PART

2

Physical Layer and Media

Objectives

We start the discussion of the Internet model with the bottom-most layer, the physical layer. It is the layer that actually interacts with the transmission media, the physical part of the network that connects network components together. This layer is involved in physically carrying information from one node in the network to the next.

The physical layer has complex tasks to perform. One major task is to provide services for the data link layer. The data in the data link layer consists of 0s and 1s organized into frames that are ready to be sent across the transmission medium. This stream of 0s and 1s must first be converted into another entity: signals. One of the services provided by the physical layer is to create a signal that represents this stream of bits.

The physical layer must also take care of the physical network, the transmission medium. The transmission medium is a passive entity; it has no internal program or logic for control like other layers. The transmission medium must be controlled by the physical layer. The physical layer decides on the directions of data flow. The physical layer decides on the number of logical channels for transporting data coming from different sources.

In Part 2 of the book, we discuss issues related to the physical layer and the transmission medium that is controlled by the physical layer. In the last chapter of Part 2, we discuss the structure and the physical layers of the telephone network and the cable network.

Part 2 of the book is devoted to the physical layer and the transmission media.

Chapters

This part consists of seven chapters: Chapters 3 to 9.

Chapter 3

Chapter 3 discusses the relationship between data, which are created by a device, and electromagnetic signals, which are transmitted over a medium.

Chapter 4

Chapter 4 deals with digital transmission. We discuss how we can covert digital or analog data to digital signals.

Chapter 5

Chapter 5 deals with analog transmission. We discuss how we can covert digital or analog data to analog signals.

Chapter 6

Chapter 6 shows how we can use the available bandwidth efficiently. We discuss two separate, but related topics, multiplexing and spreading.

Chapter 7

After explaining some ideas about data and signals and how we can use them efficiently, we discuss the characteristics of transmission media, both guided and unguided, in this chapter. Although transmission media operates under the physical layer, they are controlled by the physical layer.

Chapter 8

Although the previous chapters in this part are issues related to the physical layer or transmission media, Chapter 8 discusses switching, a topic that can be related to several layers. We have included this topic in this part of the book to avoid repeating the discussion for each layer.

Chapter 9

Chapter 9 shows how the issues discussed in the previous chapters can be used in actual networks. In this chapter, we first discuss the telephone network as designed to carry voice. We then show how it can be used to carry data. Second, we discuss the cable network as a television network. We then show how it can also be used to carry data.

CHAPTER 3

Data and Signals

One of the major functions of the physical layer is to move data in the form of electromagnetic signals across a transmission medium. Whether you are collecting numerical statistics from another computer, sending animated pictures from a design workstation, or causing a bell to ring at a distant control center, you are working with the transmission of **data** across network connections.

Generally, the data usable to a person or application are not in a form that can be transmitted over a network. For example, a photograph must first be changed to a form that transmission media can accept. Transmission media work by conducting energy along a physical path.

To be transmitted, data must be transformed to electromagnetic signals.

3.1 ANALOG AND DIGITAL

Both data and the signals that represent them can be either **analog** or **digital** in form.

Analog and Digital Data

Data can be analog or digital. The term **analog data** refers to information that is continuous; **digital data** refers to information that has discrete states. For example, an analog clock that has hour, minute, and second hands gives information in a continuous form; the movements of the hands are continuous. On the other hand, a digital clock that reports the hours and the minutes will change suddenly from 8:05 to 8:06.

Analog data, such as the sounds made by a human voice, take on continuous values. When someone speaks, an analog wave is created in the air. This can be captured by a microphone and converted to an analog signal or sampled and converted to a digital signal.

Digital data take on discrete values. For example, data are stored in computer memory in the form of 0s and 1s. They can be converted to a digital signal or modulated into an analog signal for transmission across a medium.

Data can be analog or digital. Analog data are continuous and take continuous values. Digital data have discrete states and take discrete values.

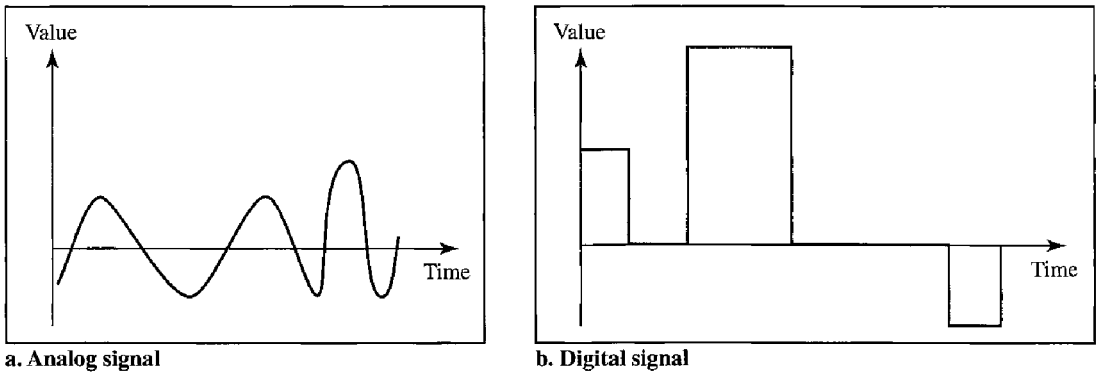
Analog and Digital Signals

Like the data they represent, **signals** can be either analog or digital. An **analog signal** has infinitely many levels of intensity over a period of time. As the wave moves from value *A* to value *B*, it passes through and includes an infinite number of values along its path. A **digital signal,** on the other hand, can have only a limited number of defined values. Although each value can be any number, it is often as simple as 1 and 0.

The simplest way to show signals is by plotting them on a pair of perpendicular axes. The vertical axis represents the value or strength of a signal. The horizontal axis represents time. Figure 3.1 illustrates an analog signal and a digital signal. The curve representing the analog signal passes through an infinite number of points. The vertical lines of the digital signal, however, demonstrate the sudden jump that the signal makes from value to value.

Signals can be analog or digital. Analog signals can have an infinite number of values in a range; digital signals can have only a limited number of values.

Figure 3.1 *Comparison of analog and digital signals*



a. Analog signal **b. Digital signal**

Periodic and Nonperiodic Signals

Both analog and digital signals can take one of two forms: *periodic* or *nonperiodic* (sometimes refer to as *aperiodic,* because the prefix *a* in Greek means "non").

A **periodic signal** completes a pattern within a measurable time frame, called a **period,** and repeats that pattern over subsequent identical periods. The completion of one full pattern is called a **cycle.** A **nonperiodic signal** changes without exhibiting a pattern or cycle that repeats over time.

Both analog and digital signals can be periodic or nonperiodic. In data communications, we commonly use periodic analog signals (because they need less bandwidth,

as we will see in Chapter 5) and nonperiodic digital signals (because they can represent variation in data, as we will see in Chapter 6).

In data communications, we commonly use periodic analog signals and nonperiodic digital signals.

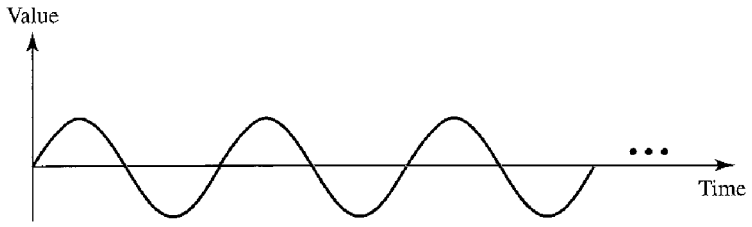
3.2 PERIODIC ANALOG SIGNALS

Periodic analog signals can be classified as simple or composite. A simple periodic analog signal, a **sine wave,** cannot be decomposed into simpler signals. A composite periodic analog signal is composed of multiple sine waves.

Sine Wave

The sine wave is the most fundamental form of a periodic analog signal. When we visualize it as a simple oscillating curve, its change over the course of a cycle is smooth and consistent, a continuous, rolling flow. Figure 3.2 shows a sine wave. Each cycle consists of a single arc above the time axis followed by a single arc below it.

Figure 3.2 *A sine wave*



We discuss a mathematical approach to sine waves in Appendix C.

A sine wave can be represented by three parameters: the *peak amplitude,* the *frequency,* and the *phase*. These three parameters fully describe a sine wave.

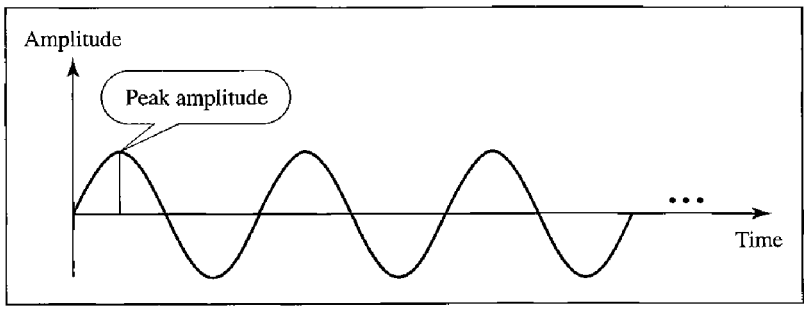
Peak Amplitude

The **peak amplitude** of a signal is the absolute value of its highest intensity, proportional to the energy it carries. For electric signals, peak amplitude is normally measured in *volts*. Figure 3.3 shows two signals and their peak amplitudes.

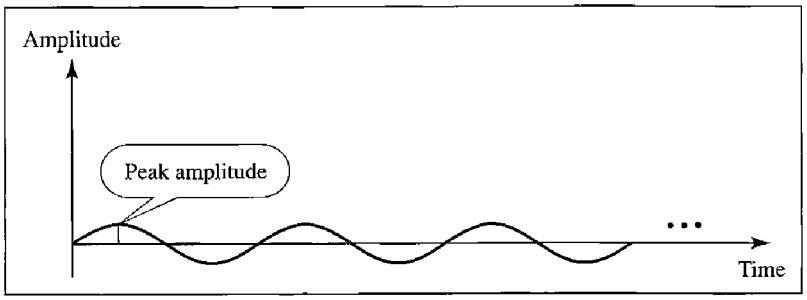
Example 3.1

The power in your house can be represented by a sine wave with a peak amplitude of 155 to 170 V. However, it is common knowledge that the voltage of the power in U.S. homes is 110 to 120 V.

Figure 3.3 *Two signals with the same phase and frequency, but different amplitudes*



a. A signal with high peak amplitude



b. A signal with low peak amplitude

This discrepancy is due to the fact that these are root mean square (rms) values. The signal is squared and then the average amplitude is calculated. The peak value is equal to $2^{1/2} \times$ rms value.

Example 3.2

The voltage of battery is a constant; this constant value can be considered a sine wave, as we will see later. For example, the peak value of an AA battery is normally 1.5 V.

Period and Frequency

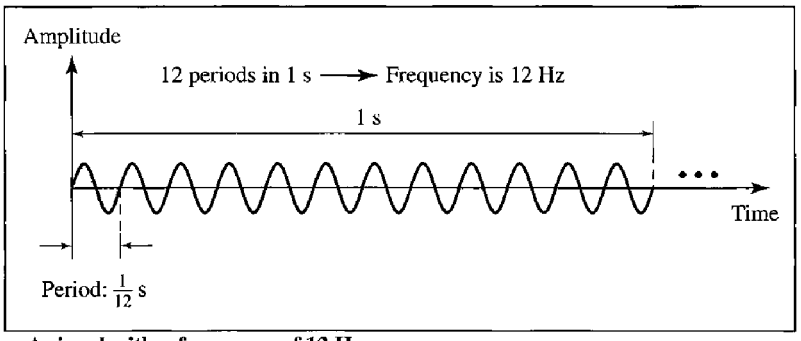
Period refers to the amount of time, in seconds, a signal needs to complete 1 cycle. **Frequency** refers to the number of periods in 1 s. Note that period and frequency are just one characteristic defined in two ways. Period is the inverse of frequency, and frequency is the inverse of period, as the following formulas show.

$$f = \frac{1}{T} \quad \text{and} \quad T = \frac{1}{f}$$

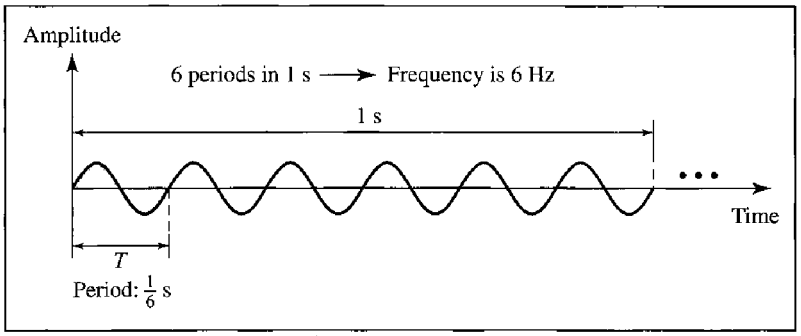
Frequency and period are the inverse of each other.

Figure 3.4 shows two signals and their frequencies.

Figure 3.4 *Two signals with the same amplitude and phase, but different frequencies*



a. A signal with a frequency of 12 Hz



b. A signal with a frequency of 6 Hz

Period is formally expressed in seconds. Frequency is formally expressed in **Hertz (Hz),** which is cycle per second. Units of period and frequency are shown in Table 3.1.

Table 3.1 *Units of period and frequency*

Unit	*Equivalent*	*Unit*	*Equivalent*
Seconds (s)	1 s	Hertz (Hz)	1 Hz
Milliseconds (ms)	10^{-3} s	Kilohertz (kHz)	10^{3} Hz
Microseconds (μs)	10^{-6} s	Megahertz (MHz)	10^{6} Hz
Nanoseconds (ns)	10^{-9} s	Gigahertz (GHz)	10^{9} Hz
Picoseconds (ps)	10^{-12} s	Terahertz (THz)	10^{12} Hz

Example 3.3

The power we use at home has a frequency of 60 Hz (50 Hz in Europe). The period of this sine wave can be determined as follows:

$$T = \frac{1}{f} = \frac{1}{60} = 0.0166 \text{ s} = 0.0166 \times 10^3 \text{ ms} = 16.6 \text{ ms}$$

This means that the period of the power for our lights at home is 0.0116 s, or 16.6 ms. Our eyes are not sensitive enough to distinguish these rapid changes in amplitude.

Example 3.4

Express a period of 100 ms in microseconds.

Solution

From Table 3.1 we find the equivalents of 1 ms (1 ms is 10^{-3} s) and 1 s (1 s is 10^{6} μs). We make the following substitutions:

$$100 \text{ ms} = 100 \times 10^{-3} \text{ s} = 100 \times 10^{-3} \times 10^{6} \text{ μs} = 10^{2} \times 10^{-3} \times 10^{6} \text{ μs} = 10^{5} \text{ μs}$$

Example 3.5

The period of a signal is 100 ms. What is its frequency in kilohertz?

Solution

First we change 100 ms to seconds, and then we calculate the frequency from the period (1 Hz = 10^{-3} kHz).

$$100 \text{ ms} = 100 \times 10^{-3} \text{ s} = 10^{-1} \text{ s}$$

$$f = \frac{1}{T} = \frac{1}{10^{-1}} \text{ Hz} = 10 \text{ Hz} = 10 \times 10^{-3} \text{ kHz} = 10^{-2} \text{ kHz}$$

More About Frequency

We already know that frequency is the relationship of a signal to time and that the frequency of a wave is the number of cycles it completes in 1 s. But another way to look at frequency is as a measurement of the rate of change. Electromagnetic signals are oscillating waveforms; that is, they fluctuate continuously and predictably above and below a mean energy level. A 40-Hz signal has one-half the frequency of an 80-Hz signal; it completes 1 cycle in twice the time of the 80-Hz signal, so each cycle also takes twice as long to change from its lowest to its highest voltage levels. Frequency, therefore, though described in cycles per second (hertz), is a general measurement of the rate of change of a signal with respect to time.

Frequency is the rate of change with respect to time. Change in a short span of time means high frequency. Change over a long span of time means low frequency.

If the value of a signal changes over a very short span of time, its frequency is high. If it changes over a long span of time, its frequency is low.

Two Extremes

What if a signal does not change at all? What if it maintains a constant voltage level for the entire time it is active? In such a case, its frequency is zero. Conceptually, this idea is a simple one. If a signal does not change at all, it never completes a cycle, so its frequency is 0 Hz.

But what if a signal changes instantaneously? What if it jumps from one level to another in no time? Then its frequency is infinite. In other words, when a signal changes instantaneously, its period is zero; since frequency is the inverse of period, in this case, the frequency is 1/0, or infinite (unbounded).

If a signal does not change at all, its frequency is zero.
If a signal changes instantaneously, its frequency is infinite.

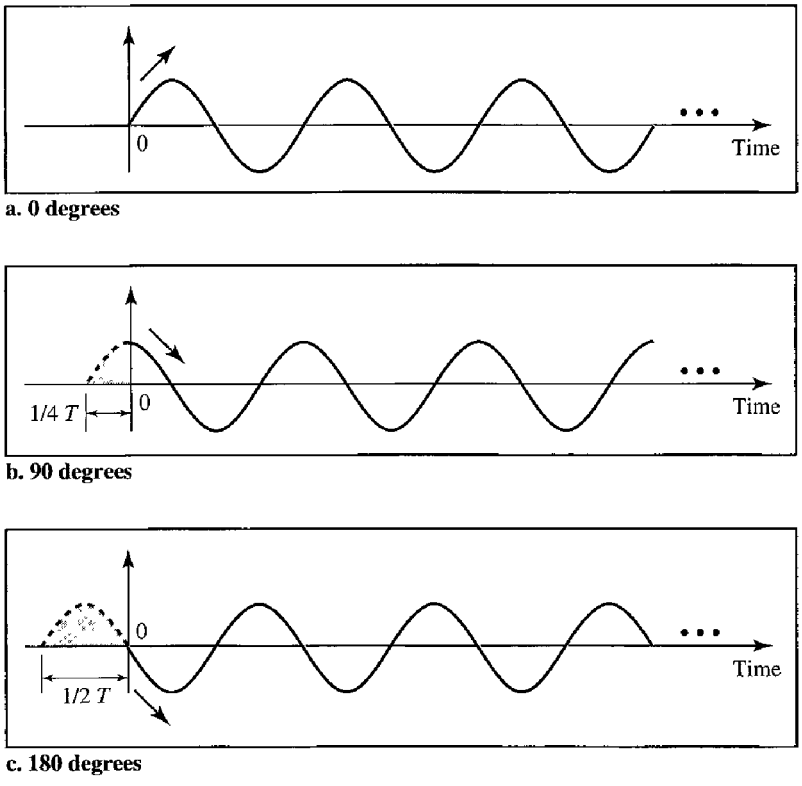
Phase

The term **phase** describes the position of the waveform relative to time 0. If we think of the wave as something that can be shifted backward or forward along the time axis, phase describes the amount of that shift. It indicates the status of the first cycle.

Phase describes the position of the waveform relative to time 0.

Phase is measured in degrees or radians [360° is 2π rad; 1° is $2\pi/360$ rad, and 1 rad is $360/(2\pi)$]. A phase shift of 360° corresponds to a shift of a complete period; a phase shift of 180° corresponds to a shift of one-half of a period; and a phase shift of 90° corresponds to a shift of one-quarter of a period (see Figure 3.5).

Figure 3.5 *Three sine waves with the same amplitude and frequency, but different phases*



Looking at Figure 3.5, we can say that

1. A sine wave with a phase of 0° starts at time 0 with a zero amplitude. The amplitude is increasing.
2. A sine wave with a phase of 90° starts at time 0 with a peak amplitude. The amplitude is decreasing.

3. A sine wave with a phase of 180° starts at time 0 with a zero amplitude. The amplitude is decreasing.

Another way to look at the phase is in terms of shift or offset. We can say that

1. A sine wave with a phase of 0° is not shifted.
2. A sine wave with a phase of 90° is shifted to the left by $\frac{1}{4}$ cycle. However, note that the signal does not really exist before time 0.
3. A sine wave with a phase of 180° is shifted to the left by $\frac{1}{2}$ cycle. However, note that the signal does not really exist before time 0.

Example 3.6

A sine wave is offset $\frac{1}{6}$ cycle with respect to time 0. What is its phase in degrees and radians?

Solution

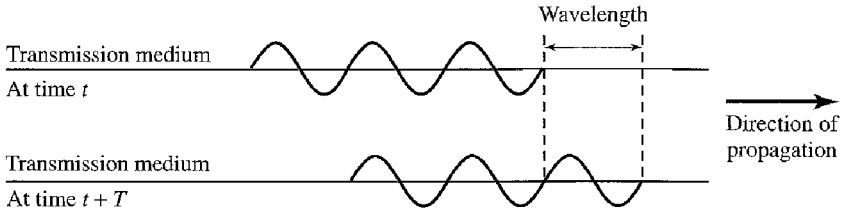
We know that 1 complete cycle is 360°. Therefore, $\frac{1}{6}$ cycle is

$$\frac{1}{6} \times 360 = 60° = 60 \times \frac{2\pi}{360} \text{ rad} = \frac{\pi}{3} \text{ rad} = 1.046 \text{ rad}$$

Wavelength

Wavelength is another characteristic of a signal traveling through a transmission medium. Wavelength binds the period or the frequency of a simple sine wave to the **propagation speed** of the medium (see Figure 3.6).

Figure 3.6 *Wavelength and period*



While the frequency of a signal is independent of the medium, the wavelength depends on both the frequency and the medium. Wavelength is a property of any type of signal. In data communications, we often use wavelength to describe the transmission of light in an optical fiber. The wavelength is the distance a simple signal can travel in one period.

Wavelength can be calculated if one is given the propagation speed (the speed of light) and the period of the signal. However, since period and frequency are related to each other, if we represent wavelength by λ, propagation speed by c (speed of light), and frequency by f, we get

$$\textbf{Wavelength} = \textbf{propagation speed} \times \textbf{period} = \frac{\textbf{propagation speed}}{\textbf{frequency}}$$

$$\lambda = \frac{c}{f}$$

The propagation speed of electromagnetic signals depends on the medium and on the frequency of the signal. For example, in a vacuum, light is propagated with a speed of 3×10^8 m/s. That speed is lower in air and even lower in cable.

The wavelength is normally measured in micrometers (microns) instead of meters. For example, the wavelength of red light (frequency = 4×10^{14}) in air is

$$\lambda = \frac{c}{f} = \frac{3 \times 10^8}{4 \times 10^{14}} = 0.75 \times 10^{-6} \text{ m} = 0.75 \ \mu\text{m}$$

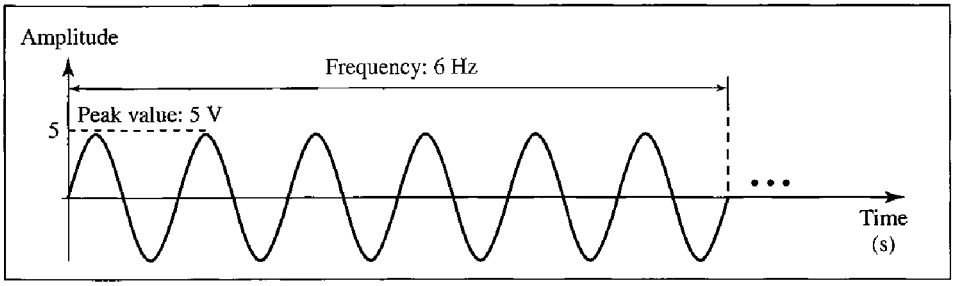
In a coaxial or fiber-optic cable, however, the wavelength is shorter (0.5 μm) because the propagation speed in the cable is decreased.

Time and Frequency Domains

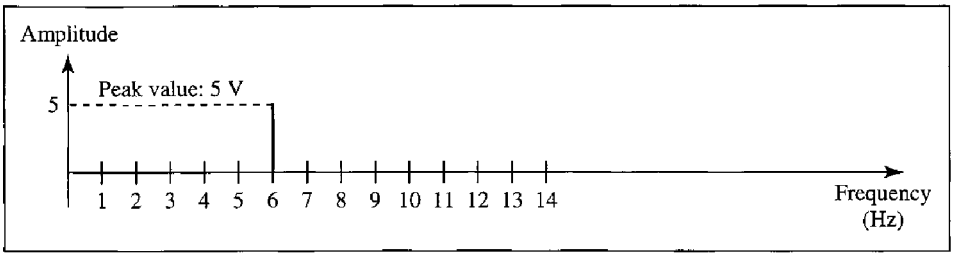
A sine wave is comprehensively defined by its amplitude, frequency, and phase. We have been showing a sine wave by using what is called a **time-domain** plot. The time-domain plot shows changes in signal amplitude with respect to time (it is an amplitude-versus-time plot). Phase is not explicitly shown on a time-domain plot.

To show the relationship between amplitude and frequency, we can use what is called a **frequency-domain** plot. A frequency-domain plot is concerned with only the peak value and the frequency. Changes of amplitude during one period are not shown. Figure 3.7 shows a signal in both the time and frequency domains.

Figure 3.7 *The time-domain and frequency-domain plots of a sine wave*



a. A sine wave in the time domain (peak value: 5 V, frequency: 6 Hz)



b. The same sine wave in the frequency domain (peak value: 5 V, frequency: 6 Hz)

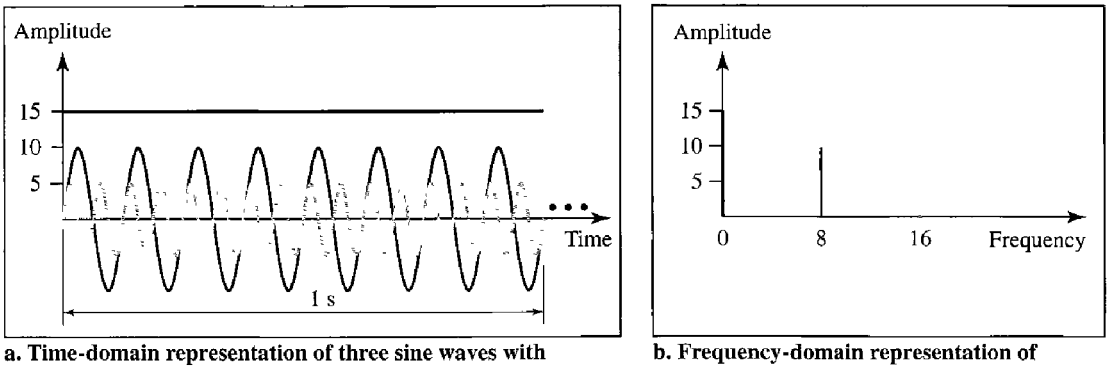
It is obvious that the frequency domain is easy to plot and conveys the information that one can find in a time domain plot. The advantage of the frequency domain is that we can immediately see the values of the frequency and peak amplitude. A complete sine wave is represented by one spike. The position of the spike shows the frequency; its height shows the peak amplitude.

A complete sine wave in the time domain can be represented by one single spike in the frequency domain.

Example 3.7

The frequency domain is more compact and useful when we are dealing with more than one sine wave. For example, Figure 3.8 shows three sine waves, each with different amplitude and frequency. All can be represented by three spikes in the frequency domain.

Figure 3.8 *The time domain and frequency domain of three sine waves*



a. Time-domain representation of three sine waves with frequencies 0, 8, and 16

b. Frequency-domain representation of the same three signals

Composite Signals

So far, we have focused on simple sine waves. Simple sine waves have many applications in daily life. We can send a single sine wave to carry electric energy from one place to another. For example, the power company sends a single sine wave with a frequency of 60 Hz to distribute electric energy to houses and businesses. As another example, we can use a single sine wave to send an alarm to a security center when a burglar opens a door or window in the house. In the first case, the sine wave is carrying energy; in the second, the sine wave is a signal of danger.

If we had only one single sine wave to convey a conversation over the phone, it would make no sense and carry no information. We would just hear a buzz. As we will see in Chapters 4 and 5, we need to send a composite signal to communicate data. A **composite signal** is made of many simple sine waves.

A single-frequency sine wave is not useful in data communications; we need to send a composite signal, a signal made of many simple sine waves.

In the early 1900s, the French mathematician Jean-Baptiste Fourier showed that any composite signal is actually a combination of simple sine waves with different frequencies, amplitudes, and phases. **Fourier analysis** is discussed in Appendix C; for our purposes, we just present the concept.

According to Fourier analysis, any composite signal is a combination of simple sine waves with different frequencies, amplitudes, and phases. Fourier analysis is discussed in Appendix C.

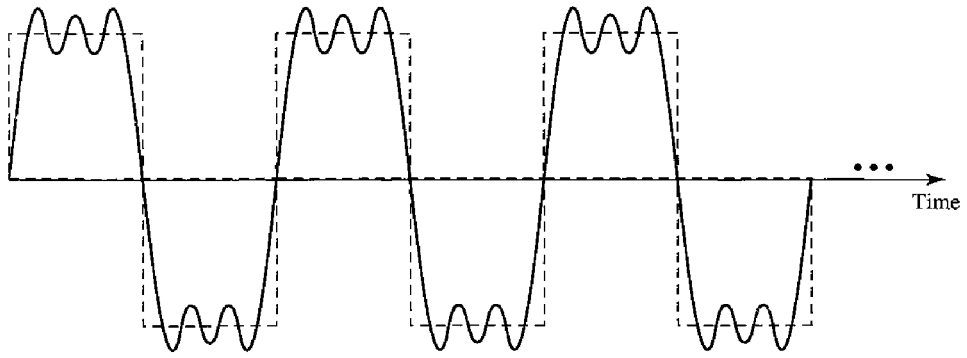
A composite signal can be periodic or nonperiodic. A periodic composite signal can be decomposed into a series of simple sine waves with discrete frequencies—frequencies that have integer values (1, 2, 3, and so on). A nonperiodic composite signal can be decomposed into a combination of an infinite number of simple sine waves with continuous frequencies, frequencies that have real values.

If the composite signal is periodic, the decomposition gives a series of signals with discrete frequencies; if the composite signal is nonperiodic, the decomposition gives a combination of sine waves with continuous frequencies.

Example 3.8

Figure 3.9 shows a periodic composite signal with frequency f. This type of signal is not typical of those found in data communications.We can consider it to be three alarm systems, each with a different frequency. The analysis of this signal can give us a good understanding of how to decompose signals.

