australia	ACA, ACC, AIIA, ATUG
Europe	AFNOR (France), CEN and CENLEC, CEPT, DIN (Germany), DTI (UK), ETSI, European Community (EU)
Japanese	JISC, TTC
New Zealand	ITANZ
North America	ANSI (USA), EIA, FCC (USA), IEEE, NIST, SCC (Canadian)

For more learning resources, quizzes, and discussion forums on concepts related to this chapter, see <http://www.telecomessentials.com/learningcenter>.

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