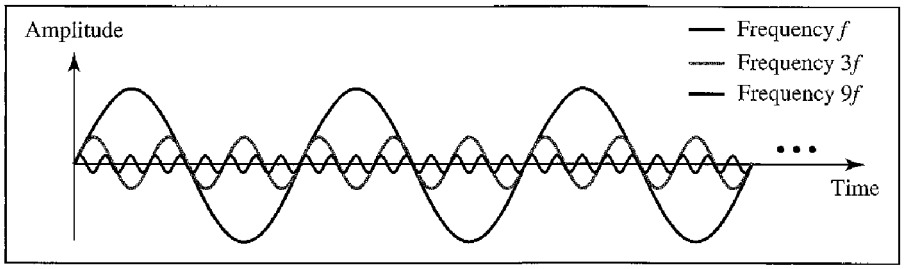
Figure 3.9 *A composite periodic signal*



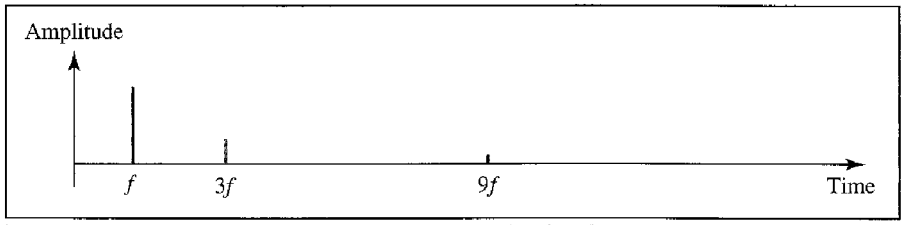
It is very difficult to manually decompose this signal into a series of simple sine waves. However, there are tools, both hardware and software, that can help us do the job. We are not concerned about how it is done; we are only interested in the result. Figure 3.10 shows the result of decomposing the above signal in both the time and frequency domains.

The amplitude of the sine wave with frequency f is almost the same as the peak amplitude of the composite signal. The amplitude of the sine wave with frequency $3f$ is one-third of that of the first, and the amplitude of the sine wave with frequency $9f$ is one-ninth of the first. The frequency

Figure 3.10 *Decomposition of a composite periodic signal in the time and frequency domains*



a. Time-domain decomposition of a composite signal



b. Frequency-domain decomposition of the composite signal

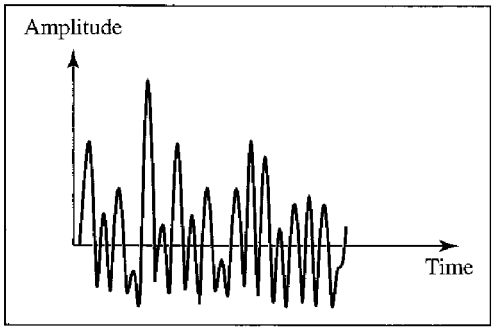
of the sine wave with frequency f is the same as the frequency of the composite signal; it is called the **fundamental frequency,** or first **harmonic.** The sine wave with frequency $3f$ has a frequency of 3 times the fundamental frequency; it is called the third harmonic. The third sine wave with frequency $9f$ has a frequency of 9 times the fundamental frequency; it is called the ninth harmonic.

Note that the frequency decomposition of the signal is discrete; it has frequencies f, $3f$, and $9f$. Because f is an integral number, $3f$ and $9f$ are also integral numbers. There are no frequencies such as $1.2f$ or $2.6f$. The frequency domain of a periodic composite signal is always made of discrete spikes.

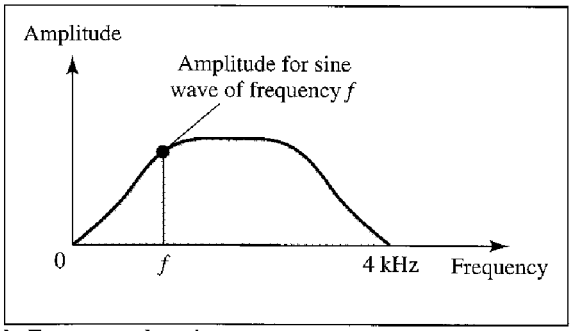
Example 3.9

Figure 3.11 shows a nonperiodic composite signal. It can be the signal created by a microphone or a telephone set when a word or two is pronounced. In this case, the composite signal cannot be periodic, because that implies that we are repeating the same word or words with exactly the same tone.

Figure 3.11 *The time and frequency domains of a nonperiodic signal*



a. Time domain



b. Frequency domain

In a time-domain representation of this composite signal, there are an infinite number of simple sine frequencies. Although the number of frequencies in a human voice is infinite, the range is limited. A normal human being can create a continuous range of frequencies between 0 and 4 kHz.

Note that the frequency decomposition of the signal yields a continuous curve. There are an infinite number of frequencies between 0.0 and 4000.0 (real values). To find the amplitude related to frequency f, we draw a vertical line at f to intersect the envelope curve. The height of the vertical line is the amplitude of the corresponding frequency.

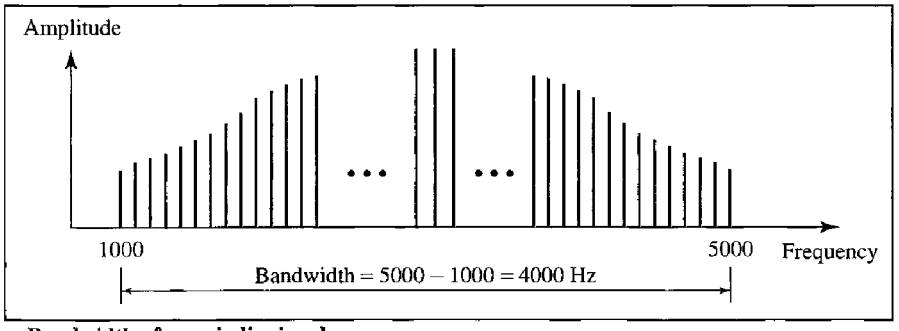
Bandwidth

The range of frequencies contained in a composite signal is its **bandwidth.** The bandwidth is normally a difference between two numbers. For example, if a composite signal contains frequencies between 1000 and 5000, its bandwidth is 5000 – 1000, or 4000.

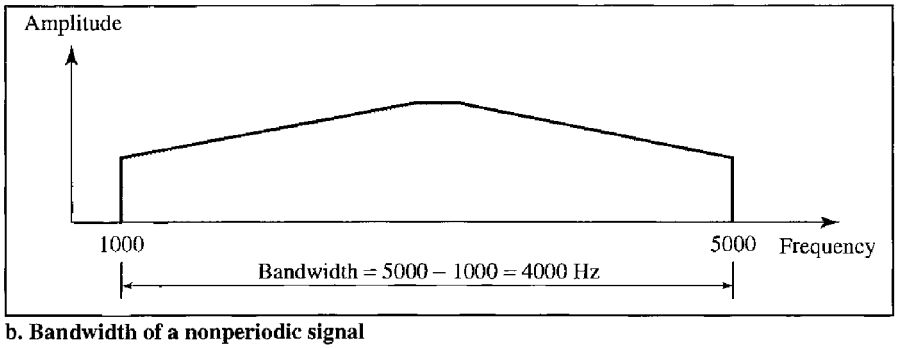
The bandwidth of a composite signal is the difference between the highest and the lowest frequencies contained in that signal.

Figure 3.12 shows the concept of bandwidth. The figure depicts two composite signals, one periodic and the other nonperiodic. The bandwidth of the periodic signal contains all integer frequencies between 1000 and 5000 (1000, 1001, 1002, . . .). The bandwidth of the nonperiodic signals has the same range, but the frequencies are continuous.

Figure 3.12 *The bandwidth of periodic and nonperiodic composite signals*



a. Bandwidth of a periodic signal



b. Bandwidth of a nonperiodic signal

Example 3.10

If a periodic signal is decomposed into five sine waves with frequencies of 100, 300, 500, 700, and 900 Hz, what is its bandwidth? Draw the spectrum, assuming all components have a maximum amplitude of 10 V.

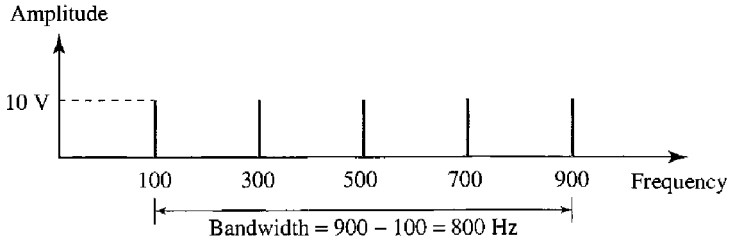
Solution

Let f_h be the highest frequency, f_l the lowest frequency, and B the bandwidth. Then

$$B = f_h - f_l = 900 - 100 = 800 \text{ Hz}$$

The spectrum has only five spikes, at 100, 300, 500, 700, and 900 Hz (see Figure 3.13).

Figure 3.13 *The bandwidth for Example 3.10*



Example 3.11

A periodic signal has a bandwidth of 20 Hz. The highest frequency is 60 Hz. What is the lowest frequency? Draw the spectrum if the signal contains all frequencies of the same amplitude.

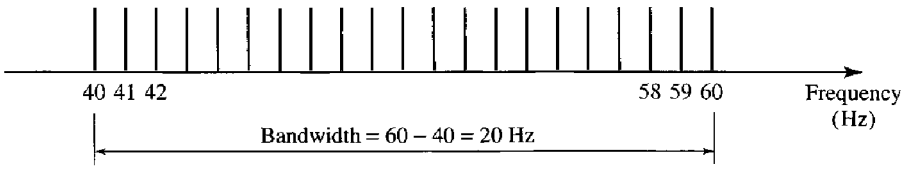
Solution

Let f_h be the highest frequency, f_l the lowest frequency, and B the bandwidth. Then

$$B = f_h - f_l \Rightarrow 20 = 60 - f_l \Rightarrow f_l = 60 - 20 = 40 \text{ Hz}$$

The spectrum contains all integer frequencies. We show this by a series of spikes (see Figure 3.14).

Figure 3.14 *The bandwidth for Example 3.11*



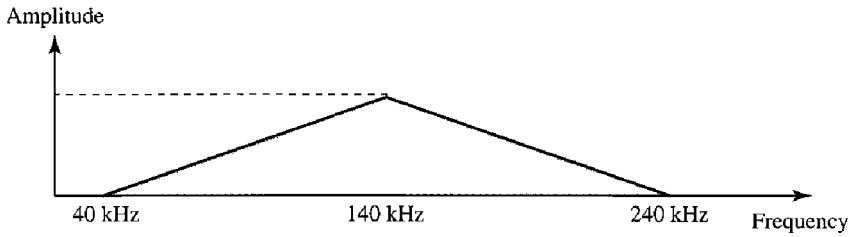
Example 3.12

A nonperiodic composite signal has a bandwidth of 200 kHz, with a middle frequency of 140 kHz and peak amplitude of 20 V. The two extreme frequencies have an amplitude of 0. Draw the frequency domain of the signal.

Solution

The lowest frequency must be at 40 kHz and the highest at 240 kHz. Figure 3.15 shows the frequency domain and the bandwidth.

Figure 3.15 *The bandwidth for Example 3.12*



Example 3.13

An example of a nonperiodic composite signal is the signal propagated by an AM radio station. In the United States, each AM radio station is assigned a 10-kHz bandwidth. The total bandwidth dedicated to AM radio ranges from 530 to 1700 kHz. We will show the rationale behind this 10-kHz bandwidth in Chapter 5.

Example 3.14

Another example of a nonperiodic composite signal is the signal propagated by an FM radio station. In the United States, each FM radio station is assigned a 200-kHz bandwidth. The total bandwidth dedicated to FM radio ranges from 88 to 108 MHz. We will show the rationale behind this 200-kHz bandwidth in Chapter 5.

Example 3.15

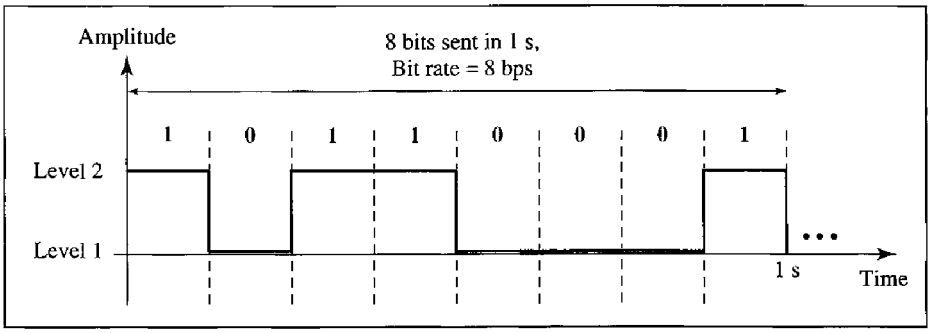
Another example of a nonperiodic composite signal is the signal received by an old-fashioned analog black-and-white TV. A TV screen is made up of pixels (picture elements) with each pixel being either white or black. The screen is scanned 30 times per second. (Scanning is actually 60 times per second, but odd lines are scanned in one round and even lines in the next and then interleaved.) If we assume a resolution of 525×700 (525 vertical lines and 700 horizontal lines), which is a ratio of 3:4, we have 367,500 pixels per screen. If we scan the screen 30 times per second, this is $367{,}500 \times 30 = 11{,}025{,}000$ pixels per second. The worst-case scenario is alternating black and white pixels. In this case, we need to represent one color by the minimum amplitude and the other color by the maximum amplitude. We can send 2 pixels per cycle. Therefore, we need $11{,}025{,}000 / 2 = 5{,}512{,}500$ cycles per second, or Hz. The bandwidth needed is 5.5124 MHz. This worst-case scenario has such a low probability of occurrence that the assumption is that we need only 70 percent of this bandwidth, which is 3.85 MHz. Since audio and synchronization signals are also needed, a 4-MHz bandwidth has been set aside for each black and white TV channel. An analog color TV channel has a 6-MHz bandwidth.

3.3 DIGITAL SIGNALS

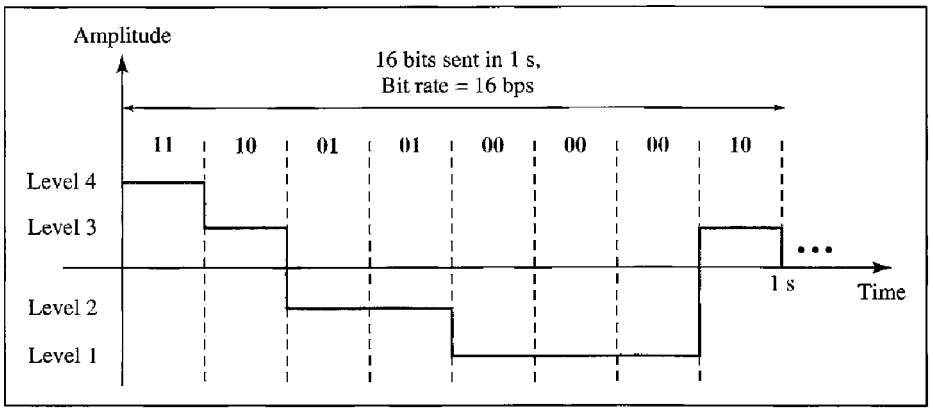
In addition to being represented by an analog signal, information can also be represented by a digital signal. For example, a 1 can be encoded as a positive voltage and a 0 as zero voltage. A digital signal can have more than two levels. In this case, we can

send more than 1 bit for each level. Figure 3.16 shows two signals, one with two levels and the other with four.

Figure 3.16 *Two digital signals: one with two signal levels and the other with four signal levels*



a. A digital signal with two levels



b. A digital signal with four levels

We send 1 bit per level in part a of the figure and 2 bits per level in part b of the figure. In general, if a signal has L levels, each level needs $\log_2 L$ bits.

Appendix C reviews information about exponential and logarithmic functions.

Example 3.16

A digital signal has eight levels. How many bits are needed per level? We calculate the number of bits from the formula

$$\text{Number of bits per level} = \log_2 8 = 3$$

Each signal level is represented by 3 bits.

Example 3.17

A digital signal has nine levels. How many bits are needed per level? We calculate the number of bits by using the formula. Each signal level is represented by 3.17 bits. However, this answer is not realistic. The number of bits sent per level needs to be an integer as well as a power of 2. For this example, 4 bits can represent one level.

Bit Rate

Most digital signals are nonperiodic, and thus period and frequency are not appropriate characteristics. Another term—*bit rate* (instead of *frequency*)—is used to describe digital signals. The **bit rate** is the number of bits sent in 1s, expressed in **bits per second (bps).** Figure 3.16 shows the bit rate for two signals.

Example 3.18

Assume we need to download text documents at the rate of 100 pages per minute. What is the required bit rate of the channel?

Solution

A page is an average of 24 lines with 80 characters in each line. If we assume that one character requires 8 bits, the bit rate is

$$100 \times 24 \times 80 \times 8 = 1{,}636{,}000 \text{ bps} = 1.636 \text{ Mbps}$$

Example 3.19

A digitized voice channel, as we will see in Chapter 4, is made by digitizing a 4-kHz bandwidth analog voice signal. We need to sample the signal at twice the highest frequency (two samples per hertz). We assume that each sample requires 8 bits. What is the required bit rate?

Solution

The bit rate can be calculated as

$$2 \times 4000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$$

Example 3.20

What is the bit rate for high-definition TV (HDTV)?

Solution

HDTV uses digital signals to broadcast high quality video signals. The HDTV screen is normally a ratio of 16 : 9 (in contrast to 4 : 3 for regular TV), which means the screen is wider. There are 1920 by 1080 pixels per screen, and the screen is renewed 30 times per second. Twenty-four bits represents one color pixel. We can calculate the bit rate as

$$1920 \times 1080 \times 30 \times 24 = 1{,}492{,}992{,}000 \text{ or } 1.5 \text{ Gbps}$$

The TV stations reduce this rate to 20 to 40 Mbps through compression.

Bit Length

We discussed the concept of the wavelength for an analog signal: the distance one cycle occupies on the transmission medium. We can define something similar for a digital signal: the bit length. The **bit length** is the distance one bit occupies on the transmission medium.

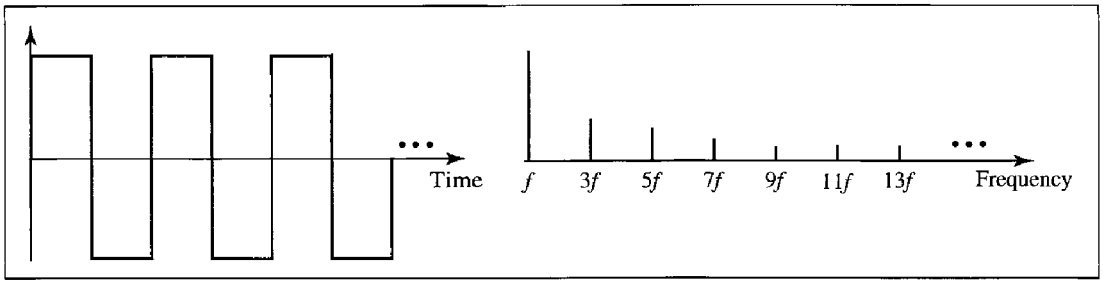
$$\textbf{Bit length} = \textbf{propagation speed} \times \textbf{bit duration}$$

Digital Signal as a Composite Analog Signal

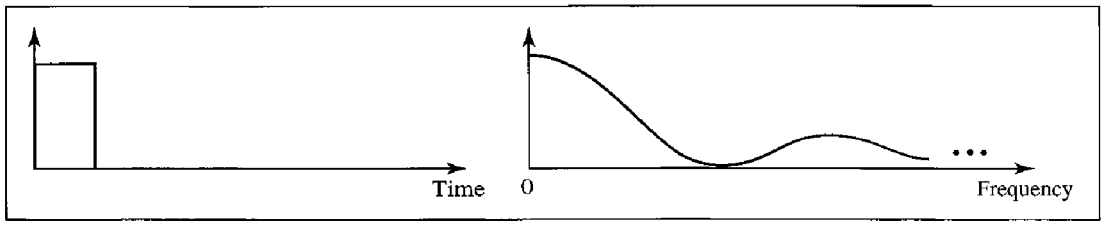
Based on Fourier analysis, a digital signal is a composite analog signal. The bandwidth is infinite, as you may have guessed. We can intuitively come up with this concept when we consider a digital signal. A digital signal, in the time domain, comprises connected vertical and horizontal line segments. A vertical line in the time domain means a frequency of infinity (sudden change in time); a horizontal line in the time domain means a frequency of zero (no change in time). Going from a frequency of zero to a frequency of infinity (and vice versa) implies all frequencies in between are part of the domain.

Fourier analysis can be used to decompose a digital signal. If the digital signal is periodic, which is rare in data communications, the decomposed signal has a frequency-domain representation with an infinite bandwidth and discrete frequencies. If the digital signal is nonperiodic, the decomposed signal still has an infinite bandwidth, but the frequencies are continuous. Figure 3.17 shows a periodic and a nonperiodic digital signal and their bandwidths.

Figure 3.17 *The time and frequency domains of periodic and nonperiodic digital signals*



a. Time and frequency domains of periodic digital signal



b. Time and frequency domains of nonperiodic digital signal

Note that both bandwidths are infinite, but the periodic signal has discrete frequencies while the nonperiodic signal has continuous frequencies.

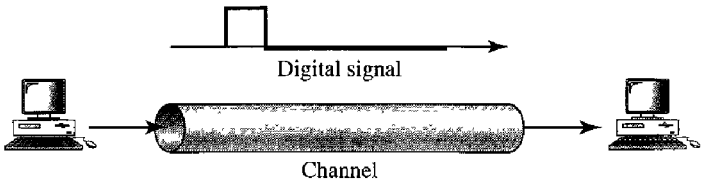
Transmission of Digital Signals

The previous discussion asserts that a digital signal, periodic or nonperiodic, is a composite analog signal with frequencies between zero and infinity. For the remainder of the discussion, let us consider the case of a nonperiodic digital signal, similar to the ones we encounter in data communications. The fundamental question is, How can we send a digital signal from point *A* to point *B*? We can transmit a digital signal by using one of two different approaches: baseband transmission or broadband transmission (using modulation).

Baseband Transmission

Baseband transmission means sending a digital signal over a channel without changing the digital signal to an analog signal. Figure 3.18 shows **baseband** transmission.

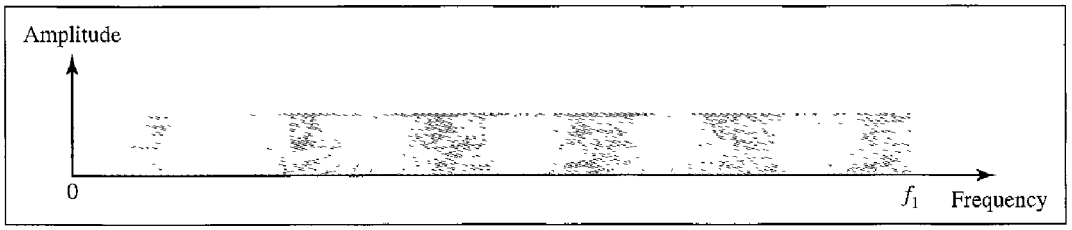
Figure 3.18 *Baseband transmission*



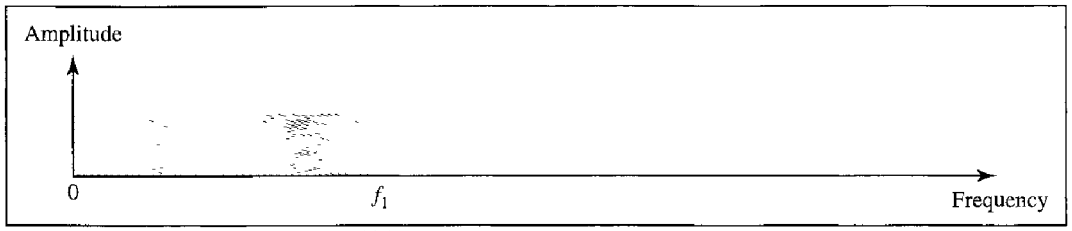
A digital signal is a composite analog signal with an infinite bandwidth.

Baseband transmission requires that we have a **low-pass channel,** a channel with a bandwidth that starts from zero. This is the case if we have a dedicated medium with a bandwidth constituting only one channel. For example, the entire bandwidth of a cable connecting two computers is one single channel. As another example, we may connect several computers to a bus, but not allow more than two stations to communicate at a time. Again we have a low-pass channel, and we can use it for baseband communication. Figure 3.19 shows two low-pass channels: one with a narrow bandwidth and the other with a wide bandwidth. We need to remember that a low-pass channel with infinite bandwidth is ideal, but we cannot have such a channel in real life. However, we can get close.

Figure 3.19 *Bandwidths of two low-pass channels*



a. Low-pass channel, wide bandwidth



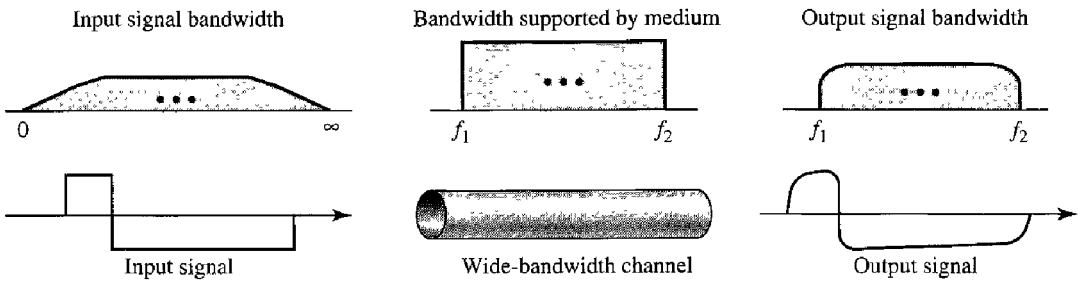
b. Low-pass channel, narrow bandwidth

Let us study two cases of a baseband communication: a low-pass channel with a wide bandwidth and one with a limited bandwidth.

Case 1: Low-Pass Channel with Wide Bandwidth

If we want to preserve the exact form of a nonperiodic digital signal with vertical segments vertical and horizontal segments horizontal, we need to send the entire spectrum, the continuous range of frequencies between zero and infinity. This is possible if we have a dedicated medium with an infinite bandwidth between the sender and receiver that preserves the exact amplitude of each component of the composite signal. Although this may be possible inside a computer (e.g., between CPU and memory), it is not possible between two devices. Fortunately, the amplitudes of the frequencies at the border of the bandwidth are so small that they can be ignored. This means that if we have a medium, such as a coaxial cable or fiber optic, with a very wide bandwidth, two stations can communicate by using digital signals with very good accuracy, as shown in Figure 3.20. Note that f_1 is close to zero, and f_2 is very high.

Figure 3.20 *Baseband transmission using a dedicated medium*



Although the output signal is not an exact replica of the original signal, the data can still be deduced from the received signal. Note that although some of the frequencies are blocked by the medium, they are not critical.

Baseband transmission of a digital signal that preserves the shape of the digital signal is possible only if we have a low-pass channel with an infinite or very wide bandwidth.

Example 3.21

An example of a dedicated channel where the entire bandwidth of the medium is used as one single channel is a LAN. Almost every wired LAN today uses a dedicated channel for two stations communicating with each other. In a bus topology LAN with multipoint connections, only two stations can communicate with each other at each moment in time (timesharing); the other stations need to refrain from sending data. In a star topology LAN, the entire channel between each station and the hub is used for communication between these two entities. We study LANs in Chapter 14.

Case 2: Low-Pass Channel with Limited Bandwidth

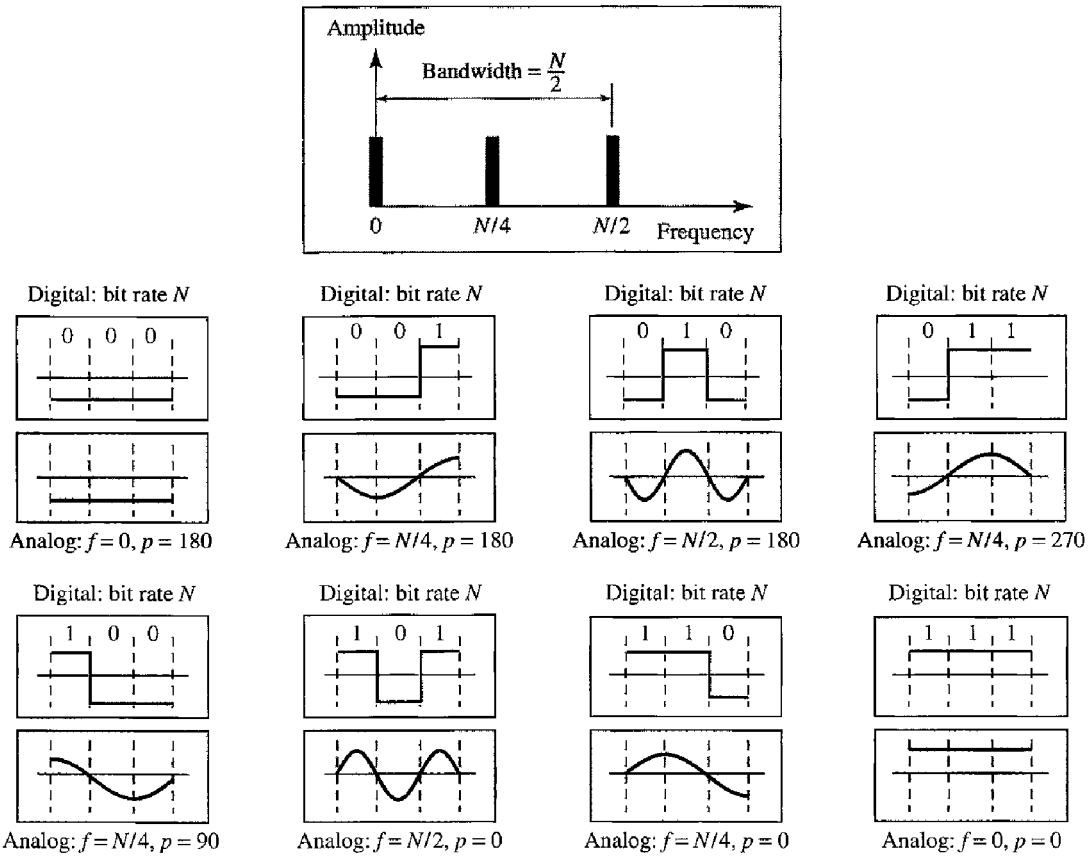
In a low-pass channel with limited bandwidth, we approximate the digital signal with an analog signal. The level of approximation depends on the bandwidth available.

Rough Approximation Let us assume that we have a digital signal of bit rate N. If we want to send analog signals to roughly simulate this signal, we need to consider the worst case, a maximum number of changes in the digital signal. This happens when the signal

carries the sequence 01010101 · · · or the sequence 10101010· · · · To simulate these two cases, we need an analog signal of frequency $f = N/2$. Let 1 be the positive peak value and 0 be the negative peak value. We send 2 bits in each cycle; the frequency of the analog signal is one-half of the bit rate, or $N/2$. However, just this one frequency cannot make all patterns; we need more components. The maximum frequency is $N/2$. As an example of this concept, let us see how a digital signal with a 3-bit pattern can be simulated by using analog signals. Figure 3.21 shows the idea. The two similar cases (000 and 111) are simulated with a signal with frequency $f = 0$ and a phase of 180° for 000 and a phase of 0° for 111. The two worst cases (010 and 101) are simulated with an analog signal with frequency $f = N/2$ and phases of 180° and 0°. The other four cases can only be simulated with an analog signal with $f = N/4$ and phases of 180°, 270°, 90°, and 0°. In other words, we need a channel that can handle frequencies 0, $N/4$, and $N/2$. This rough approximation is referred to as using the first harmonic ($N/2$) frequency. The required bandwidth is

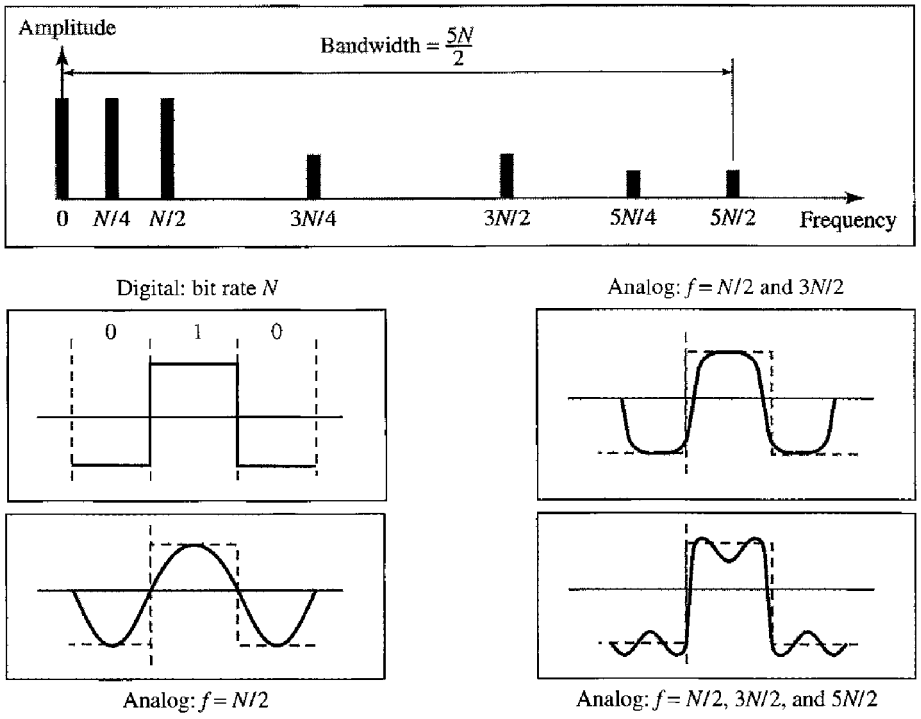
$$\text{Bandwidth} = \frac{N}{2} - 0 = \frac{N}{2}$$

Figure 3.21 *Rough approximation of a digital signal using the first harmonic for worst case*



Better Approximation To make the shape of the analog signal look more like that of a digital signal, we need to add more harmonics of the frequencies. We need to increase the bandwidth. We can increase the bandwidth to $3N/2$, $5N/2$, $7N/2$, and so on. Figure 3.22 shows the effect of this increase for one of the worst cases, the pattern 010.

Figure 3.22 *Simulating a digital signal with three first harmonics*



Note that we have shown only the highest frequency for each harmonic. We use the first, third, and fifth harmonics. The required bandwidth is now $5N/2$, the difference between the lowest frequency 0 and the highest frequency $5N/2$. As we emphasized before, we need to remember that the required bandwidth is proportional to the bit rate.

In baseband transmission, the required bandwidth is proportional to the bit rate; if we need to send bits faster, we need more bandwidth.

By using this method, Table 3.2 shows how much bandwidth we need to send data at different rates.

Table 3.2 *Bandwidth requirements*

Bit Rate	*Harmonic 1*	*Harmonics 1, 3*	*Harmonics 1, 3, 5*
$n = 1$ kbps	$B = 500$ Hz	$B = 1.5$ kHz	$B = 2.5$ kHz
$n = 10$ kbps	$B = 5$ kHz	$B = 15$ kHz	$B = 25$ kHz
$n = 100$ kbps	$B = 50$ kHz	$B = 150$ kHz	$B = 250$ kHz

Example 3.22

What is the required bandwidth of a low-pass channel if we need to send 1 Mbps by using baseband transmission?

Solution

The answer depends on the accuracy desired.

a. The minimum bandwidth, a rough approximation, is B = bit rate /2, or 500 kHz. We need a low-pass channel with frequencies between 0 and 500 kHz.
b. A better result can be achieved by using the first and the third harmonics with the required bandwidth $B = 3 \times 500$ kHz = 1.5 MHz.
c. Still a better result can be achieved by using the first, third, and fifth harmonics with $B = 5 \times 500$ kHz = 2.5 MHz.

Example 3.23

We have a low-pass channel with bandwidth 100 kHz. What is the maximum bit rate of this channel?

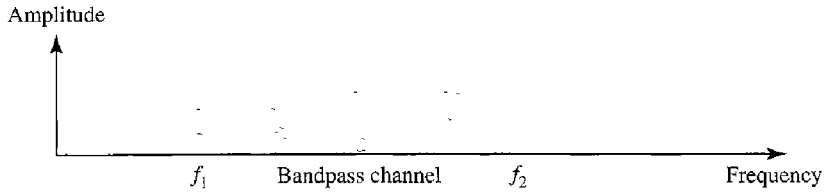
Solution

The maximum bit rate can be achieved if we use the first harmonic. The bit rate is 2 times the available bandwidth, or 200 kbps.

Broadband Transmission (Using Modulation)

Broadband transmission or modulation means changing the digital signal to an analog signal for transmission. Modulation allows us to use a **bandpass channel**—a channel with a bandwidth that does not start from zero. This type of channel is more available than a low-pass channel. Figure 3.23 shows a bandpass channel.

Figure 3.23 *Bandwidth of a bandpass channel*

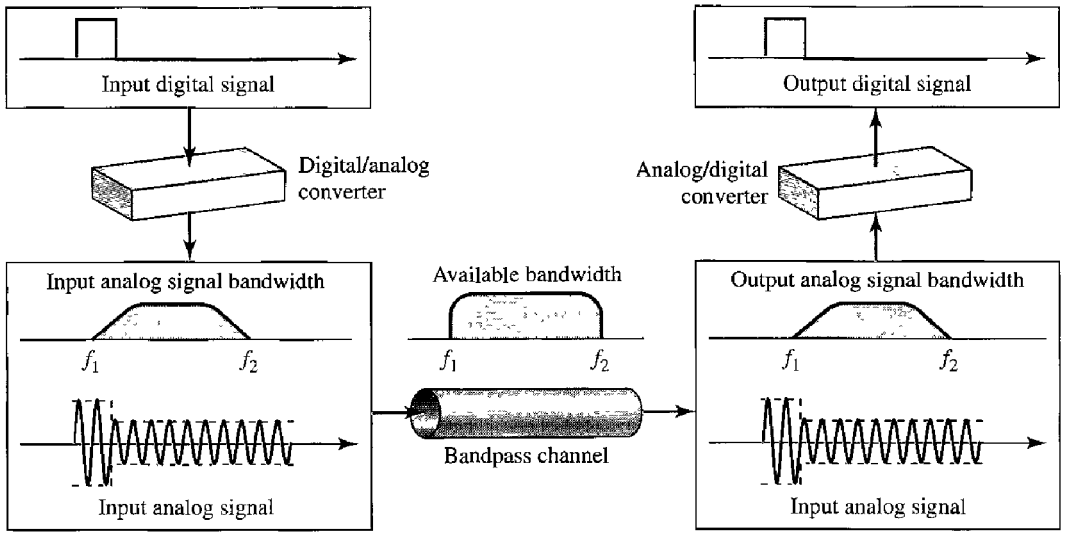


Note that a low-pass channel can be considered a bandpass channel with the lower frequency starting at zero.

Figure 3.24 shows the modulation of a digital signal. In the figure, a digital signal is converted to a composite analog signal. We have used a single-frequency analog signal (called a carrier); the amplitude of the carrier has been changed to look like the digital signal. The result, however, is not a single-frequency signal; it is a composite signal, as we will see in Chapter 5. At the receiver, the received analog signal is converted to digital, and the result is a replica of what has been sent.

If the available channel is a bandpass channel, we cannot send the digital signal directly to the channel; we need to convert the digital signal to an analog signal before transmission.

Figure 3.24 *Modulation of a digital signal for transmission on a bandpass channel*



Example 3.24

An example of broadband transmission using modulation is the sending of computer data through a telephone subscriber line, the line connecting a resident to the central telephone office. These lines, installed many years ago, are designed to carry voice (analog signal) with a limited bandwidth (frequencies between 0 and 4 kHz). Although this channel can be used as a low-pass channel, it is normally considered a bandpass channel. One reason is that the bandwidth is so narrow (4 kHz) that if we treat the channel as low-pass and use it for baseband transmission, the maximum bit rate can be only 8 kbps. The solution is to consider the channel a bandpass channel, convert the digital signal from the computer to an analog signal, and send the analog signal. We can install two converters to change the digital signal to analog and vice versa at the receiving end. The converter, in this case, is called a *modem* (*mo*dulator/*dem*odulator), which we discuss in detail in Chapter 5.

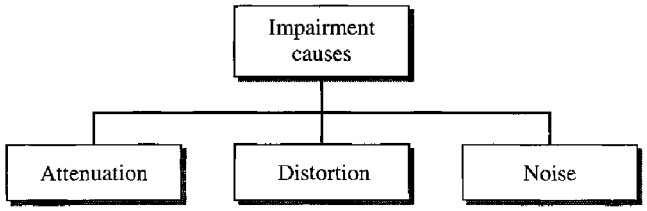
Example 3.25

A second example is the digital cellular telephone. For better reception, digital cellular phones convert the analog voice signal to a digital signal (see Chapter 16). Although the bandwidth allocated to a company providing digital cellular phone service is very wide, we still cannot send the digital signal without conversion. The reason is that we only have a bandpass channel available between caller and callee. For example, if the available bandwidth is W and we allow 1000 couples to talk simultaneously, this means the available channel is W/1000, just part of the entire bandwidth. We need to convert the digitized voice to a composite analog signal before sending. The digital cellular phones convert the analog audio signal to digital and then convert it again to analog for transmission over a bandpass channel.

3.4 TRANSMISSION IMPAIRMENT

Signals travel through transmission media, which are not perfect. The imperfection causes signal impairment. This means that the signal at the beginning of the medium is not the same as the signal at the end of the medium. What is sent is not what is received. Three causes of impairment are attenuation, distortion, and noise (see Figure 3.25).

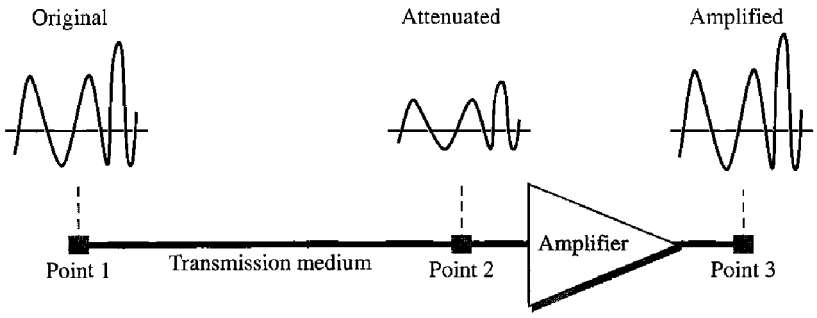
Figure 3.25 *Causes of impairment*



Attenuation

Attenuation means a loss of energy. When a signal, simple or composite, travels through a medium, it loses some of its energy in overcoming the resistance of the medium. That is why a wire carrying electric signals gets warm, if not hot, after a while. Some of the electrical energy in the signal is converted to heat. To compensate for this loss, amplifiers are used to amplify the signal. Figure 3.26 shows the effect of attenuation and amplification.

Figure 3.26 *Attenuation*



Decibel

To show that a signal has lost or gained strength, engineers use the unit of the decibel. The **decibel (dB)** measures the relative strengths of two signals or one signal at two different points. Note that the decibel is negative if a signal is attenuated and positive if a signal is amplified.

$$\mathbf{dB} = \mathbf{10 \log_{10}} \frac{P_2}{P_1}$$

Variables P_1 and P_2 are the powers of a signal at points 1 and 2, respectively. Note that some engineering books define the decibel in terms of voltage instead of power. In this case, because power is proportional to the square of the voltage, the formula is dB = $20 \log_{10} (V_2/V_1)$. In this text, we express dB in terms of power.

Example 3.26

Suppose a signal travels through a transmission medium and its power is reduced to one-half. This means that $P_2 = \frac{1}{2} P_1$. In this case, the attenuation (loss of power) can be calculated as

$$10 \log_{10} \frac{P_2}{P_1} = 10 \log_{10} \frac{0.5 P_1}{P_1} = 10 \log_{10} 0.5 = 10(-0.3) = -3 \text{ dB}$$

A loss of 3 dB (–3 dB) is equivalent to losing one-half the power.

Example 3.27

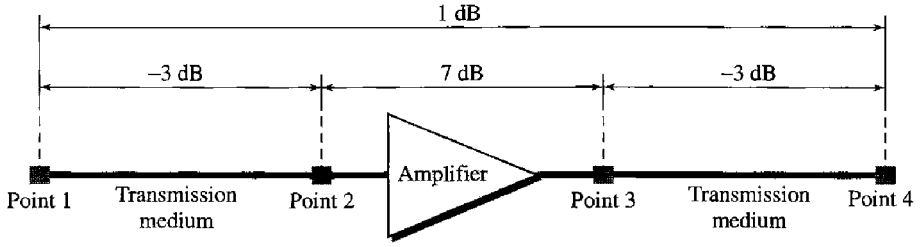
A signal travels through an amplifier, and its power is increased 10 times. This means that $P_2 = 10P_1$. In this case, the amplification (gain of power) can be calculated as

$$10 \log_{10} \frac{P_2}{P_1} = 10 \log_{10} \frac{10 P_1}{P_1} = 10 \log_{10} 10 = 10(1) = 10 \text{ dB}$$

Example 3.28

One reason that engineers use the decibel to measure the changes in the strength of a signal is that decibel numbers can be added (or subtracted) when we are measuring several points (cascading) instead of just two. In Figure 3.27 a signal travels from point 1 to point 4. The signal is attenuated by the time it reaches point 2. Between points 2 and 3, the signal is amplified. Again, between points 3 and 4, the signal is attenuated. We can find the resultant decibel value for the signal just by adding the decibel measurements between each set of points.

Figure 3.27 *Decibels for Example 3.28*



In this case, the decibel value can be calculated as

$$\text{dB} = -3 + 7 - 3 = +1$$

The signal has gained in power.

Example 3.29

Sometimes the decibel is used to measure signal power in milliwatts. In this case, it is referred to as dB_m and is calculated as $\text{dB}_\text{m} = 10 \log_{10} P_m$, where P_m is the power in milliwatts. Calculate the power of a signal if its $\text{dB}_\text{m} = -30$.

Solution

We can calculate the power in the signal as

$$\text{dB}_\text{m} = 10 \log_{10} P_m = -30$$
$$\log_{10} P_m = -3 \qquad P_m = 10^{-3} \text{ mW}$$

Example 3.30

The loss in a cable is usually defined in decibels per kilometer (dB/km). If the signal at the beginning of a cable with −0.3 dB/km has a power of 2 mW, what is the power of the signal at 5 km?

Solution

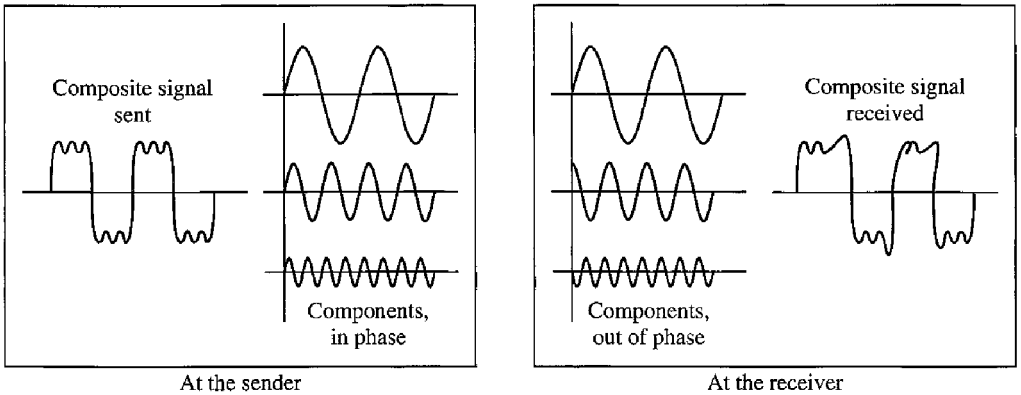
The loss in the cable in decibels is $5 \times (-0.3) = -1.5$ dB. We can calculate the power as

$$\text{dB} = 10 \log_{10} \frac{P_2}{P_1} = -1.5$$
$$\frac{P_2}{P_1} = 10^{-0.15} = 0.71$$
$$P_2 = 0.71 P_1 = 0.7 \times 2 = 1.4 \text{ mW}$$

Distortion

Distortion means that the signal changes its form or shape. Distortion can occur in a composite signal made of different frequencies. Each signal component has its own propagation speed (see the next section) through a medium and, therefore, its own delay in arriving at the final destination. Differences in delay may create a difference in phase if the delay is not exactly the same as the period duration. In other words, signal components at the receiver have phases different from what they had at the sender. The shape of the composite signal is therefore not the same. Figure 3.28 shows the effect of distortion on a composite signal.

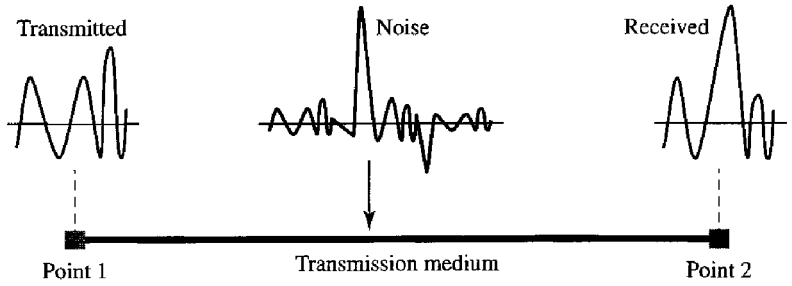
Figure 3.28 *Distortion*



Noise

Noise is another cause of impairment. Several types of noise, such as thermal noise, induced noise, crosstalk, and impulse noise, may corrupt the signal. Thermal noise is the random motion of electrons in a wire which creates an extra signal not originally sent by the transmitter. Induced noise comes from sources such as motors and appliances. These devices act as a sending antenna, and the transmission medium acts as the receiving antenna. Crosstalk is the effect of one wire on the other. One wire acts as a sending antenna and the other as the receiving antenna. Impulse noise is a spike (a signal with high energy in a very short time) that comes from power lines, lightning, and so on. Figure 3.29 shows the effect of noise on a signal. We discuss error in Chapter 10.

Figure 3.29 *Noise*



Signal-to-Noise Ratio (SNR)

As we will see later, to find the theoretical bit rate limit, we need to know the ratio of the signal power to the noise power. The **signal-to-noise ratio** is defined as

$$\text{SNR} = \frac{\text{average signal power}}{\text{average noise power}}$$

We need to consider the average signal power and the average noise power because these may change with time. Figure 3.30 shows the idea of SNR.

SNR is actually the ratio of what is wanted (signal) to what is not wanted (noise). A high SNR means the signal is less corrupted by noise; a low SNR means the signal is more corrupted by noise.

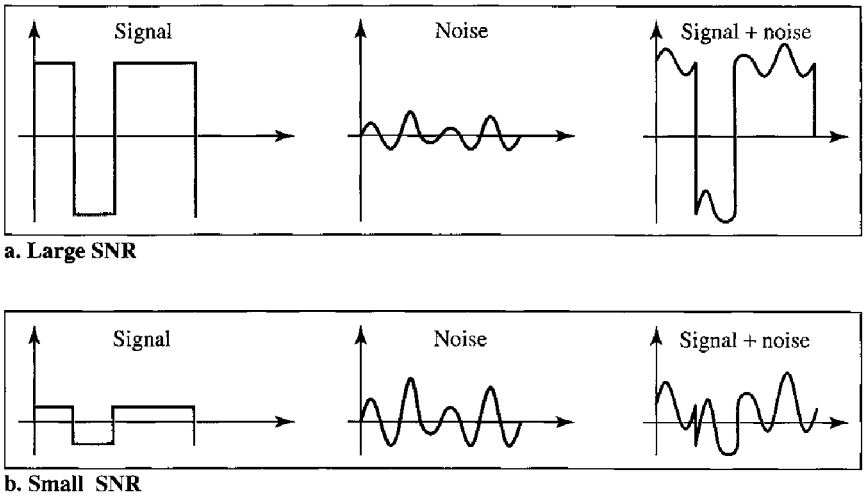
Because SNR is the ratio of two powers, it is often described in decibel units, SNR_{dB}, defined as

$$\text{SNR}_{\text{dB}} = 10 \log_{10} \text{SNR}$$

Example 3.31

The power of a signal is 10 mW and the power of the noise is 1 μW; what are the values of SNR and SNR_{dB}?

Figure 3.30 *Two cases of SNR: a high SNR and a low SNR*



Solution

The values of SNR and SNR_{dB} can be calculated as follows:

$$SNR = \frac{10{,}000\ \mu W}{1\ mW} = 10{,}000$$

$$SNR_{dB} = 10 \log_{10} 10{,}000 = 10 \log_{10} 10^4 = 40$$

Example 3.32

The values of SNR and SNR_{dB} for a noiseless channel are

$$SNR = \frac{\text{signal power}}{0} = \infty$$

$$SNR_{dB} = 10 \log_{10} \infty = \infty$$

We can never achieve this ratio in real life; it is an ideal.

3.5 DATA RATE LIMITS

A very important consideration in data communications is how fast we can send data, in bits per second, over a channel. Data rate depends on three factors:

1. The bandwidth available
2. The level of the signals we use
3. The quality of the channel (the level of noise)

Two theoretical formulas were developed to calculate the data rate: one by Nyquist for a noiseless channel, another by Shannon for a noisy channel.

Noiseless Channel: Nyquist Bit Rate

For a noiseless channel, the **Nyquist bit rate** formula defines the theoretical maximum bit rate

$$\text{BitRate} = 2 \times \text{bandwidth} \times \log_2 L$$

In this formula, bandwidth is the bandwidth of the channel, L is the number of signal levels used to represent data, and BitRate is the bit rate in bits per second.

According to the formula, we might think that, given a specific bandwidth, we can have any bit rate we want by increasing the number of signal levels. Although the idea is theoretically correct, practically there is a limit. When we increase the number of signal levels, we impose a burden on the receiver. If the number of levels in a signal is just 2, the receiver can easily distinguish between a 0 and a 1. If the level of a signal is 64, the receiver must be very sophisticated to distinguish between 64 different levels. In other words, increasing the levels of a signal reduces the reliability of the system.

Increasing the levels of a signal may reduce the reliability of the system.

Example 3.33

Does the Nyquist theorem bit rate agree with the intuitive bit rate described in baseband transmission?

Solution

They match when we have only two levels. We said, in baseband transmission, the bit rate is 2 times the bandwidth if we use only the first harmonic in the worst case. However, the Nyquist formula is more general than what we derived intuitively; it can be applied to baseband transmission and modulation. Also, it can be applied when we have two or more levels of signals.

Example 3.34

Consider a noiseless channel with a bandwidth of 3000 Hz transmitting a signal with two signal levels. The maximum bit rate can be calculated as

$$\text{BitRate} = 2 \times 3000 \times \log_2 2 = 6000 \text{ bps}$$

Example 3.35

Consider the same noiseless channel transmitting a signal with four signal levels (for each level, we send 2 bits). The maximum bit rate can be calculated as

$$\text{BitRate} = 2 \times 3000 \times \log_2 4 = 12{,}000 \text{ bps}$$

Example 3.36

We need to send 265 kbps over a noiseless channel with a bandwidth of 20 kHz. How many signal levels do we need?

Solution

We can use the Nyquist formula as shown:

$$265{,}000 = 2 \times 20{,}000 \times \log_2 L$$
$$\log_2 L = 6.625 \qquad L = 2^{6.625} = 98.7 \text{ levels}$$

Since this result is not a power of 2, we need to either increase the number of levels or reduce the bit rate. If we have 128 levels, the bit rate is 280 kbps. If we have 64 levels, the bit rate is 240 kbps.

Noisy Channel: Shannon Capacity

In reality, we cannot have a noiseless channel; the channel is always noisy. In 1944, Claude Shannon introduced a formula, called the **Shannon capacity,** to determine the theoretical highest data rate for a noisy channel:

$$\textbf{Capacity} = \textbf{bandwidth} \times \log_2 (1 + \text{SNR})$$

In this formula, bandwidth is the bandwidth of the channel, SNR is the signal-to-noise ratio, and capacity is the capacity of the channel in bits per second. Note that in the Shannon formula there is no indication of the signal level, which means that no matter how many levels we have, we cannot achieve a data rate higher than the capacity of the channel. In other words, the formula defines a characteristic of the channel, not the method of transmission.

Example 3.37

Consider an extremely noisy channel in which the value of the signal-to-noise ratio is almost zero. In other words, the noise is so strong that the signal is faint. For this channel the capacity C is calculated as

$$C = B \log_2 (1 + \text{SNR}) = B \log_2 (1 + 0) = B \log_2 1 = B \times 0 = 0$$

This means that the capacity of this channel is zero regardless of the bandwidth. In other words, we cannot receive any data through this channel.

Example 3.38

We can calculate the theoretical highest bit rate of a regular telephone line. A telephone line normally has a bandwidth of 3000 Hz (300 to 3300 Hz) assigned for data communications. The signal-to-noise ratio is usually 3162. For this channel the capacity is calculated as

$$C = B \log_2 (1 + \text{SNR}) = 3000 \log_2 (1 + 3162) = 3000 \log_2 3163$$
$$= 3000 \times 11.62 = 34{,}860 \text{ bps}$$

This means that the highest bit rate for a telephone line is 34.860 kbps. If we want to send data faster than this, we can either increase the bandwidth of the line or improve the signal-to-noise ratio.

Example 3.39

The signal-to-noise ratio is often given in decibels. Assume that $SNR_{dB} = 36$ and the channel bandwidth is 2 MHz. The theoretical channel capacity can be calculated as

$$SNR_{dB} = 10 \log_{10} SNR \quad \rightarrow \quad SNR = 10^{SNR_{dB}/10} \quad \rightarrow \quad SNR = 10^{3.6} = 3981$$
$$C = B \log_2 (1 + SNR) = 2 \times 10^6 \times \log_2 3982 = 24 \text{ Mbps}$$

Example 3.40

For practical purposes, when the SNR is very high, we can assume that SNR + 1 is almost the same as SNR. In these cases, the theoretical channel capacity can be simplified to

$$C = B \times \frac{SNR_{dB}}{3}$$

For example, we can calculate the theoretical capacity of the previous example as

$$C = 2 \text{ MHz} \times \frac{36}{3} = 24 \text{ Mbps}$$

Using Both Limits

In practice, we need to use both methods to find the limits and signal levels. Let us show this with an example.

Example 3.41

We have a channel with a 1-MHz bandwidth. The SNR for this channel is 63. What are the appropriate bit rate and signal level?

Solution

First, we use the Shannon formula to find the upper limit.

$$C = B \log_2 (1 + SNR) = 10^6 \log_2 (1 + 63) = 10^6 \log_2 64 = 6 \text{ Mbps}$$

The Shannon formula gives us 6 Mbps, the upper limit. For better performance we choose something lower, 4 Mbps, for example. Then we use the Nyquist formula to find the number of signal levels.

$$4 \text{ Mbps} = 2 \times 1 \text{ MHz} \times \log_2 L \quad \rightarrow \quad L = 4$$

The Shannon capacity gives us the upper limit;
the Nyquist formula tells us how many signal levels we need.

3.6 PERFORMANCE

Up to now, we have discussed the tools of transmitting data (signals) over a network and how the data behave. One important issue in networking is the performance of the network—how good is it? We discuss quality of service, an overall measurement of network performance, in greater detail in Chapter 24. In this section, we introduce terms that we need for future chapters.

Bandwidth

One characteristic that measures network performance is bandwidth. However, the term can be used in two different contexts with two different measuring values: bandwidth in hertz and bandwidth in bits per second.

Bandwidth in Hertz

We have discussed this concept. Bandwidth in hertz is the range of frequencies contained in a composite signal or the range of frequencies a channel can pass. For example, we can say the bandwidth of a subscriber telephone line is 4 kHz.

Bandwidth in Bits per Seconds

The term *bandwidth* can also refer to the number of bits per second that a channel, a link, or even a network can transmit. For example, one can say the bandwidth of a Fast Ethernet network (or the links in this network) is a maximum of 100 Mbps. This means that this network can send 100 Mbps.

Relationship

There is an explicit relationship between the bandwidth in hertz and bandwidth in bits per seconds. Basically, an increase in bandwidth in hertz means an increase in bandwidth in bits per second. The relationship depends on whether we have baseband transmission or transmission with modulation. We discuss this relationship in Chapters 4 and 5.

In networking, we use the term *bandwidth* in two contexts.

- The first, *bandwidth in hertz,* refers to the range of frequencies in a composite signal or the range of frequencies that a channel can pass.
- The second, *bandwidth in bits per second,* refers to the speed of bit transmission in a channel or link.

Example 3.42

The bandwidth of a subscriber line is 4 kHz for voice or data. The bandwidth of this line for data transmission can be up to 56,000 bps using a sophisticated modem to change the digital signal to analog.

Example 3.43

If the telephone company improves the quality of the line and increases the bandwidth to 8 kHz, we can send 112,000 bps by using the same technology as mentioned in Example 3.42.

Throughput

The **throughput** is a measure of how fast we can actually send data through a network. Although, at first glance, bandwidth in bits per second and throughput seem the same, they are different. A link may have a bandwidth of B bps, but we can only send T bps through this link with T always less than B. In other words, the bandwidth is a potential measurement of a link; the throughput is an actual measurement of how fast we can send data. For example, we may have a link with a bandwidth of 1 Mbps, but the devices connected to the end of the link may handle only 200 kbps. This means that we cannot send more than 200 kbps through this link.

Imagine a highway designed to transmit 1000 cars per minute from one point to another. However, if there is congestion on the road, this figure may be reduced to 100 cars per minute. The bandwidth is 1000 cars per minute; the throughput is 100 cars per minute.

Example 3.44

A network with bandwidth of 10 Mbps can pass only an average of 12,000 frames per minute with each frame carrying an average of 10,000 bits. What is the throughput of this network?

Solution

We can calculate the throughput as

$$\text{Throughput} = \frac{12{,}000 \times 10{,}000}{60} = 2 \text{ Mbps}$$

The throughput is almost one-fifth of the bandwidth in this case.

Latency (Delay)

The latency or delay defines how long it takes for an entire message to completely arrive at the destination from the time the first bit is sent out from the source. We can say that latency is made of four components: **propagation time, transmission time, queuing time** and **processing delay.**

$$\textbf{Latency} = \textbf{propagation time} + \textbf{transmission time} + \textbf{queuing time} + \textbf{processing delay}$$

Propagation Time

Propagation time measures the time required for a bit to travel from the source to the destination. The propagation time is calculated by dividing the distance by the propagation speed.

$$\textbf{Propagation time} = \frac{\textbf{Distance}}{\textbf{Propagation speed}}$$

The propagation speed of electromagnetic signals depends on the medium and on the frequency of the signal. For example, in a vacuum, light is propagated with a speed of 3×10^8 m/s. It is lower in air; it is much lower in cable.

Example 3.45

What is the propagation time if the distance between the two points is 12,000 km? Assume the propagation speed to be 2.4×10^8 m/s in cable.

Solution

We can calculate the propagation time as

$$\text{Propagation time} = \frac{12{,}000 \times 1000}{2.4 \times 10^8} = 50 \text{ ms}$$

The example shows that a bit can go over the Atlantic Ocean in only 50 ms if there is a direct cable between the source and the destination.

Transmission Time

In data communications we don't send just 1 bit, we send a message. The first bit may take a time equal to the propagation time to reach its destination; the last bit also may take the same amount of time. However, there is a time between the first bit leaving the sender and the last bit arriving at the receiver. The first bit leaves earlier and arrives earlier; the last bit leaves later and arrives later. The time required for transmission of a message depends on the size of the message and the bandwidth of the channel.

$$\textbf{Transmission time} = \frac{\textbf{Message size}}{\textbf{Bandwidth}}$$

Example 3.46

What are the propagation time and the transmission time for a 2.5-kbyte message (an e-mail) if the bandwidth of the network is 1 Gbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at 2.4×10^8 m/s.

Solution

We can calculate the propagation and transmission time as

$$\text{Propagation time} = \frac{12{,}000 \times 1000}{2.4 \times 10^8} = 50 \text{ ms}$$

$$\text{Transmission time} = \frac{2500 \times 8}{10^9} = 0.020 \text{ ms}$$

Note that in this case, because the message is short and the bandwidth is high, the dominant factor is the propagation time, not the transmission time. The transmission time can be ignored.

Example 3.47

What are the propagation time and the transmission time for a 5-Mbyte message (an image) if the bandwidth of the network is 1 Mbps? Assume that the distance between the sender and the receiver is 12,000 km and that light travels at 2.4×10^8 m/s.

Solution

We can calculate the propagation and transmission times as

$$\text{Propagation time} = \frac{12{,}000 \times 1000}{2.4 \times 10^8} = 50 \text{ ms}$$

$$\text{Transmission time} = \frac{5{,}000{,}000 \times 8}{10^6} = 40 \text{ s}$$

Note that in this case, because the message is very long and the bandwidth is not very high, the dominant factor is the transmission time, not the propagation time. The propagation time can be ignored.

Queuing Time

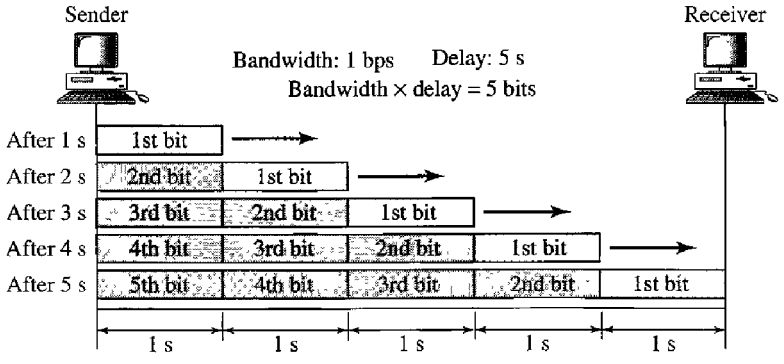
The third component in latency is the queuing time, the time needed for each intermediate or end device to hold the message before it can be processed. The queuing time is not a fixed factor; it changes with the load imposed on the network. When there is heavy traffic on the network, the queuing time increases. An intermediate device, such as a router, queues the arrived messages and processes them one by one. If there are many messages, each message will have to wait.

Bandwidth-Delay Product

Bandwidth and delay are two performance metrics of a link. However, as we will see in this chapter and future chapters, what is very important in data communications is the product of the two, the bandwidth-delay product. Let us elaborate on this issue, using two hypothetical cases as examples.

❑ **Case 1.** Figure 3.31 shows case 1.

Figure 3.31 *Filling the link with bits for case 1*

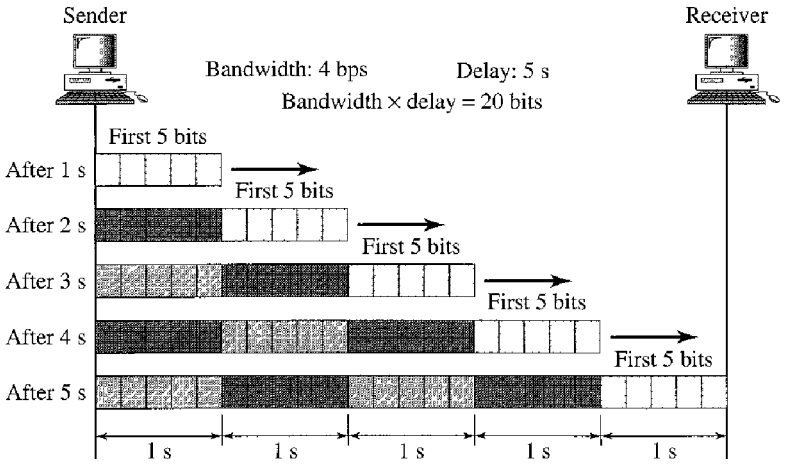


Let us assume that we have a link with a bandwidth of 1 bps (unrealistic, but good for demonstration purposes). We also assume that the delay of the link is 5 s (also unrealistic). We want to see what the bandwidth-delay product means in this case.

Looking at figure, we can say that this product 1×5 is the maximum number of bits that can fill the link. There can be no more than 5 bits at any time on the link.

- **Case 2.** Now assume we have a bandwidth of 4 bps. Figure 3.32 shows that there can be maximum $4 \times 5 = 20$ bits on the line. The reason is that, at each second, there are 4 bits on the line; the duration of each bit is 0.25 s.

Figure 3.32 *Filling the link with bits in case 2*



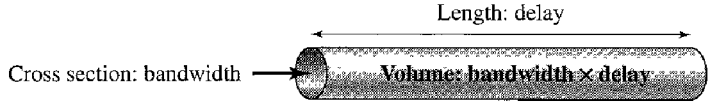
The above two cases show that the product of bandwidth and delay is the number of bits that can fill the link. This measurement is important if we need to send data in bursts and wait for the acknowledgment of each burst before sending the next one. To use the maximum capability of the link, we need to make the size of our burst 2 times the product of bandwidth and delay; we need to fill up the full-duplex channel (two directions). The sender should send a burst of data of ($2 \times$ bandwidth $\times$ delay) bits. The sender then waits for receiver acknowledgment for part of the burst before sending another burst. The amount $2 \times$ bandwidth $\times$ delay is the number of bits that can be in transition at any time.

The bandwidth-delay product defines the number of bits that can fill the link.

Example 3.48

We can think about the link between two points as a pipe. The cross section of the pipe represents the bandwidth, and the length of the pipe represents the delay. We can say the volume of the pipe defines the bandwidth-delay product, as shown in Figure 3.33.

Figure 3.33 *Concept of bandwidth-delay product*



Jitter

Another performance issue that is related to delay is **jitter.** We can roughly say that jitter is a problem if different packets of data encounter different delays and the application using the data at the receiver site is time-sensitive (audio and video data, for example). If the delay for the first packet is 20 ms, for the second is 45 ms, and for the third is 40 ms, then the real-time application that uses the packets endures jitter. We discuss jitter in greater detail in Chapter 29.

3.7 RECOMMENDED READING

For more details about subjects discussed in this chapter, we recommend the following books. The items in brackets [. . .] refer to the reference list at the end of the text.

Books

Data and signals are elegantly discussed in Chapters 1 to 6 of [Pea92]. [Cou01] gives an excellent coverage about signals in Chapter 2. More advanced materials can be found in [Ber96]. [Hsu03] gives a good mathematical approach to signaling. Complete coverage of Fourier Analysis can be found in [Spi74]. Data and signals are discussed in Chapter 3 of [Sta04] and Section 2.1 of [Tan03].

3.8 KEY TERMS

analog
analog data
analog signal
attenuation
bandpass channel
bandwidth
baseband transmission
bit rate
bits per second (bps)
broadband transmission
composite signal
cycle
decibel (dB)
digital
digital data
digital signal
distortion
Fourier analysis
frequency
frequency-domain
fundamental frequency
harmonic
Hertz (Hz)
jitter
low-pass channel
noise
nonperiodic signal
Nyquist bit rate
peak amplitude
period
periodic signal
phase
processing delay
propagation speed

propagation time
queuing time
Shannon capacity
signal
signal-to-noise ratio (SNR)
sine wave
throughput
time-domain
transmission time
wavelength

3.9 SUMMARY

- Data must be transformed to electromagnetic signals to be transmitted.
- Data can be analog or digital. Analog data are continuous and take continuous values. Digital data have discrete states and take discrete values.
- Signals can be analog or digital. Analog signals can have an infinite number of values in a range; digital signals can have only a limited number of values.
- In data communications, we commonly use periodic analog signals and nonperiodic digital signals.
- Frequency and period are the inverse of each other.
- Frequency is the rate of change with respect to time.
- Phase describes the position of the waveform relative to time 0.
- A complete sine wave in the time domain can be represented by one single spike in the frequency domain.
- A single-frequency sine wave is not useful in data communications; we need to send a composite signal, a signal made of many simple sine waves.
- According to Fourier analysis, any composite signal is a combination of simple sine waves with different frequencies, amplitudes, and phases.
- The bandwidth of a composite signal is the difference between the highest and the lowest frequencies contained in that signal.
- A digital signal is a composite analog signal with an infinite bandwidth.
- Baseband transmission of a digital signal that preserves the shape of the digital signal is possible only if we have a low-pass channel with an infinite or very wide bandwidth.
- If the available channel is a bandpass channel, we cannot send a digital signal directly to the channel; we need to convert the digital signal to an analog signal before transmission.
- For a noiseless channel, the Nyquist bit rate formula defines the theoretical maximum bit rate. For a noisy channel, we need to use the Shannon capacity to find the maximum bit rate.
- Attenuation, distortion, and noise can impair a signal.
- Attenuation is the loss of a signal's energy due to the resistance of the medium.
- Distortion is the alteration of a signal due to the differing propagation speeds of each of the frequencies that make up a signal.
- Noise is the external energy that corrupts a signal.
- The bandwidth-delay product defines the number of bits that can fill the link.

3.10 PRACTICE SET

Review Questions

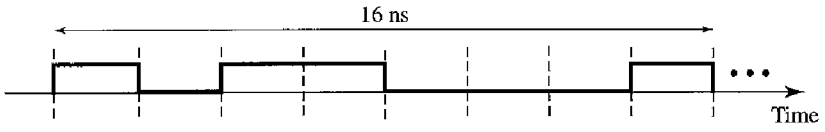
1. What is the relationship between period and frequency?
2. What does the amplitude of a signal measure? What does the frequency of a signal measure? What does the phase of a signal measure?
3. How can a composite signal be decomposed into its individual frequencies?
4. Name three types of transmission impairment.
5. Distinguish between baseband transmission and broadband transmission.
6. Distinguish between a low-pass channel and a band-pass channel.
7. What does the Nyquist theorem have to do with communications?
8. What does the Shannon capacity have to do with communications?
9. Why do optical signals used in fiber optic cables have a very short wave length?
10. Can we say if a signal is periodic or nonperiodic by just looking at its frequency domain plot? How?
11. Is the frequency domain plot of a voice signal discrete or continuous?
12. Is the frequency domain plot of an alarm system discrete or continuous?
13. We send a voice signal from a microphone to a recorder. Is this baseband or broadband transmission?
14. We send a digital signal from one station on a LAN to another station. Is this baseband or broadband transmission?
15. We modulate several voice signals and send them through the air. Is this baseband or broadband transmission?

Exercises

16. Given the frequencies listed below, calculate the corresponding periods.
    a. 24 Hz
    b. 8 MHz
    c. 140 KHz
17. Given the following periods, calculate the corresponding frequencies.
    a. 5 s
    b. 12 μs
    c. 220 ns
18. What is the phase shift for the following?
    a. A sine wave with the maximum amplitude at time zero
    b. A sine wave with maximum amplitude after 1/4 cycle
    c. A sine wave with zero amplitude after 3/4 cycle and increasing
19. What is the bandwidth of a signal that can be decomposed into five sine waves with frequencies at 0, 20, 50, 100, and 200 Hz? All peak amplitudes are the same. Draw the bandwidth.

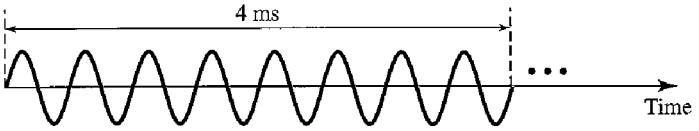
20. A periodic composite signal with a bandwidth of 2000 Hz is composed of two sine waves. The first one has a frequency of 100 Hz with a maximum amplitude of 20 V; the second one has a maximum amplitude of 5 V. Draw the bandwidth.
21. Which signal has a wider bandwidth, a sine wave with a frequency of 100 Hz or a sine wave with a frequency of 200 Hz?
22. What is the bit rate for each of the following signals?
    a. A signal in which 1 bit lasts 0.001 s
    b. A signal in which 1 bit lasts 2 ms
    c. A signal in which 10 bits last 20 μs
23. A device is sending out data at the rate of 1000 bps.
    a. How long does it take to send out 10 bits?
    b. How long does it take to send out a single character (8 bits)?
    c. How long does it take to send a file of 100,000 characters?
24. What is the bit rate for the signal in Figure 3.34?

Figure 3.34 *Exercise 24*



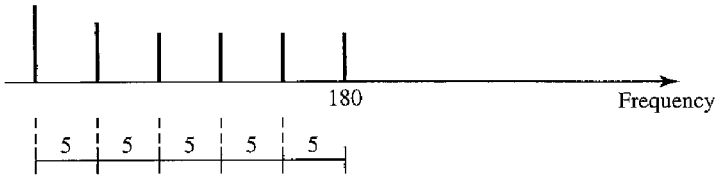
25. What is the frequency of the signal in Figure 3.35?

Figure 3.35 *Exercise 25*



26. What is the bandwidth of the composite signal shown in Figure 3.36.

Figure 3.36 *Exercise 26*



27. A periodic composite signal contains frequencies from 10 to 30 KHz, each with an amplitude of 10 V. Draw the frequency spectrum.
28. A non-periodic composite signal contains frequencies from 10 to 30 KHz. The peak amplitude is 10 V for the lowest and the highest signals and is 30 V for the 20-KHz signal. Assuming that the amplitudes change gradually from the minimum to the maximum, draw the frequency spectrum.
29. A TV channel has a bandwidth of 6 MHz. If we send a digital signal using one channel, what are the data rates if we use one harmonic, three harmonics, and five harmonics?
30. A signal travels from point A to point B. At point A, the signal power is 100 W. At point B, the power is 90 W. What is the attenuation in decibels?
31. The attenuation of a signal is –10 dB. What is the final signal power if it was originally 5 W?
32. A signal has passed through three cascaded amplifiers, each with a 4 dB gain. What is the total gain? How much is the signal amplified?
33. If the bandwidth of the channel is 5 Kbps, how long does it take to send a frame of 100,000 bits out of this device?
34. The light of the sun takes approximately eight minutes to reach the earth. What is the distance between the sun and the earth?
35. A signal has a wavelength of 1 μm in air. How far can the front of the wave travel during 1000 periods?
36. A line has a signal-to-noise ratio of 1000 and a bandwidth of 4000 KHz. What is the maximum data rate supported by this line?
37. We measure the performance of a telephone line (4 KHz of bandwidth). When the signal is 10 V, the noise is 5 mV. What is the maximum data rate supported by this telephone line?
38. A file contains 2 million bytes. How long does it take to download this file using a 56-Kbps channel? 1-Mbps channel?
39. A computer monitor has a resolution of 1200 by 1000 pixels. If each pixel uses 1024 colors, how many bits are needed to send the complete contents of a screen?
40. A signal with 200 milliwatts power passes through 10 devices, each with an average noise of 2 microwatts. What is the SNR? What is the SNR_{dB}?
41. If the peak voltage value of a signal is 20 times the peak voltage value of the noise, what is the SNR? What is the SNR_{dB}?
42. What is the theoretical capacity of a channel in each of the following cases:
    a. Bandwidth: 20 KHz $SNR_{dB} = 40$
    b. Bandwidth: 200 KHz $SNR_{dB} = 4$
    c. Bandwidth: 1 MHz $SNR_{dB} = 20$
43. We need to upgrade a channel to a higher bandwidth. Answer the following questions:
    a. How is the rate improved if we double the bandwidth?
    b. How is the rate improved if we double the SNR?

44. We have a channel with 4 KHz bandwidth. If we want to send data at 100 Kbps, what is the minimum SNR_{dB}? What is SNR?
45. What is the transmission time of a packet sent by a station if the length of the packet is 1 million bytes and the bandwidth of the channel is 200 Kbps?
46. What is the length of a bit in a channel with a propagation speed of 2×10^8 m/s if the channel bandwidth is
    a. 1 Mbps?
    b. 10 Mbps?
    c. 100 Mbps?
47. How many bits can fit on a link with a 2 ms delay if the bandwidth of the link is
    a. 1 Mbps?
    b. 10 Mbps?
    c. 100 Mbps?
48. What is the total delay (latency) for a frame of size 5 million bits that is being sent on a link with 10 routers each having a queuing time of 2 μs and a processing time of 1 μs. The length of the link is 2000 Km. The speed of light inside the link is 2×10^8 m/s. The link has a bandwidth of 5 Mbps. Which component of the total delay is dominant? Which one is negligible?

Down

CHAPTER 4

Digital Transmission

A computer network is designed to send information from one point to another. This information needs to be converted to either a digital signal or an analog signal for transmission. In this chapter, we discuss the first choice, conversion to digital signals; in Chapter 5, we discuss the second choice, conversion to analog signals.

We discussed the advantages and disadvantages of digital transmission over analog transmission in Chapter 3. In this chapter, we show the schemes and techniques that we use to transmit data digitally. First, we discuss **digital-to-digital conversion** techniques, methods which convert digital data to digital signals. Second, we discuss **analog-to-digital conversion** techniques, methods which change an analog signal to a digital signal. Finally, we discuss **transmission modes.**

4.1 DIGITAL-TO-DIGITAL CONVERSION

In Chapter 3, we discussed data and signals. We said that data can be either digital or analog. We also said that signals that represent data can also be digital or analog. In this section, we see how we can represent digital data by using digital signals. The conversion involves three techniques: line coding, block coding, and scrambling. Line coding is always needed; block coding and scrambling may or may not be needed.

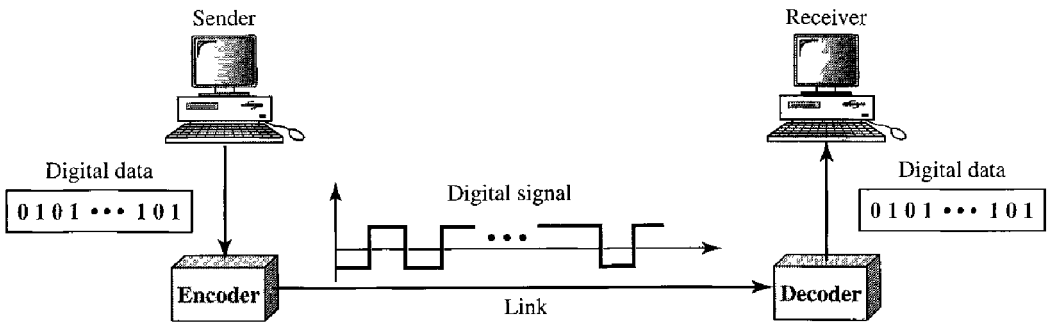
Line Coding

Line coding is the process of converting digital data to digital signals. We assume that data, in the form of text, numbers, graphical images, audio, or video, are stored in computer memory as sequences of bits (see Chapter 1). Line coding converts a sequence of bits to a digital signal. At the sender, digital data are encoded into a digital signal; at the receiver, the digital data are recreated by decoding the digital signal. Figure 4.1 shows the process.

Characteristics

Before discussing different line coding schemes, we address their common characteristics.

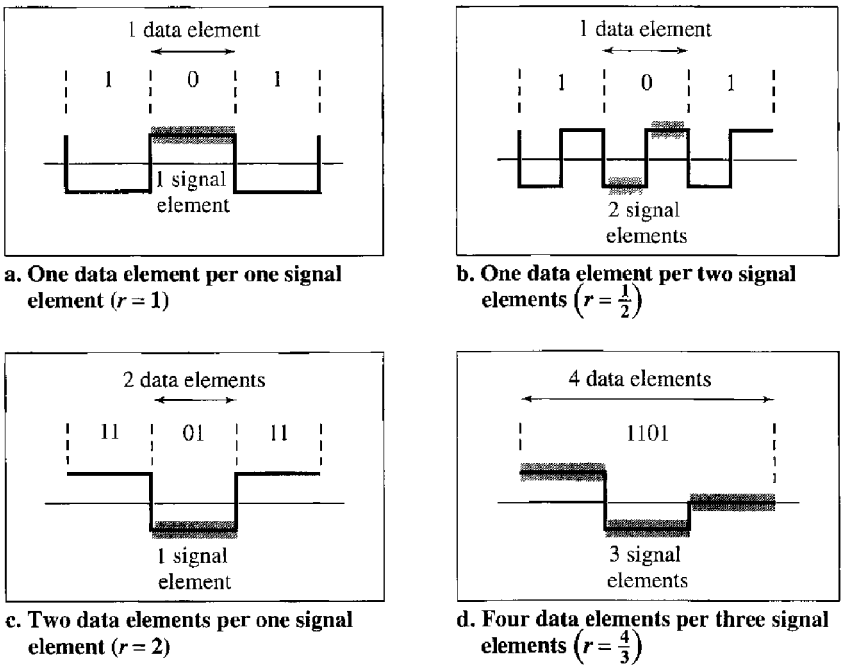
Figure 4.1 *Line coding and decoding*



Signal Element Versus Data Element Let us distinguish between a **data element** and a **signal element.** In data communications, our goal is to send data elements. A data element is the smallest entity that can represent a piece of information: this is the bit. In digital data communications, a signal element carries data elements. A signal element is the shortest unit (timewise) of a digital signal. In other words, data elements are what we need to send; signal elements are what we can send. Data elements are being carried; signal elements are the carriers.

We define a ratio r which is the number of data elements carried by each signal element. Figure 4.2 shows several situations with different values of r.

Figure 4.2 *Signal element versus data element*



a. One data element per one signal element ($r = 1$)

b. One data element per two signal elements ($r = \frac{1}{2}$)

c. Two data elements per one signal element ($r = 2$)

d. Four data elements per three signal elements ($r = \frac{4}{3}$)

In part a of the figure, one data element is carried by one signal element ($r = 1$). In part b of the figure, we need two signal elements (two transitions) to carry each data

element ($r = \frac{1}{2}$). We will see later that the extra signal element is needed to guarantee synchronization. In part c of the figure, a signal element carries two data elements ($r = 2$). Finally, in part d, a group of 4 bits is being carried by a group of three signal elements ($r = \frac{4}{3}$). For every line coding scheme we discuss, we will give the value of r.

An analogy may help here. Suppose each data element is a person who needs to be carried from one place to another. We can think of a signal element as a vehicle that can carry people. When $r = 1$, it means each person is driving a vehicle. When $r > 1$, it means more than one person is travelling in a vehicle (a carpool, for example). We can also have the case where one person is driving a car and a trailer ($r = \frac{1}{2}$).

Data Rate Versus Signal Rate The **data rate** defines the number of data elements (bits) sent in 1s. The unit is bits per second (bps). The **signal rate** is the number of signal elements sent in 1s. The unit is the baud. There are several common terminologies used in the literature. The data rate is sometimes called the **bit rate;** the signal rate is sometimes called the **pulse rate,** the **modulation rate,** or the **baud rate.**

One goal in data communications is to increase the data rate while decreasing the signal rate. Increasing the data rate increases the speed of transmission; decreasing the signal rate decreases the bandwidth requirement. In our vehicle-people analogy, we need to carry more people in fewer vehicles to prevent traffic jams. We have a limited *bandwidth* in our transportation system.

We now need to consider the relationship between data rate and signal rate (bit rate and baud rate). This relationship, of course, depends on the value of r. It also depends on the data pattern. If we have a data pattern of all 1s or all 0s, the signal rate may be different from a data pattern of alternating 0s and 1s. To derive a formula for the relationship, we need to define three cases: the worst, best, and average. The worst case is when we need the maximum signal rate; the best case is when we need the minimum. In data communications, we are usually interested in the average case. We can formulate the relationship between data rate and signal rate as

$$S = c \times N \times \frac{1}{r} \quad \text{baud}$$

where N is the data rate (bps); c is the case factor, which varies for each case; S is the number of signal elements; and r is the previously defined factor.

Example 4.1

A signal is carrying data in which one data element is encoded as one signal element ($r = 1$). If the bit rate is 100 kbps, what is the average value of the baud rate if c is between 0 and 1?

Solution

We assume that the average value of c is $\frac{1}{2}$. The baud rate is then

$$S = c \times N \times \frac{1}{r} = \frac{1}{2} \times 100{,}000 \times \frac{1}{1} = 50{,}000 = 50 \text{ kbaud}$$

Bandwidth We discussed in Chapter 3 that a digital signal that carries information is nonperiodic. We also showed that the bandwidth of a nonperiodic signal is continuous with an infinite range. However, most digital signals we encounter in real life have a

bandwidth with finite values. In other words, the bandwidth is theoretically infinite, but many of the components have such a small amplitude that they can be ignored. The effective bandwidth is finite. From now on, when we talk about the bandwidth of a digital signal, we need to remember that we are talking about this effective bandwidth.

Although the actual bandwidth of a digital signal is infinite, the effective bandwidth is finite.

We can say that the baud rate, not the bit rate, determines the required bandwidth for a digital signal. If we use the transportation analogy, the number of vehicles affects the traffic, not the number of people being carried. More changes in the signal mean injecting more frequencies into the signal. (Recall that frequency means change and change means frequency.) The bandwidth reflects the range of frequencies we need. There is a relationship between the baud rate (signal rate) and the bandwidth. Bandwidth is a complex idea. When we talk about the bandwidth, we normally define a range of frequencies. We need to know where this range is located as well as the values of the lowest and the highest frequencies. In addition, the amplitude (if not the phase) of each component is an important issue. In other words, we need more information about the bandwidth than just its value; we need a diagram of the bandwidth. We will show the bandwidth for most schemes we discuss in the chapter. For the moment, we can say that the bandwidth (range of frequencies) is proportional to the signal rate (baud rate). The minimum bandwidth can be given as

$$B_{\text{min}} = c \times N \times \frac{1}{r}$$

We can solve for the maximum data rate if the bandwidth of the channel is given.

$$N_{\text{max}} = \frac{1}{c} \times B \times r$$

Example 4.2

The maximum data rate of a channel (see Chapter 3) is $N_{\text{max}} = 2 \times B \times \log_2 L$ (defined by the Nyquist formula). Does this agree with the previous formula for N_{max}?

Solution

A signal with L levels actually can carry $\log_2 L$ bits per level. If each level corresponds to one signal element and we assume the average case ($c = \frac{1}{2}$), then we have

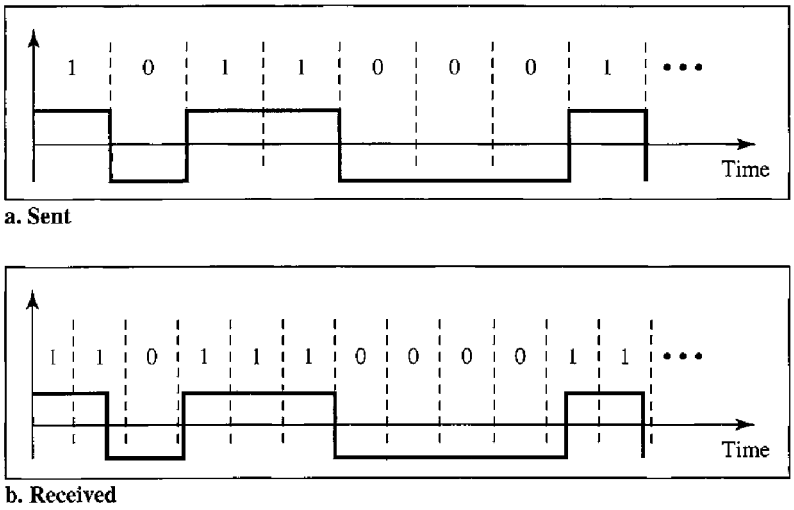
$$N_{\text{max}} = \frac{1}{c} \times B \times r = 2 \times B \times \log_2 L$$

Baseline Wandering In decoding a digital signal, the receiver calculates a running average of the received signal power. This average is called the ***baseline.*** The incoming signal power is evaluated against this baseline to determine the value of the data element. A long string of 0s or 1s can cause a drift in the baseline (**baseline wandering**) and make it difficult for the receiver to decode correctly. A good line coding scheme needs to prevent baseline wandering.

DC Components When the voltage level in a digital signal is constant for a while, the spectrum creates very low frequencies (results of Fourier analysis). These frequencies around zero, called DC (direct-current) *components,* present problems for a system that cannot pass low frequencies or a system that uses electrical coupling (via a transformer). For example, a telephone line cannot pass frequencies below 200 Hz. Also a long-distance link may use one or more transformers to isolate different parts of the line electrically. For these systems, we need a scheme with no **DC component.**

Self-synchronization To correctly interpret the signals received from the sender, the receiver's bit intervals must correspond exactly to the sender's bit intervals. If the receiver clock is faster or slower, the bit intervals are not matched and the receiver might misinterpret the signals. Figure 4.3 shows a situation in which the receiver has a shorter bit duration. The sender sends 10110001, while the receiver receives 110111000011.

Figure 4.3 *Effect of lack of synchronization*



A **self-synchronizing** digital signal includes timing information in the data being transmitted. This can be achieved if there are transitions in the signal that alert the receiver to the beginning, middle, or end of the pulse. If the receiver's clock is out of synchronization, these points can reset the clock.

Example 4.3

In a digital transmission, the receiver clock is 0.1 percent faster than the sender clock. How many extra bits per second does the receiver receive if the data rate is 1 kbps? How many if the data rate is 1 Mbps?

Solution

At 1 kbps, the receiver receives 1001 bps instead of 1000 bps.

1000 bits sent 1001 bits received 1 extra bps

At 1 Mbps, the receiver receives 1,001,000 bps instead of 1,000,000 bps.

1,000,000 bits sent 1,001,000 bits received 1000 extra bps

Built-in Error Detection It is desirable to have a built-in error-detecting capability in the generated code to detect some of or all the errors that occurred during transmission. Some encoding schemes that we will discuss have this capability to some extent.

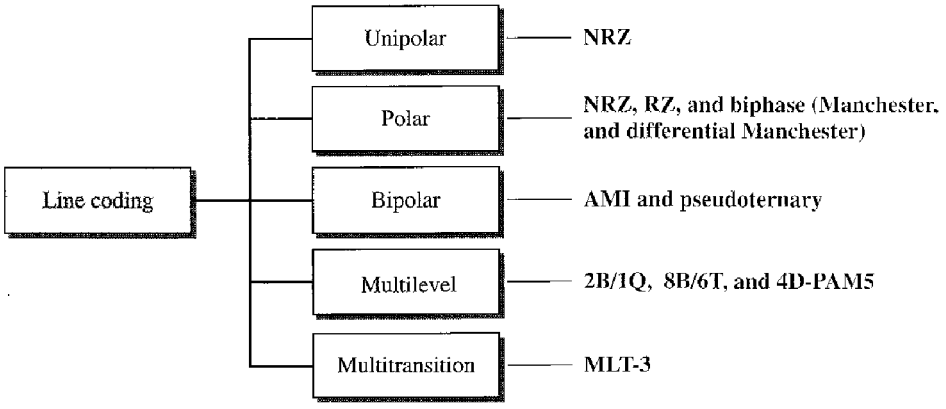
Immunity to Noise and Interference Another desirable code characteristic is a code that is immune to noise and other interferences. Some encoding schemes that we will discuss have this capability.

Complexity A complex scheme is more costly to implement than a simple one. For example, a scheme that uses four signal levels is more difficult to interpret than one that uses only two levels.

Line Coding Schemes

We can roughly divide line coding schemes into five broad categories, as shown in Figure 4.4.

Figure 4.4 *Line coding schemes*



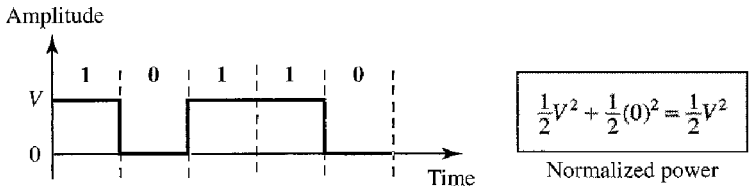
There are several schemes in each category. We need to be familiar with all schemes discussed in this section to understand the rest of the book. This section can be used as a reference for schemes encountered later.

Unipolar Scheme

In a unipolar scheme, all the signal levels are on one side of the time axis, either above or below.

NRZ (Non-Return-to-Zero) Traditionally, a unipolar scheme was designed as a **non-return-to-zero (NRZ)** scheme in which the positive voltage defines bit 1 and the zero voltage defines bit 0. It is called NRZ because the signal does not return to zero at the middle of the bit. Figure 4.5 show a unipolar NRZ scheme.

Figure 4.5 *Unipolar NRZ scheme*



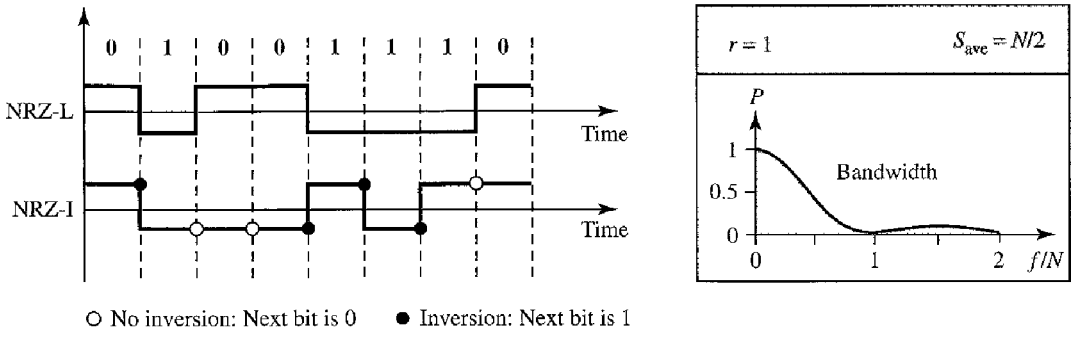
Compared with its polar counterpart (see the next section), this scheme is very costly. As we will see shortly, the normalized power (power needed to send 1 bit per unit line resistance) is double that for polar NRZ. For this reason, this scheme is normally not used in data communications today.

Polar Schemes

In polar schemes, the voltages are on the both sides of the time axis. For example, the voltage level for 0 can be positive and the voltage level for 1 can be negative.

Non-Return-to-Zero (NRZ) In **polar NRZ** encoding, we use two levels of voltage amplitude. We can have two versions of polar NRZ: NRZ-L and NRZ-I, as shown in Figure 4.6. The figure also shows the value of r, the average baud rate, and the bandwidth. In the first variation, NRZ-L (**NRZ-Level**), the level of the voltage determines the value of the bit. In the second variation, NRZ-I (**NRZ-Invert**), the change or lack of change in the level of the voltage determines the value of the bit. If there is no change, the bit is 0; if there is a change, the bit is 1.

Figure 4.6 *Polar NRZ-L and NRZ-I schemes*



In NRZ-L the level of the voltage determines the value of the bit. In NRZ-I the inversion or the lack of inversion determines the value of the bit.

Let us compare these two schemes based on the criteria we previously defined. Although baseline wandering is a problem for both variations, it is twice as severe in NRZ-L. If there is a long sequence of 0s or 1s in NRZ-L, the average signal power

becomes skewed. The receiver might have difficulty discerning the bit value. In NRZ-I this problem occurs only for a long sequence of 0s. If somehow we can eliminate the long sequence of 0s, we can avoid baseline wandering. We will see shortly how this can be done.

The synchronization problem (sender and receiver clocks are not synchronized) also exists in both schemes. Again, this problem is more serious in NRZ-L than in NRZ-I. While a long sequence of 0s can cause a problem in both schemes, a long sequence of 1s affects only NRZ-L.

Another problem with NRZ-L occurs when there is a sudden change of polarity in the system. For example, if twisted-pair cable is the medium, a change in the polarity of the wire results in all 0s interpreted as 1s and all 1s interpreted as 0s. NRZ-I does not have this problem. Both schemes have an average signal rate of $N/2$ Bd.

NRZ-L and NRZ-I both have an average signal rate of $N/2$ Bd.

Let us discuss the bandwidth. Figure 4.6 also shows the normalized bandwidth for both variations. The vertical axis shows the power density (the power for each 1 Hz of bandwidth); the horizontal axis shows the frequency. The bandwidth reveals a very serious problem for this type of encoding. The value of the power density is very high around frequencies close to zero. This means that there are DC components that carry a high level of energy. As a matter of fact, most of the energy is concentrated in frequencies between 0 and $N/2$. This means that although the average of the signal rate is $N/2$, the energy is not distributed evenly between the two halves.

NRZ-L and NRZ-I both have a DC component problem.

Example 4.4

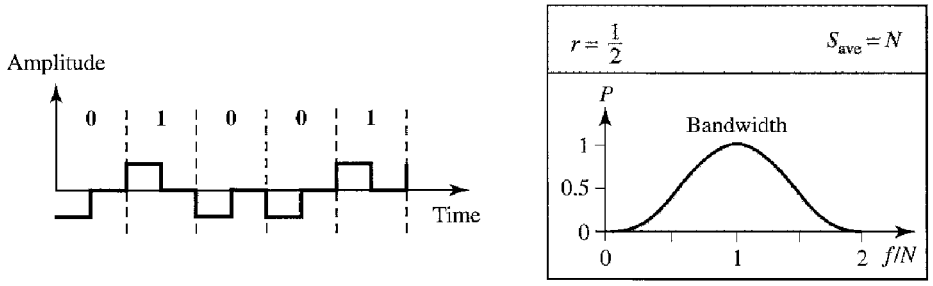
A system is using NRZ-I to transfer 10-Mbps data. What are the average signal rate and minimum bandwidth?

Solution

The average signal rate is $S = N/2 = 500$ kbaud. The minimum bandwidth for this average baud rate is $B_{min} = S = 500$ kHz.

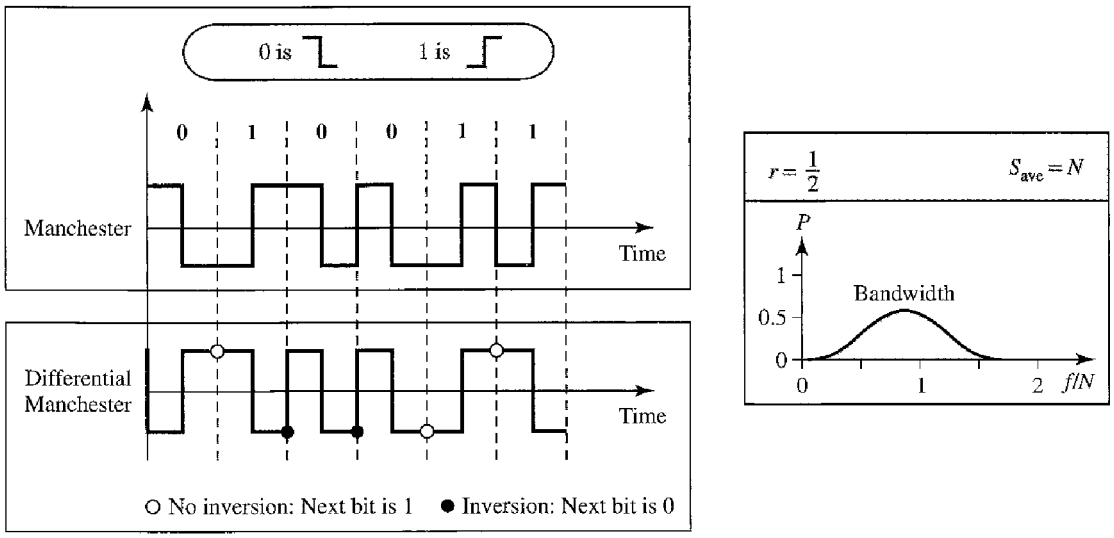
Return to Zero (RZ) The main problem with NRZ encoding occurs when the sender and receiver clocks are not synchronized. The receiver does not know when one bit has ended and the next bit is starting. One solution is the **return-to-zero (RZ)** scheme, which uses three values: positive, negative, and zero. In RZ, the signal changes not between bits but during the bit. In Figure 4.7 we see that the signal goes to 0 in the middle of each bit. It remains there until the beginning of the next bit. The main disadvantage of RZ encoding is that it requires two signal changes to encode a bit and therefore occupies greater bandwidth. The same problem we mentioned, a sudden change of polarity resulting in all 0s interpreted as 1s and all 1s interpreted as 0s, still exist here, but there is no DC component problem. Another problem is the complexity: RZ uses three levels of voltage, which is more complex to create and discern. As a result of all these deficiencies, the scheme is not used today. Instead, it has been replaced by the better-performing Manchester and differential Manchester schemes (discussed next).

Figure 4.7 *Polar RZ scheme*



Biphase: Manchester and Differential Manchester The idea of RZ (transition at the middle of the bit) and the idea of NRZ-L are combined into the **Manchester** scheme. In Manchester encoding, the duration of the bit is divided into two halves. The voltage remains at one level during the first half and moves to the other level in the second half. The transition at the middle of the bit provides synchronization. **Differential Manchester,** on the other hand, combines the ideas of RZ and NRZ-I. There is always a transition at the middle of the bit, but the bit values are determined at the beginning of the bit. If the next bit is 0, there is a transition; if the next bit is 1, there is none. Figure 4.8 shows both Manchester and differential Manchester encoding.

Figure 4.8 *Polar biphase: Manchester and differential Manchester schemes*



In Manchester and differential Manchester encoding, the transition at the middle of the bit is used for synchronization.

The Manchester scheme overcomes several problems associated with NRZ-L, and differential Manchester overcomes several problems associated with NRZ-I. First, there is no baseline wandering. There is no DC component because each bit has a positive and

negative voltage contribution. The only drawback is the signal rate. The signal rate for Manchester and differential Manchester is double that for NRZ. The reason is that there is always one transition at the middle of the bit and maybe one transition at the end of each bit. Figure 4.8 shows both Manchester and differential Manchester encoding schemes. Note that Manchester and differential Manchester schemes are also called **biphase** schemes.

The minimum bandwidth of Manchester and differential Manchester is 2 times that of NRZ.

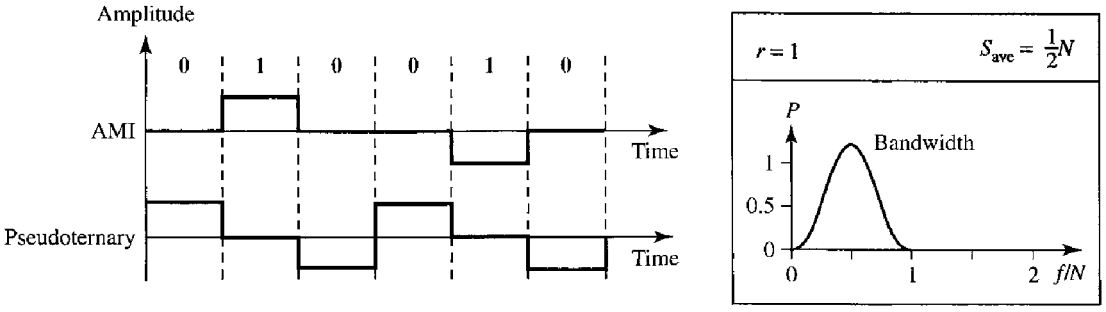
Bipolar Schemes

In **bipolar** encoding (sometimes called ***multilevel binary***), there are three voltage levels: positive, negative, and zero. The voltage level for one data element is at zero, while the voltage level for the other element alternates between positive and negative.

In bipolar encoding, we use three levels: positive, zero, and negative.

AMI and Pseudoternary Figure 4.9 shows two variations of bipolar encoding: AMI and pseudoternary. A common bipolar encoding scheme is called bipolar **alternate mark inversion (AMI).** In the term *alternate mark inversion,* the word *mark* comes from telegraphy and means 1. So AMI means alternate 1 inversion. A neutral zero voltage represents binary 0. Binary 1s are represented by alternating positive and negative voltages. A variation of AMI encoding is called **pseudoternary** in which the 1 bit is encoded as a zero voltage and the 0 bit is encoded as alternating positive and negative voltages.

Figure 4.9 *Bipolar schemes: AMI and pseudoternary*



The bipolar scheme was developed as an alternative to NRZ. The bipolar scheme has the same signal rate as NRZ, but there is no DC component. The NRZ scheme has most of its energy concentrated near zero frequency, which makes it unsuitable for transmission over channels with poor performance around this frequency. The concentration of the energy in bipolar encoding is around frequency $N/2$. Figure 4.9 shows the typical energy concentration for a bipolar scheme.

One may ask why we do not have DC component in bipolar encoding. We can answer this question by using the Fourier transform, but we can also think about it intuitively. If we have a long sequence of 1s, the voltage level alternates between positive and negative; it is not constant. Therefore, there is no DC component. For a long sequence of 0s, the voltage remains constant, but its amplitude is zero, which is the same as having no DC component. In other words, a sequence that creates a constant zero voltage does not have a DC component.

AMI is commonly used for long-distance communication, but it has a synchronization problem when a long sequence of 0s is present in the data. Later in the chapter, we will see how a scrambling technique can solve this problem.

Multilevel Schemes

The desire to increase the data speed or decrease the required bandwidth has resulted in the creation of many schemes. The goal is to increase the number of bits per baud by encoding a pattern of m data elements into a pattern of n signal elements. We only have two types of data elements (0s and 1s), which means that a group of m data elements can produce a combination of 2^m data patterns. We can have different types of signal elements by allowing different signal levels. If we have L different levels, then we can produce L^n combinations of signal patterns. If $2^m = L^n$, then each data pattern is encoded into one signal pattern. If $2^m < L^n$, data patterns occupy only a subset of signal patterns. The subset can be carefully designed to prevent baseline wandering, to provide synchronization, and to detect errors that occurred during data transmission. Data encoding is not possible if $2^m > L^n$ because some of the data patterns cannot be encoded.

The code designers have classified these types of coding as *mBnL*, where m is the length of the binary pattern, B means binary data, n is the length of the signal pattern, and L is the number of levels in the signaling. A letter is often used in place of L: B (binary) for $L = 2$, T (ternary) for $L = 3$, and Q (quaternary) for $L = 4$. Note that the first two letters define the data pattern, and the second two define the signal pattern.

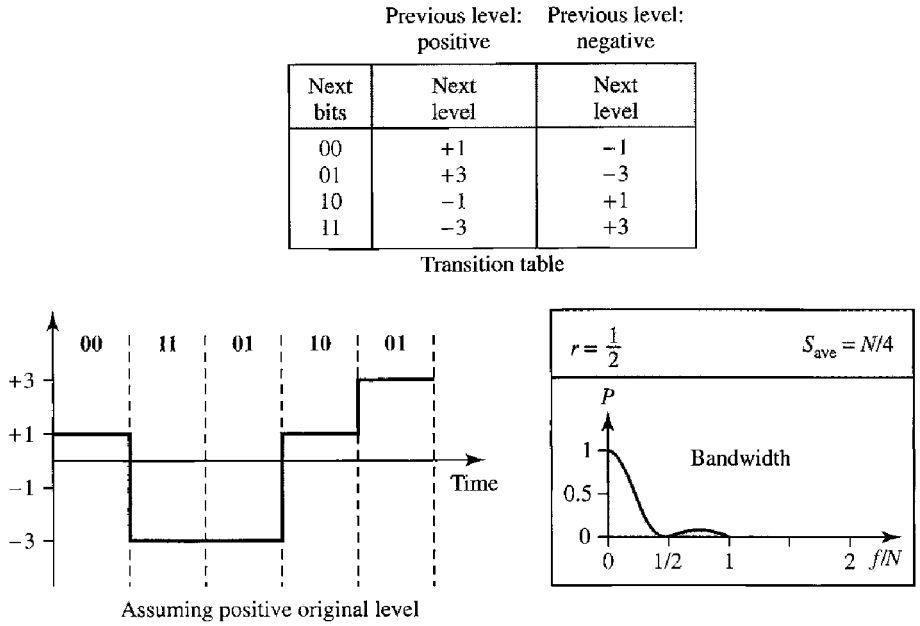
In *mBnL* schemes, a pattern of m data elements is encoded as a pattern of n signal elements in which $2^m \leq L^n$.

2B1Q The first *mBnL* scheme we discuss, **two binary, one quaternary (2B1Q),** uses data patterns of size 2 and encodes the 2-bit patterns as one signal element belonging to a four-level signal. In this type of encoding $m = 2$, $n = 1$, and $L = 4$ (quaternary). Figure 4.10 shows an example of a 2B1Q signal.

The average signal rate of 2B1Q is $S = N/4$. This means that using 2B1Q, we can send data 2 times faster than by using NRZ-L. However, 2B1Q uses four different signal levels, which means the receiver has to discern four different thresholds. The reduced bandwidth comes with a price. There are no redundant signal patterns in this scheme because $2^2 = 4^1$.

As we will see in Chapter 9, 2B1Q is used in DSL (Digital Subscriber Line) technology to provide a high-speed connection to the Internet by using subscriber telephone lines.

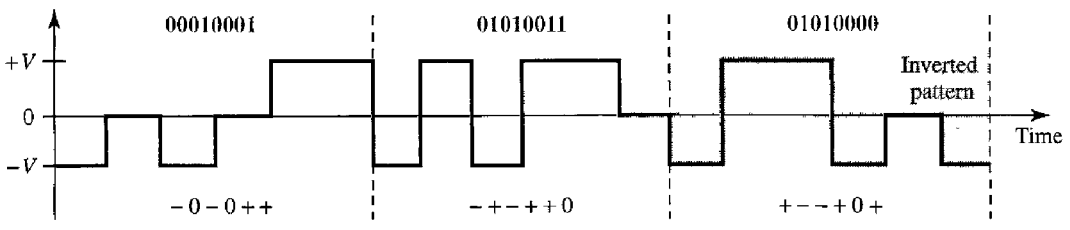
Figure 4.10 *Multilevel: 2B1Q scheme*



8B6T A very interesting scheme is **eight binary, six ternary (8B6T).** This code is used with 100BASE-4T cable, as we will see in Chapter 13. The idea is to encode a pattern of 8 bits as a pattern of 6 signal elements, where the signal has three levels (ternary). In this type of scheme, we can have $2^8 = 256$ different data patterns and $3^6 = 478$ different signal patterns. The mapping table is shown in Appendix D. There are 478 – 256 = 222 redundant signal elements that provide synchronization and error detection. Part of the redundancy is also used to provide DC balance. Each signal pattern has a weight of 0 or +1 DC values. This means that there is no pattern with the weight –1. To make the whole stream DC-balanced, the sender keeps track of the weight. If two groups of weight 1 are encountered one after another, the first one is sent as is, while the next one is totally inverted to give a weight of –1.

Figure 4.11 shows an example of three data patterns encoded as three signal patterns. The three possible signal levels are represented as –, 0, and +. The first 8-bit pattern 00010001 is encoded as the signal pattern –0–0++ with weight 0; the second 8-bit pattern 01010011 is encoded as – + – + + 0 with weight +1. The third bit pattern should be encoded as + – – + 0 + with weight +1. To create DC balance, the sender inverts the actual signal. The receiver can easily recognize that this is an inverted pattern because the weight is –1. The pattern is inverted before decoding.

Figure 4.11 *Multilevel: 8B6T scheme*

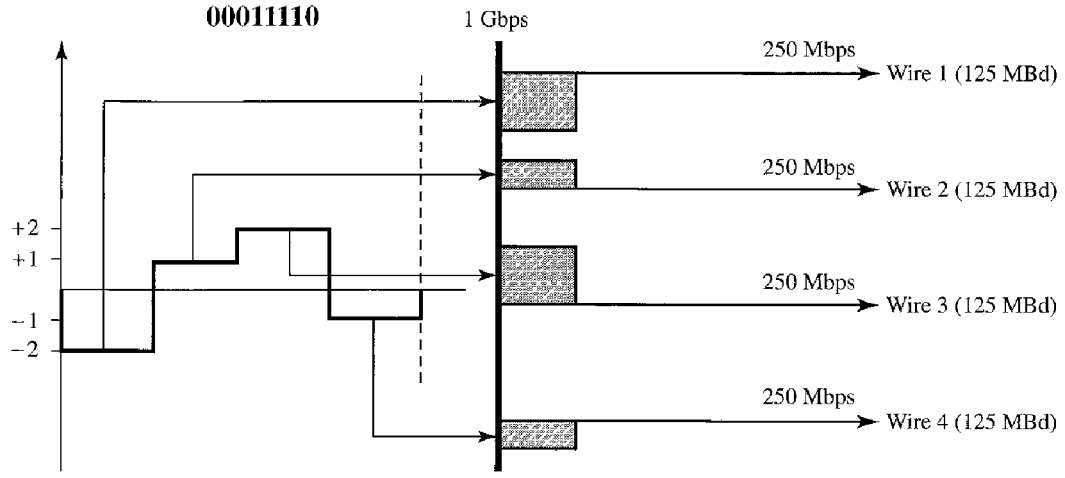


The average signal rate of the scheme is theoretically $S_{ave} = \frac{1}{2} \times N \times \frac{6}{8}$; in practice the minimum bandwidth is very close to $6N/8$.

4D-PAM5 The last signaling scheme we discuss in this category is called **four-dimensional five-level pulse amplitude modulation (4D-PAM5).** The 4D means that data is sent over four wires at the same time. It uses five voltage levels, such as –2, –1, 0, 1, and 2. However, one level, level 0, is used only for forward error detection (discussed in Chapter 10). If we assume that the code is just one-dimensional, the four levels create something similar to 8B4Q. In other words, an 8-bit word is translated to a signal element of four different levels. The worst signal rate for this imaginary one-dimensional version is $N \times 4/8$, or $N/2$.

The technique is designed to send data over four channels (four wires). This means the signal rate can be reduced to $N/8$, a significant achievement. All 8 bits can be fed into a wire simultaneously and sent by using one signal element. The point here is that the four signal elements comprising one signal group are sent simultaneously in a four-dimensional setting. Figure 4.12 shows the imaginary one-dimensional and the actual four-dimensional implementation. Gigabit LANs (see Chapter 13) use this technique to send 1-Gbps data over four copper cables that can handle 125 Mbaud. This scheme has a lot of redundancy in the signal pattern because 2^8 data patterns are matched to $4^4 = 256$ signal patterns. The extra signal patterns can be used for other purposes such as error detection.

Figure 4.12 *Multilevel: 4D-PAM5 scheme*



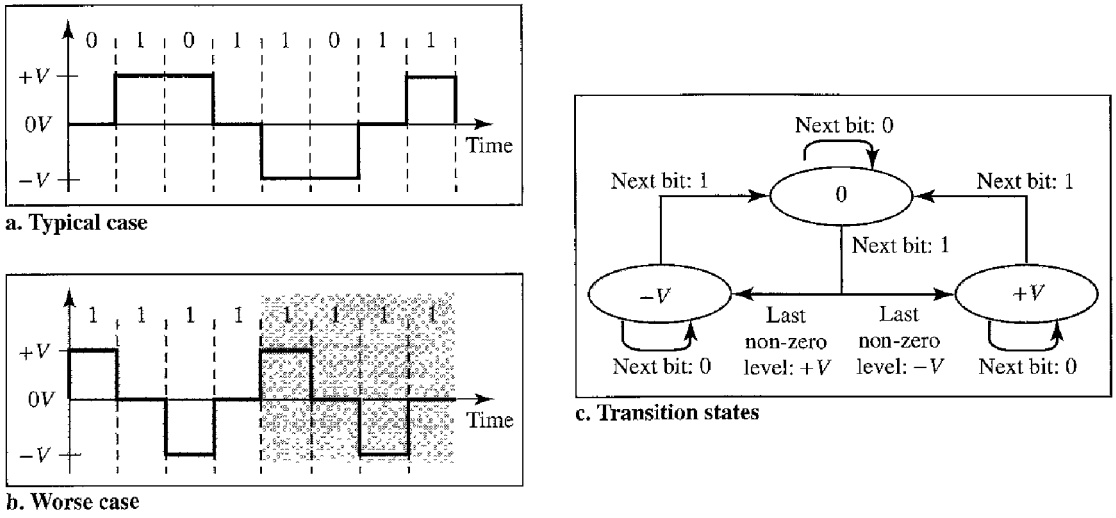
Multiline Transmission: MLT-3

NRZ-I and differential Manchester are classified as differential encoding but use two transition rules to encode binary data (no inversion, inversion). If we have a signal with more than two levels, we can design a differential encoding scheme with more than two transition rules. MLT-3 is one of them. The **multiline transmission, three level (MLT-3)** scheme uses three levels ($+V$, 0, and $-V$) and three transition rules to move between the levels.

1. If the next bit is 0, there is no transition.
2. If the next bit is 1 and the current level is not 0, the next level is 0.
3. If the next bit is 1 and the current level is 0, the next level is the opposite of the last nonzero level.

The behavior of MLT-3 can best be described by the state diagram shown in Figure 4.13. The three voltage levels ($-V$, 0, and $+V$) are shown by three states (ovals). The transition from one state (level) to another is shown by the connecting lines. Figure 4.13 also shows two examples of an MLT-3 signal.

Figure 4.13 *Multitransition: MLT-3 scheme*



One might wonder why we need to use MLT-3, a scheme that maps one bit to one signal element. The signal rate is the same as that for NRZ-I, but with greater complexity (three levels and complex transition rules). It turns out that the shape of the signal in this scheme helps to reduce the required bandwidth. Let us look at the worst-case scenario, a sequence of 1s. In this case, the signal element pattern $+V0-V0$ is repeated every 4 bits. A nonperiodic signal has changed to a periodic signal with the period equal to 4 times the bit duration. This worst-case situation can be simulated as an analog signal with a frequency one-fourth of the bit rate. In other words, the signal rate for MLT-3 is one-fourth the bit rate. This makes MLT-3 a suitable choice when we need to send 100 Mbps on a copper wire that cannot support more than 32 MHz (frequencies above this level create electromagnetic emissions). MLT-3 and LANs are discussed in Chapter 13.

Summary of Line Coding Schemes

We summarize in Table 4.1 the characteristics of the different schemes discussed.

Table 4.1 *Summary of line coding schemes*

Category	*Scheme*	*Bandwidth (average)*	*Characteristics*
Unipolar	NRZ	$B = N/2$	Costly, no self-synchronization if long 0s or 1s, DC
Unipolar	NRZ-L	$B = N/2$	No self-synchronization if long 0s or 1s, DC
	NRZ-I	$B = N/2$	No self-synchronization for long 0s, DC
	Biphase	$B = N$	Self-synchronization, no DC, high bandwidth

Table 4.1 *Summary of line coding schemes (continued)*

Category	*Scheme*	*Bandwidth (average)*	*Characteristics*
Bipolar	AMI	$B = N/2$	No self-synchronization for long 0s, DC
Multilevel	2B1Q	$B = N/4$	No self-synchronization for long same double bits
	8B6T	$B = 3N/4$	Self-synchronization, no DC
	4D-PAM5	$B = N/8$	Self-synchronization, no DC
Multiline	MLT-3	$B = N/3$	No self-synchronization for long 0s

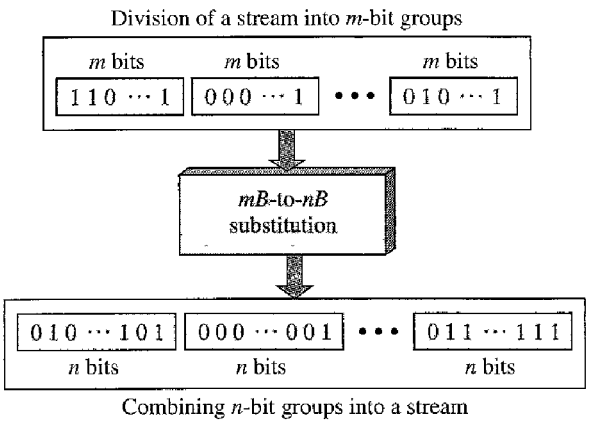
Block Coding

We need redundancy to ensure synchronization and to provide some kind of inherent error detecting. Block coding can give us this redundancy and improve the performance of line coding. In general, **block coding** changes a block of m bits into a block of n bits, where n is larger than m. Block coding is referred to as an *mB/nB* encoding technique.

Block coding is normally referred to as *mB/nB* coding; it replaces each *m*-bit group with an *n*-bit group.

The slash in block encoding (for example, 4B/5B) distinguishes block encoding from multilevel encoding (for example, 8B6T), which is written without a slash. Block coding normally involves three steps: division, substitution, and combination. In the division step, a sequence of bits is divided into groups of m bits. For example, in 4B/5B encoding, the original bit sequence is divided into 4-bit groups. The heart of block coding is the substitution step. In this step, we substitute an m-bit group for an n-bit group. For example, in 4B/5B encoding we substitute a 4-bit code for a 5-bit group. Finally, the n-bit groups are combined together to form a stream. The new stream has more bits than the original bits. Figure 4.14 shows the procedure.

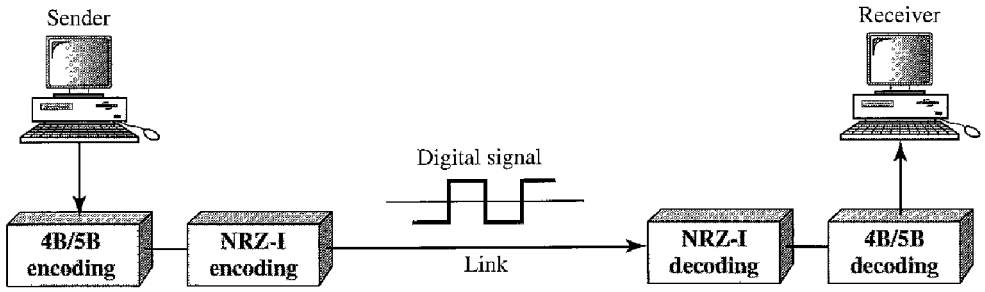
Figure 4.14 *Block coding concept*



4B/5B

The **four binary/five binary (4B/5B)** coding scheme was designed to be used in combination with NRZ-I. Recall that NRZ-I has a good signal rate, one-half that of the biphase, but it has a synchronization problem. A long sequence of 0s can make the receiver clock lose synchronization. One solution is to change the bit stream, prior to encoding with NRZ-I, so that it does not have a long stream of 0s. The 4B/5B scheme achieves this goal. The block-coded stream does not have more that three consecutive 0s, as we will see later. At the receiver, the NRZ-I encoded digital signal is first decoded into a stream of bits and then decoded to remove the redundancy. Figure 4.15 shows the idea.

Figure 4.15 *Using block coding 4B/5B with NRZ-I line coding scheme*



In 4B/5B, the 5-bit output that replaces the 4-bit input has no more than one leading zero (left bit) and no more than two trailing zeros (right bits). So when different groups are combined to make a new sequence, there are never more than three consecutive 0s. (Note that NRZ-I has no problem with sequences of 1s.) Table 4.2 shows the corresponding pairs used in 4B/5B encoding. Note that the first two columns pair a 4-bit group with a 5-bit group. A group of 4 bits can have only 16 different combinations while a group of 5 bits can have 32 different combinations. This means that there are 16 groups that are not used for 4B/5B encoding. Some of these unused groups are used for control purposes; the others are not used at all. The latter provide a kind of error detection. If a 5-bit group arrives that belongs to the unused portion of the table, the receiver knows that there is an error in the transmission.

Table 4.2 *4B/5B mapping codes*

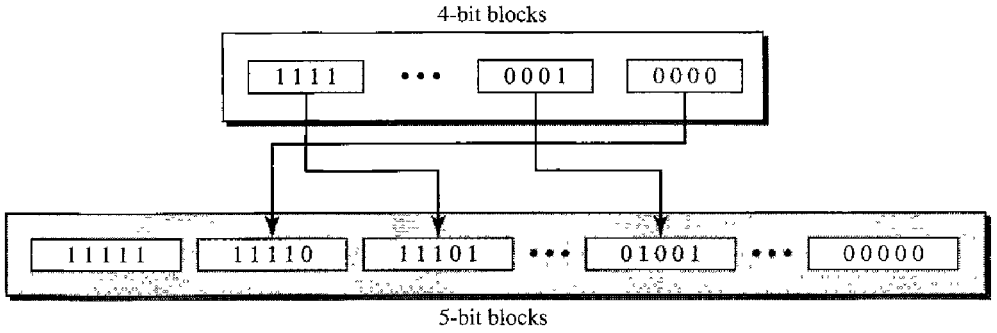
Data Sequence	*Encoded Sequence*	*Control Sequence*	*Encoded Sequence*
0000	11110	Q (Quiet)	00000
0001	01001	I (Idle)	11111
0010	10100	H (Halt)	00100
0011	10101	J (Start delimiter)	11000
0100	01010	K (Start delimiter)	10001
0101	01011	T (End delimiter)	01101

Table 4.2 *4B/5B mapping codes (continued)*

Data Sequence	*Encoded Sequence*	*Control Sequence*	*Encoded Sequence*
0110	01110	S (Set)	11001
0111	01111	R (Reset)	00111
1000	10010		
1001	10011		
1010	10110		
1011	10111		
1100	11010		
1101	11011		
1110	11100		
1111	11101		

Figure 4.16 shows an example of substitution in 4B/5B coding. 4B/5B encoding solves the problem of synchronization and overcomes one of the deficiencies of NRZ-I. However, we need to remember that it increases the signal rate of NRZ-I. The redundant bits add 20 percent more baud. Still, the result is less than the biphase scheme which has a signal rate of 2 times that of NRZ-I. However, 4B/5B block encoding does not solve the DC component problem of NRZ-I. If a DC component is unacceptable, we need to use biphase or bipolar encoding.

Figure 4.16 *Substitution in 4B/5B block coding*



Example 4.5

We need to send data at a 1-Mbps rate. What is the minimum required bandwidth, using a combination of 4B/5B and NRZ-I or Manchester coding?

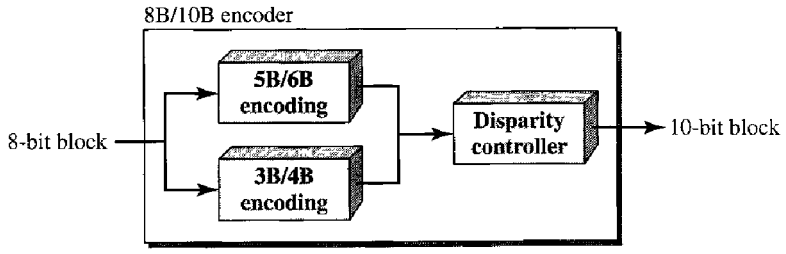
Solution

First 4B/5B block coding increases the bit rate to 1.25 Mbps. The minimum bandwidth using NRZ-I is $N/2$ or 625 kHz. The Manchester scheme needs a minimum bandwidth of 1 MHz. The first choice needs a lower bandwidth, but has a DC component problem; the second choice needs a higher bandwidth, but does not have a DC component problem.

8B/10B

The **eight binary/ten binary (8B/10B)** encoding is similar to 4B/5B encoding except that a group of 8 bits of data is now substituted by a 10-bit code. It provides greater error detection capability than 4B/5B. The 8B/10B block coding is actually a combination of 5B/6B and 3B/4B encoding, as shown in Figure 4.17.

Figure 4.17 *8B/10B block encoding*



The most five significant bits of a 10-bit block is fed into the 5B/6B encoder; the least 3 significant bits is fed into a 3B/4B encoder. The split is done to simplify the mapping table. To prevent a long run of consecutive 0s or 1s, the code uses a disparity controller which keeps track of excess 0s over 1s (or 1s over 0s). If the bits in the current block create a disparity that contributes to the previous disparity (either direction), then each bit in the code is complemented (a 0 is changed to a 1 and a 1 is changed to a 0). The coding has $2^{10} - 2^8 = 768$ redundant groups that can be used for disparity checking and error detection. In general, the technique is superior to 4B/5B because of better built-in error-checking capability and better synchronization.

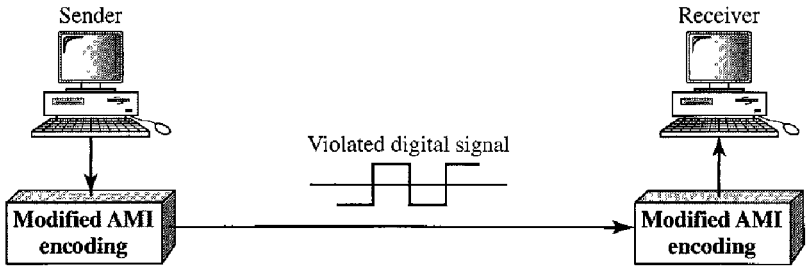
Scrambling

Biphase schemes that are suitable for dedicated links between stations in a LAN are not suitable for long-distance communication because of their wide bandwidth requirement. The combination of block coding and NRZ line coding is not suitable for long-distance encoding either, because of the DC component. Bipolar AMI encoding, on the other hand, has a narrow bandwidth and does not create a DC component. However, a long sequence of 0s upsets the synchronization. If we can find a way to avoid a long sequence of 0s in the original stream, we can use bipolar AMI for long distances. We are looking for a technique that does not increase the number of bits and does provide synchronization. We are looking for a solution that substitutes long zero-level pulses with a combination of other levels to provide synchronization. One solution is called **scrambling.** We modify part of the AMI rule to include scrambling, as shown in Figure 4.18. Note that scrambling, as opposed to block coding, is done at the same time as encoding. The system needs to insert the required pulses based on the defined scrambling rules. Two common scrambling techniques are B8ZS and HDB3.

B8ZS

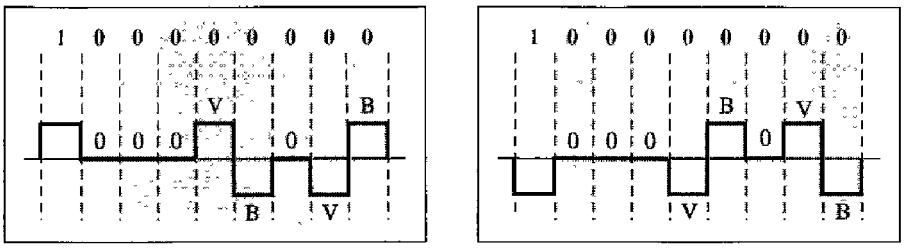
Bipolar with 8-zero substitution (B8ZS) is commonly used in North America. In this technique, eight consecutive zero-level voltages are replaced by the sequence

Figure 4.18 *AMI used with scrambling*



000VB0VB. The V in the sequence denotes *violation;* this is a nonzero voltage that breaks an AMI rule of encoding (opposite polarity from the previous). The B in the sequence denotes *bipolar,* which means a nonzero level voltage in accordance with the AMI rule. There are two cases, as shown in Figure 4.19.

Figure 4.19 *Two cases of B8ZS scrambling technique*



a. Previous level is positive.

b. Previous level is negative.

Note that the scrambling in this case does not change the bit rate. Also, the technique balances the positive and negative voltage levels (two positives and two negatives), which means that the DC balance is maintained. Note that the substitution may change the polarity of a 1 because, after the substitution, AMI needs to follow its rules.

B8ZS substitutes eight consecutive zeros with 000VB0VB.

One more point is worth mentioning. The letter V (violation) or B (bipolar) here is relative. The V means the same polarity as the polarity of the previous nonzero pulse; B means the polarity opposite to the polarity of the previous nonzero pulse.

HDB3

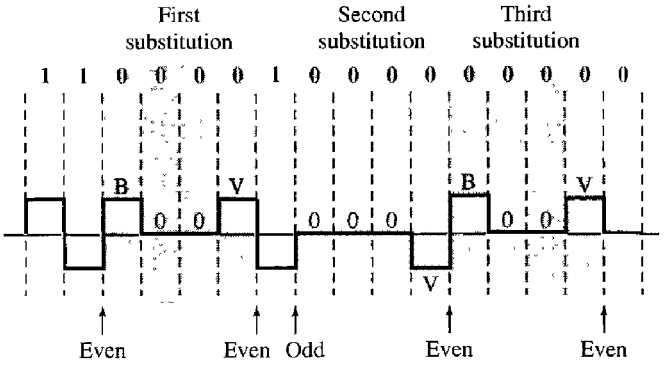
High-density bipolar 3-zero (HDB3) is commonly used outside of North America. In this technique, which is more conservative than B8ZS, four consecutive zero-level voltages are replaced with a sequence of **000V** or **B00V.** The reason for two different substitutions is to

maintain the even number of nonzero pulses after each substitution. The two rules can be stated as follows:

1. If the number of nonzero pulses after the last substitution is odd, the substitution pattern will be **000V**, which makes the total number of nonzero pulses even.
2. If the number of nonzero pulses after the last substitution is even, the substitution pattern will be **B00V**, which makes the total number of nonzero pulses even.

Figure 4.20 shows an example.

Figure 4.20 *Different situations in HDB3 scrambling technique*



There are several points we need to mention here. First, before the first substitution, the number of nonzero pulses is even, so the first substitution is B00V. After this substitution, the polarity of the 1 bit is changed because the AMI scheme, after each substitution, must follow its own rule. After this bit, we need another substitution, which is 000V because we have only one nonzero pulse (odd) after the last substitution. The third substitution is B00V because there are no nonzero pulses after the second substitution (even).

HDB3 substitutes four consecutive zeros with 000V or B00V depending on the number of nonzero pulses after the last substitution.

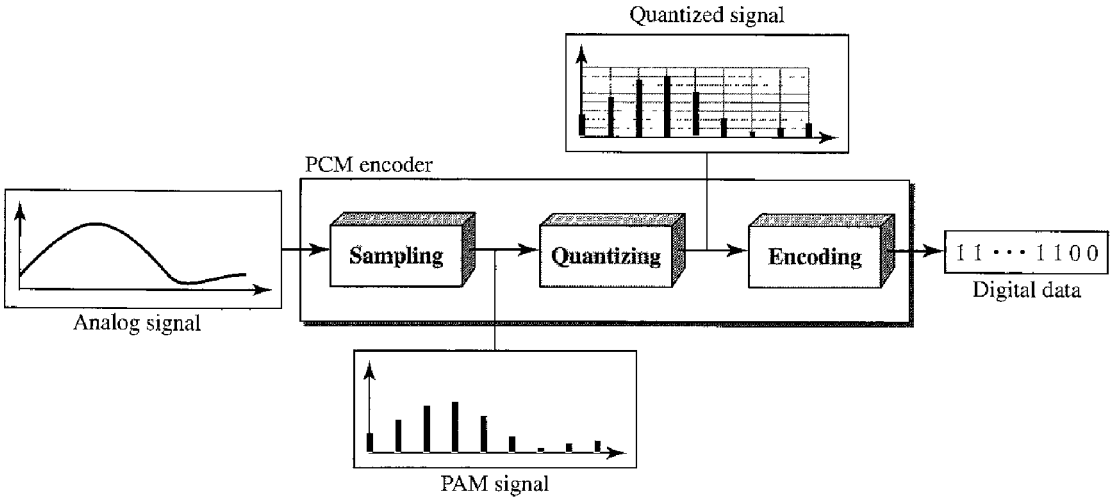
4.2 ANALOG-TO-DIGITAL CONVERSION

The techniques described in Section 4.1 convert digital data to digital signals. Sometimes, however, we have an analog signal such as one created by a microphone or camera. We have seen in Chapter 3 that a digital signal is superior to an analog signal. The tendency today is to change an analog signal to digital data. In this section we describe two techniques, pulse code modulation and delta modulation. After the digital data are created (digitization), we can use one of the techniques described in Section 4.1 to convert the digital data to a digital signal.

Pulse Code Modulation (PCM)

The most common technique to change an analog signal to digital data (**digitization**) is called **pulse code modulation (PCM).** A PCM encoder has three processes, as shown in Figure 4.21.

Figure 4.21 *Components of PCM encoder*



1. The analog signal is sampled.
2. The sampled signal is quantized.
3. The quantized values are encoded as streams of bits.

Sampling

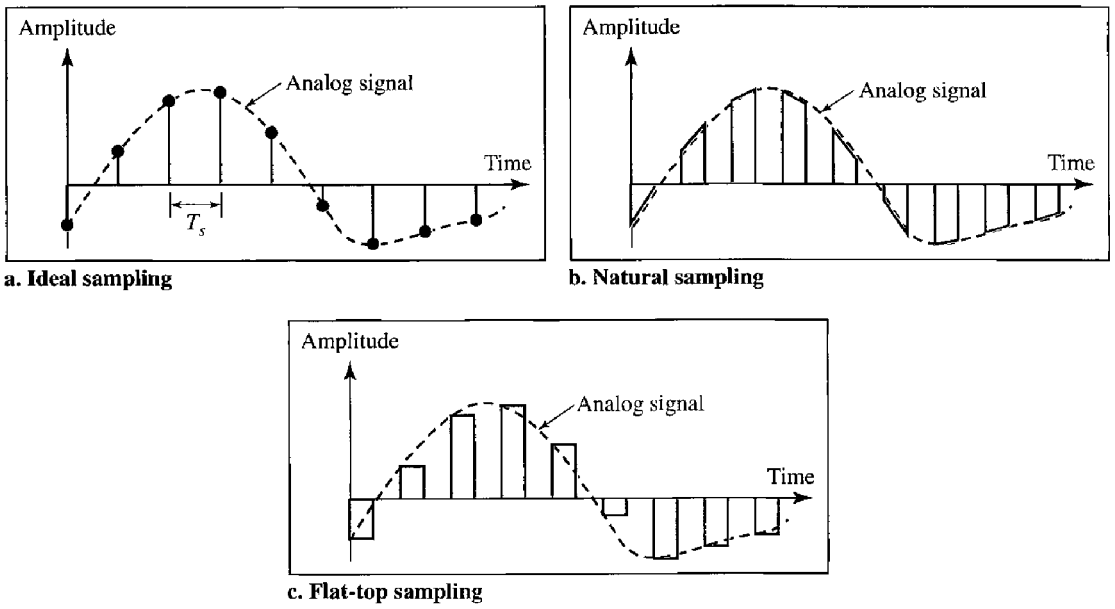
The first step in PCM is **sampling.** The analog signal is sampled every T_s s, where T_s is the sample interval or period. The inverse of the sampling interval is called the **sampling rate** or **sampling frequency** and denoted by f_s, where $f_s = 1/T_s$. There are three sampling methods—ideal, natural, and flat-top—as shown in Figure 4.22.

In ideal sampling, pulses from the analog signal are sampled. This is an ideal sampling method and cannot be easily implemented. In natural sampling, a high-speed switch is turned on for only the small period of time when the sampling occurs. The result is a sequence of samples that retains the shape of the analog signal. The most common sampling method, called **sample and hold,** however, creates flat-top samples by using a circuit.

The sampling process is sometimes referred to as **pulse amplitude modulation (PAM).** We need to remember, however, that the result is still an analog signal with nonintegral values.

Sampling Rate One important consideration is the sampling rate or frequency. What are the restrictions on T_s? This question was elegantly answered by Nyquist. According to the **Nyquist theorem,** to reproduce the original analog signal, one necessary condition is that the **sampling rate** be at least twice the highest frequency in the original signal.

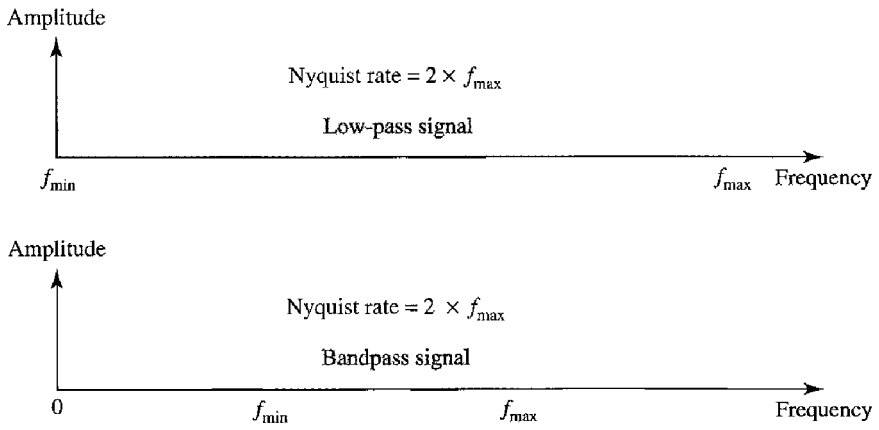
Figure 4.22 *Three different sampling methods for PCM*



According to the Nyquist theorem, the sampling rate must be at least 2 times the highest frequency contained in the signal.

We need to elaborate on the theorem at this point. First, we can sample a signal only if the signal is band-limited. In other words, a signal with an infinite bandwidth cannot be sampled. Second, the sampling rate must be at least 2 times the highest frequency, not the bandwidth. If the analog signal is low-pass, the bandwidth and the highest frequency are the same value. If the analog signal is bandpass, the bandwidth value is lower than the value of the maximum frequency. Figure 4.23 shows the value of the sampling rate for two types of signals.

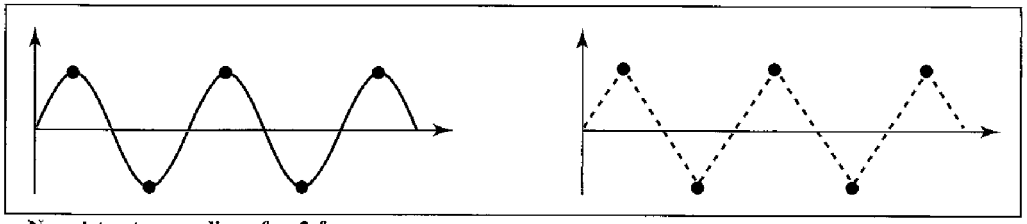
Figure 4.23 *Nyquist sampling rate for low-pass and bandpass signals*



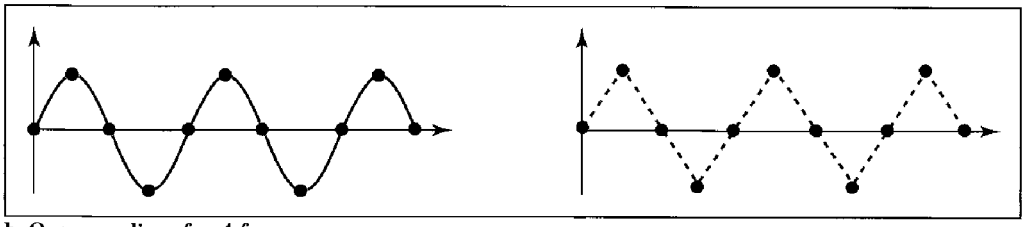
Example 4.6

For an intuitive example of the Nyquist theorem, let us sample a simple sine wave at three sampling rates: $f_s = 4f$ (2 times the Nyquist rate), $f_s = 2f$ (Nyquist rate), and $f_s = f$ (one-half the Nyquist rate). Figure 4.24 shows the sampling and the subsequent recovery of the signal.

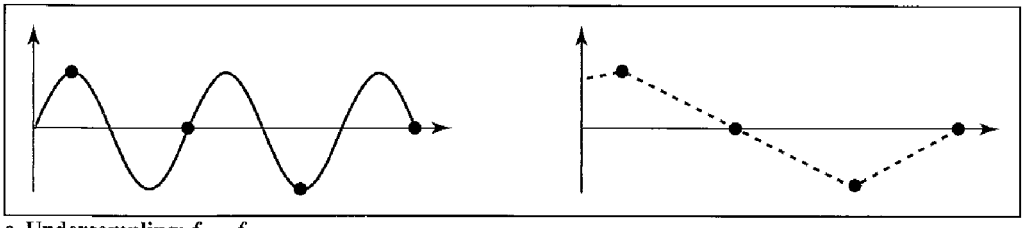
Figure 4.24 *Recovery of a sampled sine wave for different sampling rates*



a. Nyquist rate sampling: $f_s = 2f$



b. Oversampling: $f_s = 4f$



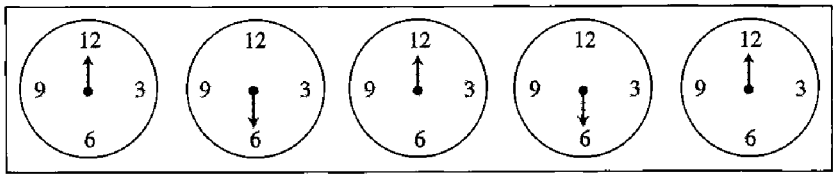
c. Undersampling: $f_s = f$

It can be seen that sampling at the Nyquist rate can create a good approximation of the original sine wave (part a). Oversampling in part b can also create the same approximation, but it is redundant and unnecessary. Sampling below the Nyquist rate (part c) does not produce a signal that looks like the original sine wave.

Example 4.7

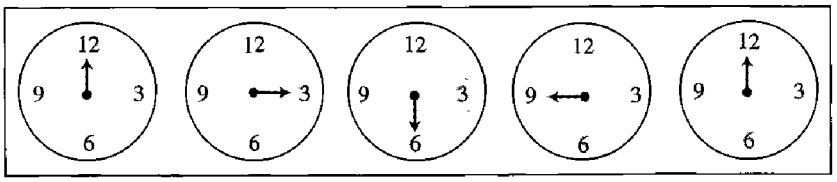
As an interesting example, let us see what happens if we sample a periodic event such as the revolution of a hand of a clock. The second hand of a clock has a period of 60 s. According to the Nyquist theorem, we need to sample the hand (take and send a picture) every 30 s ($T_s = \frac{1}{2} T$ or $f_s = 2f$). In Figure 4.25a, the sample points, in order, are 12, 6, 12, 6, 12, and 6. The receiver of the samples cannot tell if the clock is moving forward or backward. In part b, we sample at double the Nyquist rate (every 15 s). The sample points, in order, are 12, 3, 6, 9, and 12. The clock is moving forward. In part c, we sample below the Nyquist rate ($T_s = \frac{3}{4} T$ or $f_s = \frac{4}{3} f$). The sample points, in order, are 12, 9, 6, 3, and 12. Although the clock is moving forward, the receiver thinks that the clock is moving backward.

Figure 4.25 *Sampling of a clock with only one hand*



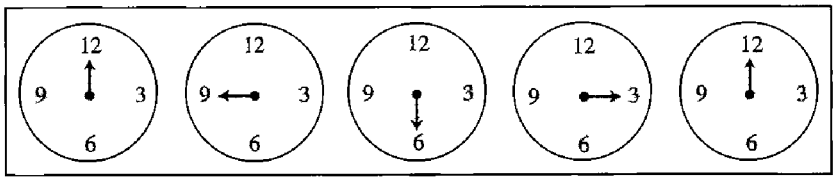
Samples can mean that the clock is moving either forward or backward. (12-6-12-6-12)

a. Sampling at Nyquist rate: $T_s = \frac{1}{2}T$



Samples show clock is moving forward. (12-3-6-9-12)

b. Oversampling (above Nyquist rate): $T_s = \frac{1}{4}T$



Samples show clock is moving backward. (12-9-6-3-12)

c. Undersampling (below Nyquist rate): $T_s = \frac{3}{4}T$

Example 4.8

An example related to Example 4.7 is the seemingly backward rotation of the wheels of a forward-moving car in a movie. This can be explained by undersampling. A movie is filmed at 24 frames per second. If a wheel is rotating more than 12 times per second, the undersampling creates the impression of a backward rotation.

Example 4.9

Telephone companies digitize voice by assuming a maximum frequency of 4000 Hz. The sampling rate therefore is 8000 samples per second.

Example 4.10

A complex low-pass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

Solution

The bandwidth of a low-pass signal is between 0 and f, where f is the maximum frequency in the signal. Therefore, we can sample this signal at 2 times the highest frequency (200 kHz). The sampling rate is therefore 400,000 samples per second.

Example 4.11

A complex bandpass signal has a bandwidth of 200 kHz. What is the minimum sampling rate for this signal?

Solution

We cannot find the minimum sampling rate in this case because we do not know where the bandwidth starts or ends. We do not know the maximum frequency in the signal.

Quantization

The result of sampling is a series of pulses with amplitude values between the maximum and minimum amplitudes of the signal. The set of amplitudes can be infinite with nonintegral values between the two limits. These values cannot be used in the encoding process. The following are the steps in quantization:

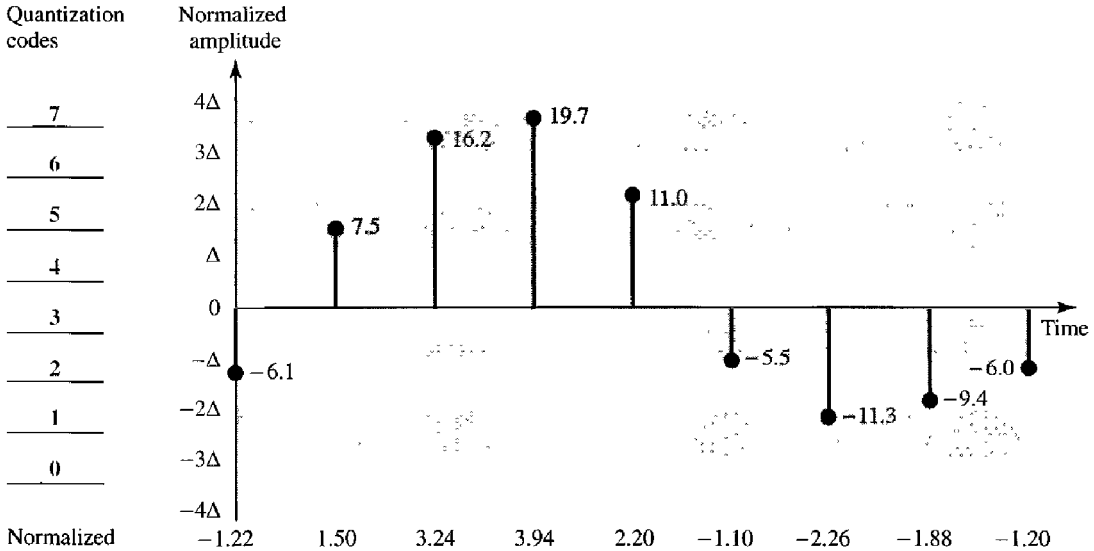
1. We assume that the original analog signal has instantaneous amplitudes between V_{min} and V_{max}.
2. We divide the range into L zones, each of height Δ (delta).

$$\Delta = \frac{V_{max} - V_{min}}{L}$$

3. We assign quantized values of 0 to $L - 1$ to the midpoint of each zone.
4. We approximate the value of the sample amplitude to the quantized values.

As a simple example, assume that we have a sampled signal and the sample amplitudes are between −20 and +20 V. We decide to have eight levels ($L = 8$). This means that $\Delta = 5$ V. Figure 4.26 shows this example.

Figure 4.26 *Quantization and encoding of a sampled signal*



Normalized PAM values	−1.22	1.50	3.24	3.94	2.20	−1.10	−2.26	−1.88	−1.20
Normalized quantized values	−1.50	1.50	3.50	3.50	2.50	−1.50	−2.50	−1.50	−1.50
Normalized error	−0.38	0	+0.26	−0.44	+0.30	−0.40	−0.24	+0.38	−0.30
Quantization code	2	5	7	7	6	2	1	2	2
Encoded words	010	101	111	111	110	010	001	010	010

We have shown only nine samples using ideal sampling (for simplicity). The value at the top of each sample in the graph shows the actual amplitude. In the chart, the first row is the normalized value for each sample (actual amplitude/Δ). The quantization process selects the quantization value from the middle of each zone. This means that the normalized quantized values (second row) are different from the normalized amplitudes. The difference is called the *normalized error* (third row). The fourth row is the quantization code for each sample based on the quantization levels at the left of the graph. The encoded words (fifth row) are the final products of the conversion.

Quantization Levels In the previous example, we showed eight quantization levels. The choice of L, the number of levels, depends on the range of the amplitudes of the analog signal and how accurately we need to recover the signal. If the amplitude of a signal fluctuates between two values only, we need only two levels; if the signal, like voice, has many amplitude values, we need more quantization levels. In audio digitizing, L is normally chosen to be 256; in video it is normally thousands. Choosing lower values of L increases the quantization error if there is a lot of fluctuation in the signal.

Quantization Error One important issue is the error created in the quantization process. (Later, we will see how this affects high-speed modems.) Quantization is an approximation process. The input values to the quantizer are the real values; the output values are the approximated values. The output values are chosen to be the middle value in the zone. If the input value is also at the middle of the zone, there is no quantization error; otherwise, there is an error. In the previous example, the normalized amplitude of the third sample is 3.24, but the normalized quantized value is 3.50. This means that there is an error of +0.26. The value of the error for any sample is less than Δ/2. In other words, we have $-\Delta/2 \leq \text{error} \leq \Delta/2$.

The quantization error changes the signal-to-noise ratio of the signal, which in turn reduces the upper limit capacity according to Shannon.

It can be proven that the contribution of the **quantization error** to the SNR_{dB} of the signal depends on the number of quantization levels L, or the bits per sample n_b, as shown in the following formula:

$$\textbf{SNR}_{\textbf{dB}} = \textbf{6.02}n_b + \textbf{1.76} \quad \textbf{dB}$$

Example 4.12

What is the SNR_{dB} in the example of Figure 4.26?

Solution

We can use the formula to find the quantization. We have eight levels and 3 bits per sample, so $\text{SNR}_{\text{dB}} = 6.02(3) + 1.76 = 19.82$ dB. Increasing the number of levels increases the SNR.

Example 4.13

A telephone subscriber line must have an SNR_{dB} above 40. What is the minimum number of bits per sample?

Solution

We can calculate the number of bits as

$$\text{SNR}_{\text{dB}} = 6.02n_b + 1.76 = 40 \longrightarrow n = 6.35$$

Telephone companies usually assign 7 or 8 bits per sample.

Uniform Versus Nonuniform Quantization For many applications, the distribution of the instantaneous amplitudes in the analog signal is not uniform. Changes in amplitude often occur more frequently in the lower amplitudes than in the higher ones. For these types of applications it is better to use nonuniform zones. In other words, the height of Δ is not fixed; it is greater near the lower amplitudes and less near the higher amplitudes. Nonuniform quantization can also be achieved by using a process called **companding** and **expanding.** The signal is companded at the sender before conversion; it is expanded at the receiver after conversion. Companding means reducing the instantaneous voltage amplitude for large values; expanding is the opposite process. Companding gives greater weight to strong signals and less weight to weak ones. It has been proved that nonuniform quantization effectively reduces the SNR_{dB} of quantization.

Encoding

The last step in PCM is encoding. After each sample is quantized and the number of bits per sample is decided, each sample can be changed to an n_b-bit code word. In Figure 4.26 the encoded words are shown in the last row. A quantization code of 2 is encoded as 010; 5 is encoded as 101; and so on. Note that the number of bits for each sample is determined from the number of quantization levels. If the number of quantization levels is L, the number of bits is $n_b = \log_2 L$. In our example L is 8 and n_b is therefore 3. The bit rate can be found from the formula

$$\textbf{Bit rate} = \textbf{sampling rate} \times \textbf{number of bits per sample} = f_s \times n_b$$

Example 4.14

We want to digitize the human voice. What is the bit rate, assuming 8 bits per sample?

Solution

The human voice normally contains frequencies from 0 to 4000 Hz. So the sampling rate and bit rate are calculated as follows:

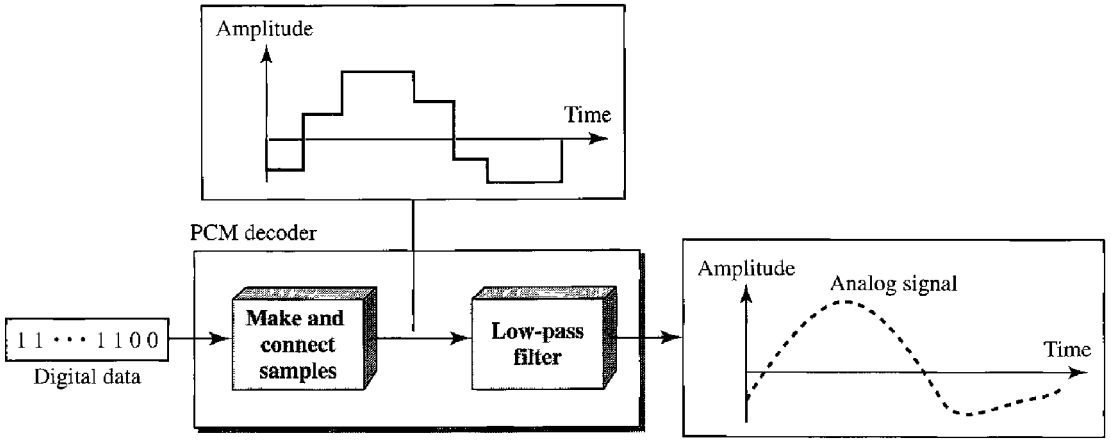
$$\text{Sampling rate} = 4000 \times 2 = 8000 \text{ samples/s}$$
$$\text{Bit rate} = 8000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$$

Original Signal Recovery

The recovery of the original signal requires the PCM decoder. The decoder first uses circuitry to convert the code words into a pulse that holds the amplitude until the next pulse. After the staircase signal is completed, it is passed through a low-pass filter to

smooth the staircase signal into an analog signal. The filter has the same cutoff frequency as the original signal at the sender. If the signal has been sampled at (or greater than) the Nyquist sampling rate and if there are enough quantization levels, the original signal will be recreated. Note that the maximum and minimum values of the original signal can be achieved by using amplification. Figure 4.27 shows the simplified process.

Figure 4.27 *Components of a PCM decoder*



PCM Bandwidth

Suppose we are given the bandwidth of a low-pass analog signal. If we then digitize the signal, what is the new minimum bandwidth of the channel that can pass this digitized signal? We have said that the minimum bandwidth of a line-encoded signal is $B_{min} = c \times N \times (1/r)$. We substitute the value of N in this formula:

$$B_{min} = c \times N \times \frac{1}{r} = c \times n_b \times f_s \times \frac{1}{r} = c \times n_b \times 2 \times B_{analog} \times \frac{1}{r}$$

When $1/r = 1$ (for a NRZ or bipolar signal) and $c = (1/2)$ (the average situation), the minimum bandwidth is

$$B_{min} = n_b \times B_{analog}$$

This means the minimum bandwidth of the digital signal is n_b times greater than the bandwidth of the analog signal. This is the price we pay for digitization.

Example 4.15

We have a low-pass analog signal of 4 kHz. If we send the analog signal, we need a channel with a minimum bandwidth of 4 kHz. If we digitize the signal and send 8 bits per sample, we need a channel with a minimum bandwidth of 8×4 kHz = 32 kHz.

Maximum Data Rate of a Channel

In Chapter 3, we discussed the Nyquist theorem which gives the data rate of a channel as $N_{max} = 2 \times B \times \log_2 L$. We can deduce this rate from the Nyquist sampling theorem by using the following arguments.

1. We assume that the available channel is low-pass with bandwidth B.
2. We assume that the digital signal we want to send has L levels, where each level is a signal element. This means $r = 1/\log_2 L$.
3. We first pass the digital signal through a low-pass filter to cut off the frequencies above B Hz.
4. We treat the resulting signal as an analog signal and sample it at $2 \times B$ samples per second and quantize it using L levels. Additional quantization levels are useless because the signal originally had L levels.
5. The resulting bit rate is $N = f_s \times n_b = 2 \times B \times \log_2 L$. This is the maximum bandwidth; if the case factor c increases, the data rate is reduced.

$$N_{max} = 2 \times B \times \log_2 L \quad \text{bps}$$

Minimum Required Bandwidth

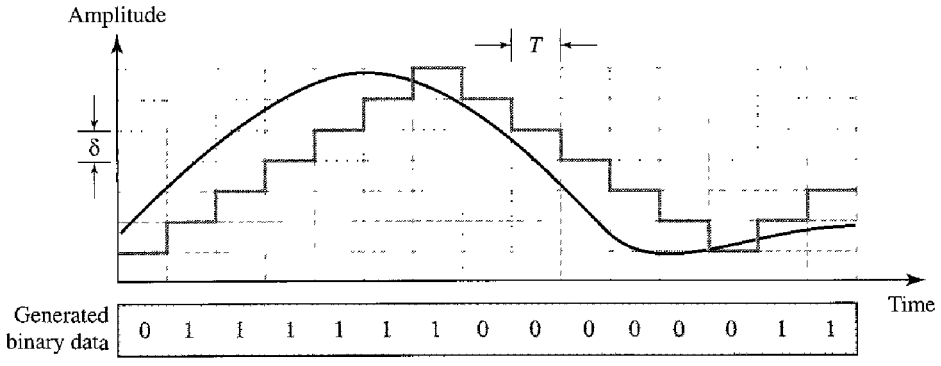
The previous argument can give us the minimum bandwidth if the data rate and the number of signal levels are fixed. We can say

$$B_{min} = \frac{N}{2 \times \log_2 L} \quad \text{Hz}$$

Delta Modulation (DM)

PCM is a very complex technique. Other techniques have been developed to reduce the complexity of PCM. The simplest is *delta modulation*. PCM finds the value of the signal amplitude for each sample; DM finds the change from the previous sample. Figure 4.28 shows the process. Note that there are no code words here; bits are sent one after another.

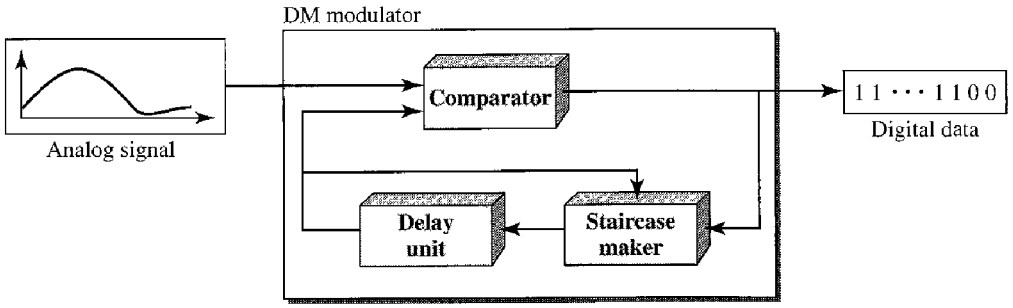
Figure 4.28 *The process of delta modulation*



Modulator

The modulator is used at the sender site to create a stream of bits from an analog signal. The process records the small positive or negative changes, called delta δ. If the delta is positive, the process records a 1; if it is negative, the process records a 0. However, the process needs a base against which the analog signal is compared. The modulator builds a second signal that resembles a staircase. Finding the change is then reduced to comparing the input signal with the gradually made staircase signal. Figure 4.29 shows a diagram of the process.

Figure 4.29 *Delta modulation components*

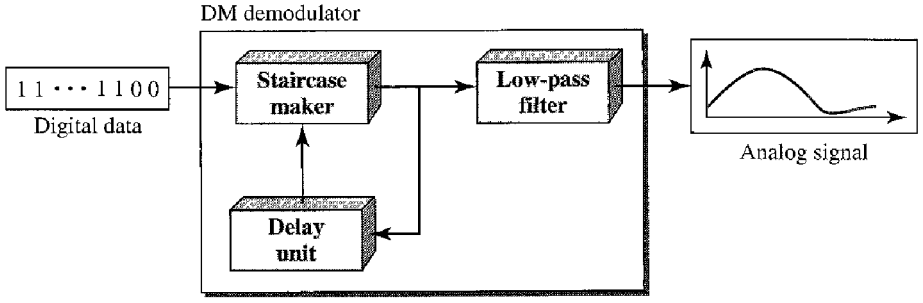


The modulator, at each sampling interval, compares the value of the analog signal with the last value of the staircase signal. If the amplitude of the analog signal is larger, the next bit in the digital data is 1; otherwise, it is 0. The output of the comparator, however, also makes the staircase itself. If the next bit is 1, the staircase maker moves the last point of the staircase signal δ up; it the next bit is 0, it moves it δ down. Note that we need a delay unit to hold the staircase function for a period between two comparisons.

Demodulator

The demodulator takes the digital data and, using the staircase maker and the delay unit, creates the analog signal. The created analog signal, however, needs to pass through a low-pass filter for smoothing. Figure 4.30 shows the schematic diagram.

Figure 4.30 *Delta demodulation components*



Adaptive DM

A better performance can be achieved if the value of δ is not fixed. In **adaptive delta modulation,** the value of δ changes according to the amplitude of the analog signal.

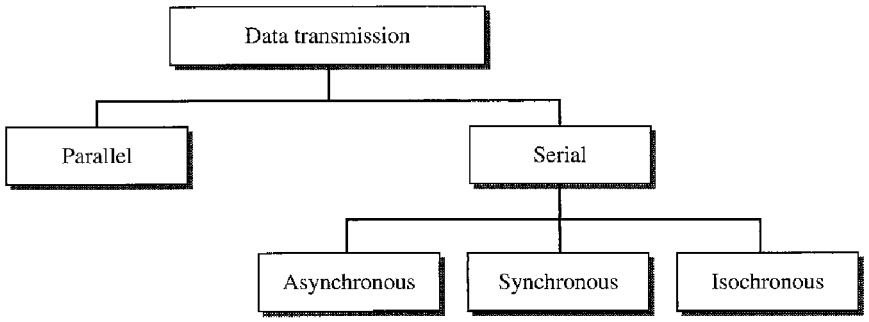
Quantization Error

It is obvious that DM is not perfect. Quantization error is always introduced in the process. The quantization error of DM, however, is much less than that for PCM.

4.3 TRANSMISSION MODES

Of primary concern when we are considering the transmission of data from one device to another is the wiring, and of primary concern when we are considering the wiring is the data stream. Do we send 1 bit at a time; or do we group bits into larger groups and, if so, how? The transmission of binary data across a link can be accomplished in either parallel or serial mode. In parallel mode, multiple bits are sent with each clock tick. In serial mode, 1 bit is sent with each clock tick. While there is only one way to send parallel data, there are three subclasses of serial transmission: asynchronous, synchronous, and isochronous (see Figure 4.31).

Figure 4.31 *Data transmission and modes*



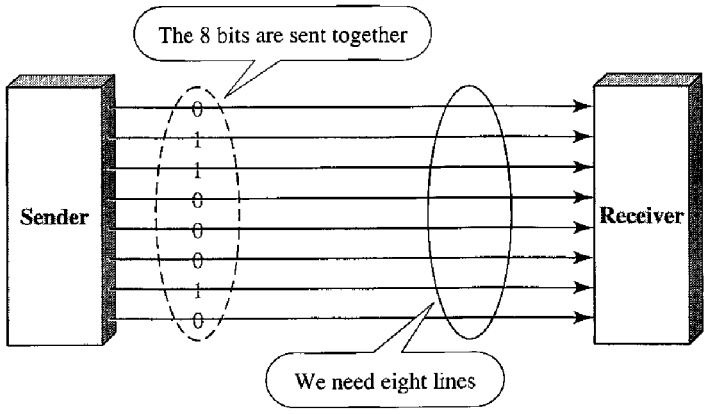
Parallel Transmission

Binary data, consisting of 1s and 0s, may be organized into groups of n bits each. Computers produce and consume data in groups of bits much as we conceive of and use spoken language in the form of words rather than letters. By grouping, we can send data n bits at a time instead of 1. This is called **parallel transmission.**

The mechanism for parallel transmission is a conceptually simple one: Use n wires to send n bits at one time. That way each bit has its own wire, and all n bits of one group can be transmitted with each clock tick from one device to another. Figure 4.32 shows how parallel transmission works for $n = 8$. Typically, the eight wires are bundled in a cable with a connector at each end.

The advantage of parallel transmission is speed. All else being equal, parallel transmission can increase the transfer speed by a factor of n over serial transmission.

Figure 4.32 *Parallel transmission*

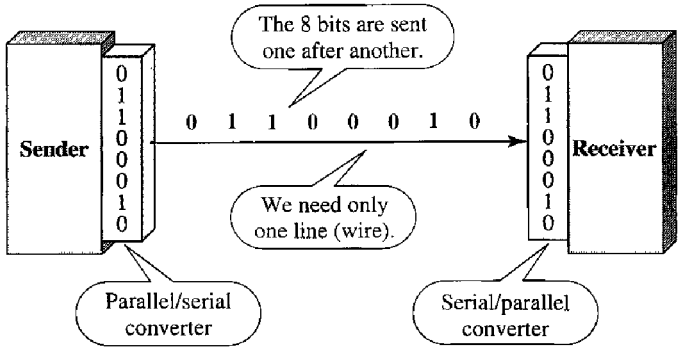


But there is a significant disadvantage: cost. Parallel transmission requires n communication lines (wires in the example) just to transmit the data stream. Because this is expensive, parallel transmission is usually limited to short distances.

Serial Transmission

In **serial transmission** one bit follows another, so we need only one communication channel rather than n to transmit data between two communicating devices (see Figure 4.33).

Figure 4.33 *Serial transmission*



The advantage of serial over parallel transmission is that with only one communication channel, serial transmission reduces the cost of transmission over parallel by roughly a factor of n.

Since communication within devices is parallel, conversion devices are required at the interface between the sender and the line (parallel-to-serial) and between the line and the receiver (serial-to-parallel).

Serial transmission occurs in one of three ways: asynchronous, synchronous, and isochronous.

Asynchronous Transmission

Asynchronous transmission is so named because the timing of a signal is unimportant. Instead, information is received and translated by agreed upon patterns. As long as those patterns are followed, the receiving device can retrieve the information without regard to the rhythm in which it is sent. Patterns are based on grouping the bit stream into bytes. Each group, usually 8 bits, is sent along the link as a unit. The sending system handles each group independently, relaying it to the link whenever ready, without regard to a timer.

Without synchronization, the receiver cannot use timing to predict when the next group will arrive. To alert the receiver to the arrival of a new group, therefore, an extra bit is added to the beginning of each byte. This bit, usually a 0, is called the **start bit.** To let the receiver know that the byte is finished, 1 or more additional bits are appended to the end of the byte. These bits, usually 1s, are called **stop bits.** By this method, each byte is increased in size to at least 10 bits, of which 8 bits is information and 2 bits or more are signals to the receiver. In addition, the transmission of each byte may then be followed by a gap of varying duration. This gap can be represented either by an idle channel or by a stream of additional stop bits.

In asynchronous transmission, we send 1 start bit (0) at the beginning and 1 or more stop bits (1s) at the end of each byte. There may be a gap between each byte.

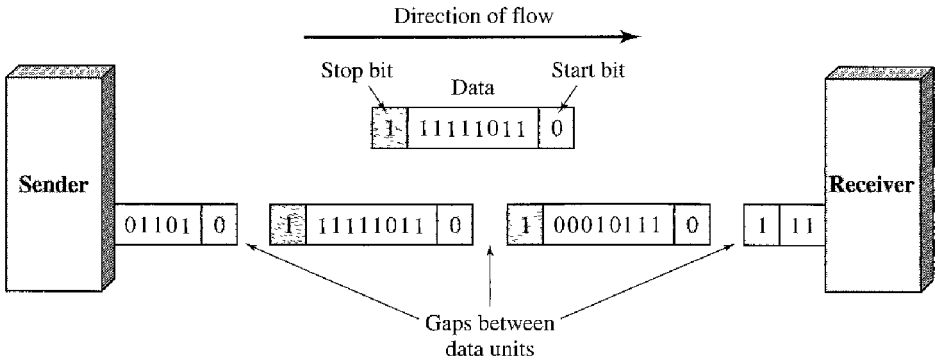
The start and stop bits and the gap alert the receiver to the beginning and end of each byte and allow it to synchronize with the data stream. This mechanism is called *asynchronous* because, at the byte level, the sender and receiver do not have to be synchronized. But within each byte, the receiver must still be synchronized with the incoming bit stream. That is, some synchronization is required, but only for the duration of a single byte. The receiving device resynchronizes at the onset of each new byte. When the receiver detects a start bit, it sets a timer and begins counting bits as they come in. After n bits, the receiver looks for a stop bit. As soon as it detects the stop bit, it waits until it detects the next start bit.

Asynchronous here means "asynchronous at the byte level," but the bits are still synchronized; their durations are the same.

Figure 4.34 is a schematic illustration of asynchronous transmission. In this example, the start bits are 0s, the stop bits are 1s, and the gap is represented by an idle line rather than by additional stop bits.

The addition of stop and start bits and the insertion of gaps into the bit stream make asynchronous transmission slower than forms of transmission that can operate without the addition of control information. But it is cheap and effective, two advantages that make it an attractive choice for situations such as low-speed communication. For example, the connection of a keyboard to a computer is a natural application for asynchronous transmission. A user types only one character at a time, types extremely slowly in data processing terms, and leaves unpredictable gaps of time between each character.

Figure 4.34 *Asynchronous transmission*



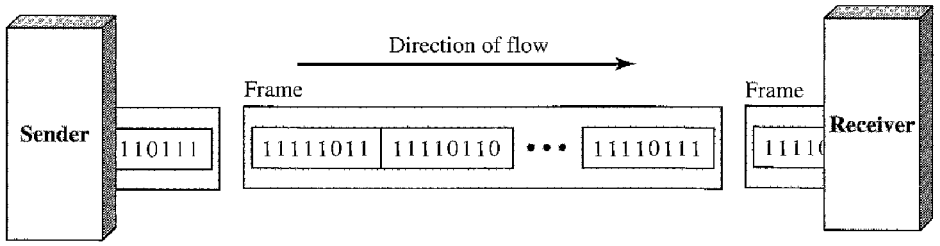
Synchronous Transmission

In **synchronous transmission,** the bit stream is combined into longer "frames," which may contain multiple bytes. Each byte, however, is introduced onto the transmission link without a gap between it and the next one. It is left to the receiver to separate the bit stream into bytes for decoding purposes. In other words, data are transmitted as an unbroken string of 1s and 0s, and the receiver separates that string into the bytes, or characters, it needs to reconstruct the information.

In synchronous transmission, we send bits one after another without start or stop bits or gaps. It is the responsibility of the receiver to group the bits.

Figure 4.35 gives a schematic illustration of synchronous transmission. We have drawn in the divisions between bytes. In reality, those divisions do not exist; the sender puts its data onto the line as one long string. If the sender wishes to send data in separate bursts, the gaps between bursts must be filled with a special sequence of 0s and 1s that means *idle*. The receiver counts the bits as they arrive and groups them in 8-bit units.

Figure 4.35 *Synchronous transmission*



Without gaps and start and stop bits, there is no built-in mechanism to help the receiving device adjust its bit synchronization midstream. Timing becomes very important, therefore, because the accuracy of the received information is completely dependent on the ability of the receiving device to keep an accurate count of the bits as they come in.

The advantage of synchronous transmission is speed. With no extra bits or gaps to introduce at the sending end and remove at the receiving end, and, by extension, with fewer bits to move across the link, synchronous transmission is faster than asynchronous transmission. For this reason, it is more useful for high-speed applications such as the transmission of data from one computer to another. Byte synchronization is accomplished in the data link layer.

We need to emphasize one point here. Although there is no gap between characters in synchronous serial transmission, there may be uneven gaps between frames.

Isochronous

In real-time audio and video, in which uneven delays between frames are not acceptable, synchronous transmission fails. For example, TV images are broadcast at the rate of 30 images per second; they must be viewed at the same rate. If each image is sent by using one or more frames, there should be no delays between frames. For this type of application, synchronization between characters is not enough; the entire stream of bits must be synchronized. The **isochronous transmission** guarantees that the data arrive at a fixed rate.

4.4 RECOMMENDED READING

For more details about subjects discussed in this chapter, we recommend the following books. The items in brackets [. . .] refer to the reference list at the end of the text.

Books

Digital to digital conversion is discussed in Chapter 7 of [Pea92], Chapter 3 of [Cou01], and Section 5.1 of [Sta04]. Sampling is discussed in Chapters 15, 16, 17, and 18 of [Pea92], Chapter 3 of [Cou01], and Section 5.3 of [Sta04]. [Hsu03] gives a good mathematical approach to modulation and sampling. More advanced materials can be found in [Ber96].

4.5 KEY TERMS

adaptive delta modulation
alternate mark inversion (AMI)
analog-to-digital conversion
asynchronous transmission
baseline
baseline wandering
baud rate
biphase
bipolar
bipolar with *8*-zero substitution (B8ZS)
bit rate
block coding
companding and expanding
data element
data rate
DC component
delta modulation (DM)
differential Manchester
digital-to-digital conversion
digitization

eight binary/ten binary (8B/10B)
eight-binary, six-ternary (8B6T)
four binary/five binary (4B/5B)
four dimensional, five-level pulse amplitude modulation (4D-PAM5)
high-density bipolar 3-zero (HDB3)
isochronous transmission
line coding
Manchester
modulation rate
multilevel binary
multiline transmission, 3 level (MLT-3)
nonreturn to zero (NRZ)
nonreturn to zero, invert (NRZ-I)
nonreturn to zero, level (NRZ-L)
Nyquist theorem
parallel transmission
polar
pseudoternary
pulse amplitude modulation (PAM)
pulse code modulation (PCM)
pulse rate
quantization
quantization error
return to zero (RZ)
sampling
sampling rate
scrambling
self-synchronizing
serial transmission
signal element
signal rate
start bit
stop bit
synchronous transmission
transmission mode
two-binary, one quaternary (2B1Q)
unipolar

4.6 SUMMARY

- Digital-to-digital conversion involves three techniques: line coding, block coding, and scrambling.
- Line coding is the process of converting digital data to a digital signal.
- We can roughly divide line coding schemes into five broad categories: unipolar, polar, bipolar, multilevel, and multitransition.
- Block coding provides redundancy to ensure synchronization and inherent error detection. Block coding is normally referred to as *mB/nB* coding; it replaces each *m*-bit group with an *n*-bit group.
- Scrambling provides synchronization without increasing the number of bits. Two common scrambling techniques are B8ZS and HDB3.
- The most common technique to change an analog signal to digital data (digitization) is called pulse code modulation (PCM).
- The first step in PCM is sampling. The analog signal is sampled every T_s s, where T_s is the sample interval or period. The inverse of the sampling interval is called the *sampling rate* or *sampling frequency* and denoted by f_s, where $f_s = 1/T_s$. There are three sampling methods—ideal, natural, and flat-top.
- According to the *Nyquist theorem,* to reproduce the original analog signal, one necessary condition is that the *sampling rate* be at least twice the highest frequency in the original signal.

- Other sampling techniques have been developed to reduce the complexity of PCM. The simplest is *delta modulation*. PCM finds the value of the signal amplitude for each sample; DM finds the change from the previous sample.
- While there is only one way to send parallel data, there are three subclasses of serial transmission: asynchronous, synchronous, and isochronous.
- In asynchronous transmission, we send 1 start bit (0) at the beginning and 1 or more stop bits (1s) at the end of each byte.
- In synchronous transmission, we send bits one after another without start or stop bits or gaps. It is the responsibility of the receiver to group the bits.
- The isochronous mode provides synchronized for the entire stream of bits must. In other words, it guarantees that the data arrive at a fixed rate.

4.7 PRACTICE SET

Review Questions

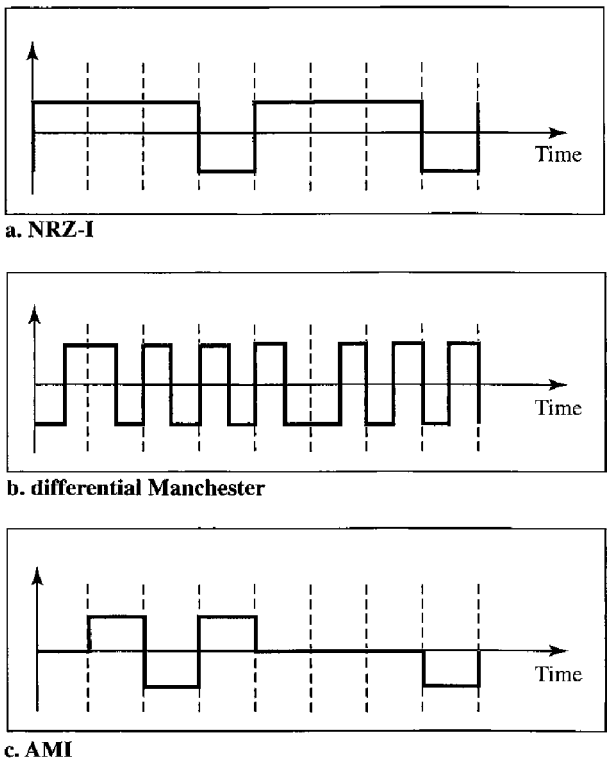
1. List three techniques of digital-to-digital conversion.
2. Distinguish between a signal element and a data element.
3. Distinguish between data rate and signal rate.
4. Define baseline wandering and its effect on digital transmission.
5. Define a DC component and its effect on digital transmission.
6. Define the characteristics of a self-synchronizing signal.
7. List five line coding schemes discussed in this book.
8. Define block coding and give its purpose.
9. Define scrambling and give its purpose.
10. Compare and contrast PCM and DM.
11. What are the differences between parallel and serial transmission?
12. List three different techniques in serial transmission and explain the differences.

Exercises

13. Calculate the value of the signal rate for each case in Figure 4.2 if the data rate is 1 Mbps and c = 1/2.
14. In a digital transmission, the sender clock is 0.2 percent faster than the receiver clock. How many extra bits per second does the sender send if the data rate is 1 Mbps?
15. Draw the graph of the NRZ-L scheme using each of the following data streams, assuming that the last signal level has been positive. From the graphs, guess the bandwidth for this scheme using the average number of changes in the signal level. Compare your guess with the corresponding entry in Table 4.1.
    a. 00000000
    b. 11111111
    c. 01010101
    d. 00110011

16. Repeat Exercise 15 for the NRZ-I scheme.
17. Repeat Exercise 15 for the Manchester scheme.
18. Repeat Exercise 15 for the differential Manchester scheme.
19. Repeat Exercise 15 for the 2B1Q scheme, but use the following data streams.
    a. 0000000000000000
    b. 1111111111111111
    c. 0101010101010101
    d. 0011001100110011
20. Repeat Exercise 15 for the MLT-3 scheme, but use the following data streams.
    a. 00000000
    b. 11111111
    c. 01010101
    d. 00011000
21. Find the 8-bit data stream for each case depicted in Figure 4.36.

Figure 4.36 *Exercise 21*



a. NRZ-I

b. differential Manchester

c. AMI

22. An NRZ-I signal has a data rate of 100 Kbps. Using Figure 4.6, calculate the value of the normalized energy (P) for frequencies at 0 Hz, 50 KHz, and 100 KHz.
23. A Manchester signal has a data rate of 100 Kbps. Using Figure 4.8, calculate the value of the normalized energy (P) for frequencies at 0 Hz, 50 KHz, 100 KHz.

24. The input stream to a 4B/5B block encoder is 0100 0000 0000 0000 0000 0001. Answer the following questions:
    a. What is the output stream?
    b. What is the length of the longest consecutive sequence of 0s in the input?
    c. What is the length of the longest consecutive sequence of 0s in the output?
25. How many invalid (unused) code sequences can we have in 5B/6B encoding? How many in 3B/4B encoding?
26. What is the result of scrambling the sequence 11100000000000 using one of the following scrambling techniques? Assume that the last non-zero signal level has been positive.
    a. B8ZS
    b. HDB3 (The number of nonzero pules is odd after the last substitution)
27. What is the Nyquist sampling rate for each of the following signals?
    a. A low-pass signal with bandwidth of 200 KHz?
    b. A band-pass signal with bandwidth of 200 KHz if the lowest frequency is 100 KHz?
28. We have sampled a low-pass signal with a bandwidth of 200 KHz using 1024 levels of quantization.
    a. Calculate the bit rate of the digitized signal.
    b. Calculate the SNR_{dB} for this signal.
    c. Calculate the PCM bandwidth of this signal.
29. What is the maximum data rate of a channel with a bandwidth of 200 KHz if we use four levels of digital signaling.
30. An analog signal has a bandwidth of 20 KHz. If we sample this signal and send it through a 30 Kbps channel what is the SNR_{dB}?
31. We have a baseband channel with a 1-MHz bandwidth. What is the data rate for this channel if we use one of the following line coding schemes?
    a. NRZ-L
    b. Manchester
    c. MLT-3
    d. 2B1Q
32. We want to transmit 1000 characters with each character encoded as 8 bits.
    a. Find the number of transmitted bits for synchronous transmission.
    b. Find the number of transmitted bits for asynchronous transmission.
    c. Find the redundancy percent in each case.

CHAPTER 5

Analog Transmission

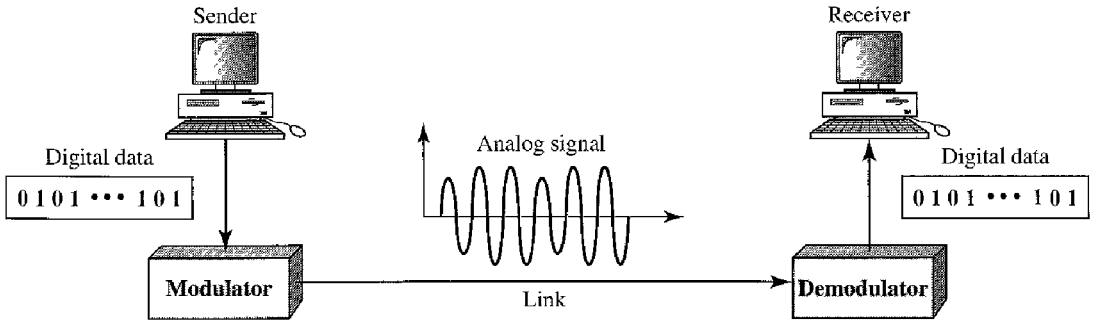
In Chapter 3, we discussed the advantages and disadvantages of digital and analog transmission. We saw that while digital transmission is very desirable, a low-pass channel is needed. We also saw that analog transmission is the only choice if we have a bandpass channel. Digital transmission was discussed in Chapter 4; we discuss analog transmission in this chapter.

Converting digital data to a bandpass analog signal is traditionally called digital-to-analog conversion. Converting a low-pass analog signal to a bandpass analog signal is traditionally called analog-to-analog conversion. In this chapter, we discuss these two types of conversions.

5.1 DIGITAL-TO-ANALOG CONVERSION

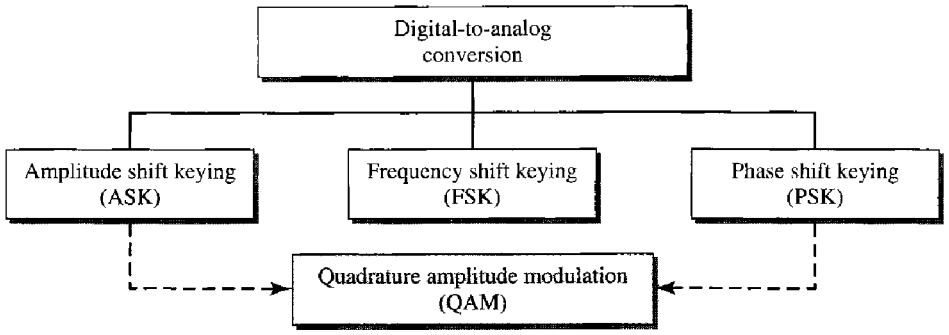
Digital-to-analog conversion is the process of changing one of the characteristics of an analog signal based on the information in digital data. Figure 5.1 shows the relationship between the digital information, the digital-to-analog modulating process, and the resultant analog signal.

Figure 5.1 *Digital-to-analog conversion*



As discussed in Chapter 3, a sine wave is defined by three characteristics: amplitude, frequency, and phase. When we vary any one of these characteristics, we create a different version of that wave. So, by changing one characteristic of a simple electric signal, we can use it to represent digital data. Any of the three characteristics can be altered in this way, giving us at least three mechanisms for modulating digital data into an analog signal: **amplitude shift keying (ASK), frequency shift keying (FSK),** and **phase shift keying (PSK).** In addition, there is a fourth (and better) mechanism that combines changing both the amplitude and phase, called **quadrature amplitude modulation (QAM).** QAM is the most efficient of these options and is the mechanism commonly used today (see Figure 5.2).

Figure 5.2 *Types of digital-to-analog conversion*



Aspects of Digital-to-Analog Conversion

Before we discuss specific methods of digital-to-analog modulation, two basic issues must be reviewed: bit and baud rates and the carrier signal.

Data Element Versus Signal Element

In Chapter 4, we discussed the concept of the data element versus the signal element. We defined a data element as the smallest piece of information to be exchanged, the bit. We also defined a signal element as the smallest unit of a signal that is constant. Although we continue to use the same terms in this chapter, we will see that the nature of the signal element is a little bit different in analog transmission.

Data Rate Versus Signal Rate

We can define the data rate (bit rate) and the signal rate (baud rate) as we did for digital transmission. The relationship between them is

$$S = N \times \frac{1}{r} \quad \textbf{baud}$$

where N is the data rate (bps) and r is the number of data elements carried in one signal element. The value of r in analog transmission is $r = \log_2 L$, where L is the type of signal element, not the level. The same nomenclature is used to simplify the comparisons.

Bit rate is the number of bits per second. Baud rate is the number of signal elements per second. In the analog transmission of digital data, the baud rate is less than or equal to the bit rate.

The same analogy we used in Chapter 4 for bit rate and baud rate applies here. In transportation, a baud is analogous to a vehicle, and a bit is analogous to a passenger. We need to maximize the number of people per car to reduce the traffic.

Example 5.1

An analog signal carries 4 bits per signal element. If 1000 signal elements are sent per second, find the bit rate.

Solution

In this case, $r = 4$, $S = 1000$, and N is unknown. We can find the value of N from

$$S = N \times \frac{1}{r} \qquad \text{or} \qquad N = S \times r = 1000 \times 4 = 4000 \text{ bps}$$

Example 5.2

An analog signal has a bit rate of 8000 bps and a baud rate of 1000 baud. How many data elements are carried by each signal element? How many signal elements do we need?

Solution

In this example, $S = 1000$, $N = 8000$, and r and L are unknown. We find first the value of r and then the value of L.

$$S = N \times \frac{1}{r} \quad \Rightarrow \quad r = \frac{N}{S} = \frac{8000}{1000} = 8 \text{ bits/baud}$$

$$r = \log_2 L \quad \Rightarrow \quad L = 2^r = 2^8 = 256$$

Bandwidth

The required bandwidth for analog transmission of digital data is proportional to the signal rate except for FSK, in which the difference between the carrier signals needs to be added. We discuss the bandwidth for each technique.

Carrier Signal

In analog transmission, the sending device produces a high-frequency signal that acts as a base for the information signal. This base signal is called the **carrier signal** or carrier frequency. The receiving device is tuned to the frequency of the carrier signal that it expects from the sender. Digital information then changes the carrier signal by modifying one or more of its characteristics (amplitude, frequency, or phase). This kind of modification is called modulation (shift keying).

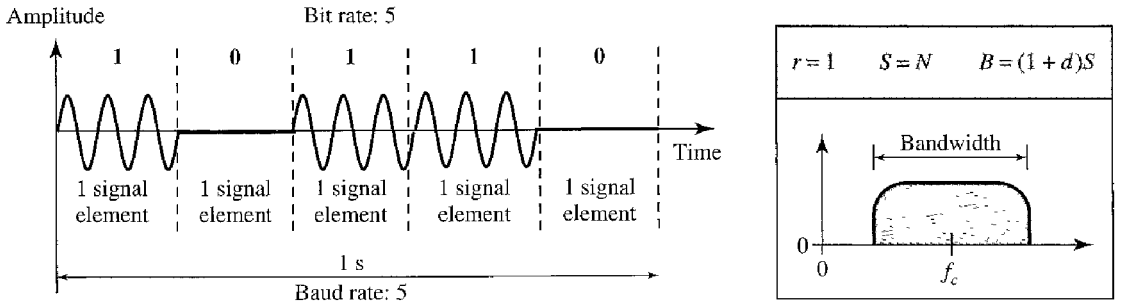
Amplitude Shift Keying

In amplitude shift keying, the amplitude of the carrier signal is varied to create signal elements. Both frequency and phase remain constant while the amplitude changes.

Binary ASK (BASK)

Although we can have several levels (kinds) of signal elements, each with a different amplitude, ASK is normally implemented using only two levels. This is referred to as binary amplitude shift keying or *on-off keying* (OOK). The peak amplitude of one signal level is 0; the other is the same as the amplitude of the carrier frequency. Figure 5.3 gives a conceptual view of binary ASK.

Figure 5.3 *Binary amplitude shift keying*



Bandwidth for ASK Figure 5.3 also shows the bandwidth for ASK. Although the carrier signal is only one simple sine wave, the process of modulation produces a nonperiodic composite signal. This signal, as was discussed in Chapter 3, has a continuous set of frequencies. As we expect, the bandwidth is proportional to the signal rate (baud rate). However, there is normally another factor involved, called d, which depends on the modulation and filtering process. The value of d is between 0 and 1. This means that the bandwidth can be expressed as shown, where S is the signal rate and the B is the bandwidth.

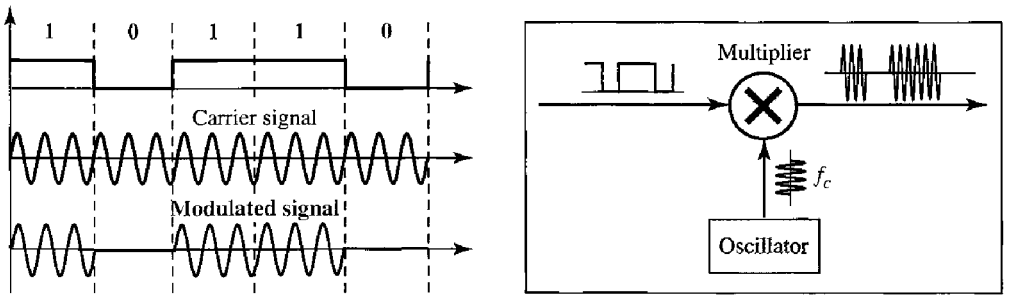
$$B = (1 + d) \times S$$

The formula shows that the required bandwidth has a minimum value of S and a maximum value of $2S$. The most important point here is the location of the bandwidth. The middle of the bandwidth is where f_c, the carrier frequency, is located. This means if we have a bandpass channel available, we can choose our f_c so that the modulated signal occupies that bandwidth. This is in fact the most important advantage of digital-to-analog conversion. We can shift the resulting bandwidth to match what is available.

Implementation The complete discussion of ASK implementation is beyond the scope of this book. However, the simple ideas behind the implementation may help us to better understand the concept itself. Figure 5.4 shows how we can simply implement binary ASK.

If digital data are presented as a unipolar NRZ (see Chapter 4) digital signal with a high voltage of 1 V and a low voltage of 0 V, the implementation can achieved by multiplying the NRZ digital signal by the carrier signal coming from an oscillator. When the amplitude of the NRZ signal is 1, the amplitude of the carrier frequency is

Figure 5.4 *Implementation of binary ASK*



held; when the amplitude of the NRZ signal is 0, the amplitude of the carrier frequency is zero.

Example 5.3

We have an available bandwidth of 100 kHz which spans from 200 to 300 kHz. What are the carrier frequency and the bit rate if we modulated our data by using ASK with $d = 1$?

Solution

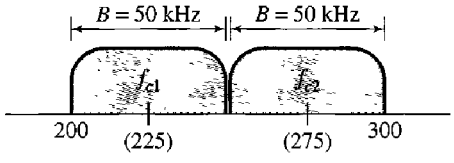
The middle of the bandwidth is located at 250 kHz. This means that our carrier frequency can be at $f_c = 250$ kHz. We can use the formula for bandwidth to find the bit rate (with $d = 1$ and $r = 1$).

$$B = (1 + d) \times S = 2 \times N \times \frac{1}{r} = 2 \times N = 100 \text{ kHz} \quad \Longrightarrow \quad N = 50 \text{ kbps}$$

Example 5.4

In data communications, we normally use full-duplex links with communication in both directions. We need to divide the bandwidth into two with two carrier frequencies, as shown in Figure 5.5. The figure shows the positions of two carrier frequencies and the bandwidths.The available bandwidth for each direction is now 50 kHz, which leaves us with a data rate of 25 kbps in each direction.

Figure 5.5 *Bandwidth of full-duplex ASK used in Example 5.4*



Multilevel ASK

The above discussion uses only two amplitude levels. We can have multilevel ASK in which there are more than two levels. We can use 4, 8, 16, or more different amplitudes for the signal and modulate the data using 2, 3, 4, or more bits at a time. In these cases,

$r = 2$, $r = 3$, $r = 4$, and so on. Although this is not implemented with pure ASK, it is implemented with QAM (as we will see later).

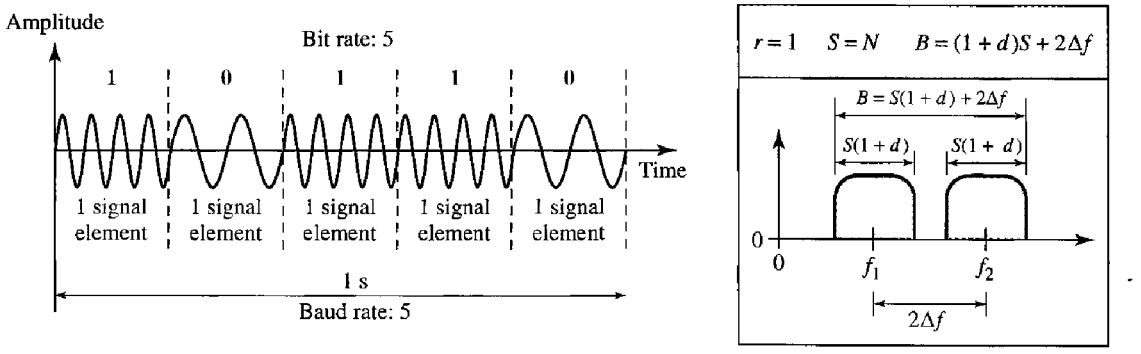
Frequency Shift Keying

In frequency shift keying, the frequency of the carrier signal is varied to represent data. The frequency of the modulated signal is constant for the duration of one signal element, but changes for the next signal element if the data element changes. Both peak amplitude and phase remain constant for all signal elements.

Binary FSK (BFSK)

One way to think about binary FSK (or BFSK) is to consider two carrier frequencies. In Figure 5.6, we have selected two carrier frequencies, f_1 and f_2. We use the first carrier if the data element is 0; we use the second if the data element is 1. However, note that this is an unrealistic example used only for demonstration purposes. Normally the carrier frequencies are very high, and the difference between them is very small.

Figure 5.6 *Binary frequency shift keying*



As Figure 5.6 shows, the middle of one bandwidth is f_1 and the middle of the other is f_2. Both f_1 and f_2 are Δf apart from the midpoint between the two bands. The difference between the two frequencies is $2\Delta f$.

Bandwidth for BFSK Figure 5.6 also shows the bandwidth of FSK. Again the carrier signals are only simple sine waves, but the modulation creates a nonperiodic composite signal with continuous frequencies. We can think of FSK as two ASK signals, each with its own carrier frequency (f_1 or f_2). If the difference between the two frequencies is $2\Delta f$, then the required bandwidth is

$$B = (1 + d) \times S + 2\Delta f$$

What should be the minimum value of $2\Delta f$? In Figure 5.6, we have chosen a value greater than $(1 + d)S$. It can be shown that the minimum value should be at least S for the proper operation of modulation and demodulation.

Example 5.5

We have an available bandwidth of 100 kHz which spans from 200 to 300 kHz. What should be the carrier frequency and the bit rate if we modulated our data by using FSK with $d = 1$?

Solution

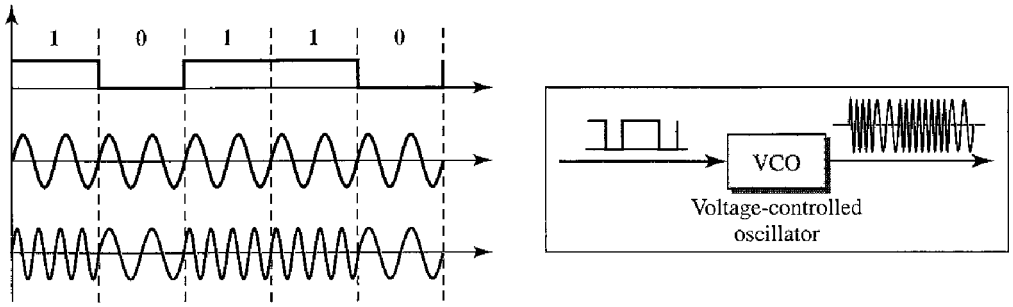
This problem is similar to Example 5.3, but we are modulating by using FSK. The midpoint of the band is at 250 kHz. We choose $2\Delta f$ to be 50 kHz; this means

$$B = (1 + d) \times S + 2\Delta f = 100 \quad \longrightarrow \quad 2S = 50 \text{ kHz} \quad S = 25 \text{ kbaud} \quad N = 25 \text{ kbps}$$

Compared to Example 5.3, we can see the bit rate for ASK is 50 kbps while the bit rate for FSK is 25 kbps.

Implementation There are two implementations of BFSK: noncoherent and coherent. In noncoherent BFSK, there may be discontinuity in the phase when one signal element ends and the next begins. In coherent BFSK, the phase continues through the boundary of two signal elements. Noncoherent BFSK can be implemented by treating BFSK as two ASK modulations and using two carrier frequencies. Coherent BFSK can be implemented by using one *voltage-controlled oscillator* (VCO) that changes its frequency according to the input voltage. Figure 5.7 shows the simplified idea behind the second implementation. The input to the oscillator is the unipolar NRZ signal. When the amplitude of NRZ is zero, the oscillator keeps its regular frequency; when the amplitude is positive, the frequency is increased.

Figure 5.7 *Implementation of BFSK*



Multilevel FSK

Multilevel modulation (MFSK) is not uncommon with the FSK method. We can use more than two frequencies. For example, we can use four different frequencies f_1, f_2, f_3, and f_4 to send 2 bits at a time. To send 3 bits at a time, we can use eight frequencies. And so on. However, we need to remember that the frequencies need to be $2\Delta f$ apart. For the proper operation of the modulator and demodulator, it can be shown that the minimum value of $2\Delta f$ needs to be S. We can show that the bandwidth with $d = 0$ is

$$B = (1 + d) \times S + (L - 1)2\Delta f \quad \longrightarrow \quad B = L \times S$$

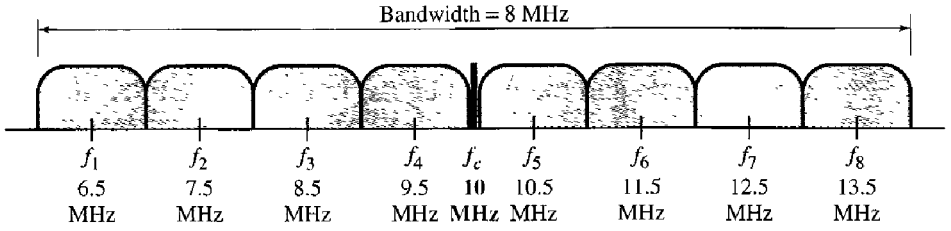
Example 5.6

We need to send data 3 bits at a time at a bit rate of 3 Mbps. The carrier frequency is 10 MHz. Calculate the number of levels (different frequencies), the baud rate, and the bandwidth.

Solution

We can have $L = 2^3 = 8$. The baud rate is $S = 3$ MHz/3 = 1000 Mbaud. This means that the carrier frequencies must be 1 MHz apart ($2\Delta f = 1$ MHz). The bandwidth is $B = 8 \times 1000 = 8000$. Figure 5.8 shows the allocation of frequencies and bandwidth.

Figure 5.8 *Bandwidth of MFSK used in Example 5.6*



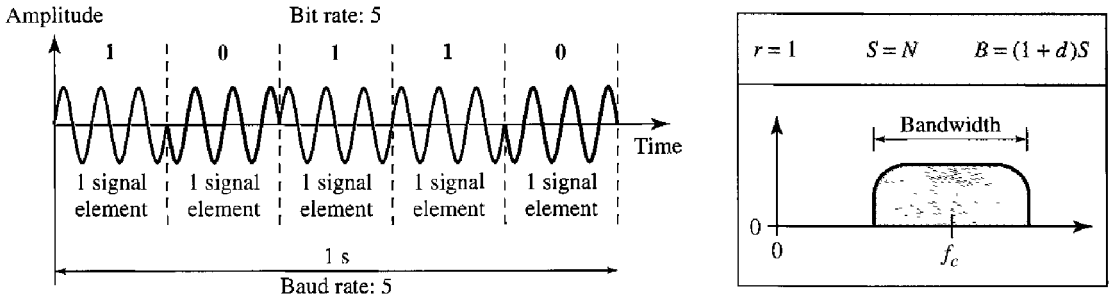
Phase Shift Keying

In phase shift keying, the phase of the carrier is varied to represent two or more different signal elements. Both peak amplitude and frequency remain constant as the phase changes. Today, PSK is more common than ASK or FSK. However, we will see shortly that QAM, which combines ASK and PSK, is the dominant method of digital-to-analog modulation.

Binary PSK (BPSK)

The simplest PSK is binary PSK, in which we have only two signal elements, one with a phase of 0°, and the other with a phase of 180°. Figure 5.9 gives a conceptual view of PSK. Binary PSK is as simple as binary ASK with one big advantage—it is less

Figure 5.9 *Binary phase shift keying*

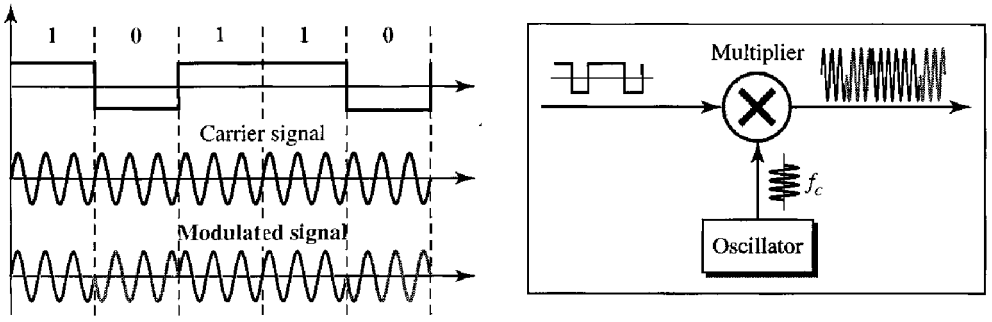


susceptible to noise. In ASK, the criterion for bit detection is the amplitude of the signal; in PSK, it is the phase. Noise can change the amplitude easier than it can change the phase. In other words, PSK is less susceptible to noise than ASK. PSK is superior to FSK because we do not need two carrier signals.

Bandwidth Figure 5.9 also shows the bandwidth for BPSK. The bandwidth is the same as that for binary ASK, but less than that for BFSK. No bandwidth is wasted for separating two carrier signals.

Implementation The implementation of BPSK is as simple as that for ASK. The reason is that the signal element with phase 180° can be seen as the complement of the signal element with phase 0°. This gives us a clue on how to implement BPSK. We use the same idea we used for ASK but with a polar NRZ signal instead of a unipolar NRZ signal, as shown in Figure 5.10. The polar NRZ signal is multiplied by the carrier frequency; the 1 bit (positive voltage) is represented by a phase starting at 0°; the 0 bit (negative voltage) is represented by a phase starting at 180°.

Figure 5.10 *Implementation of BASK*

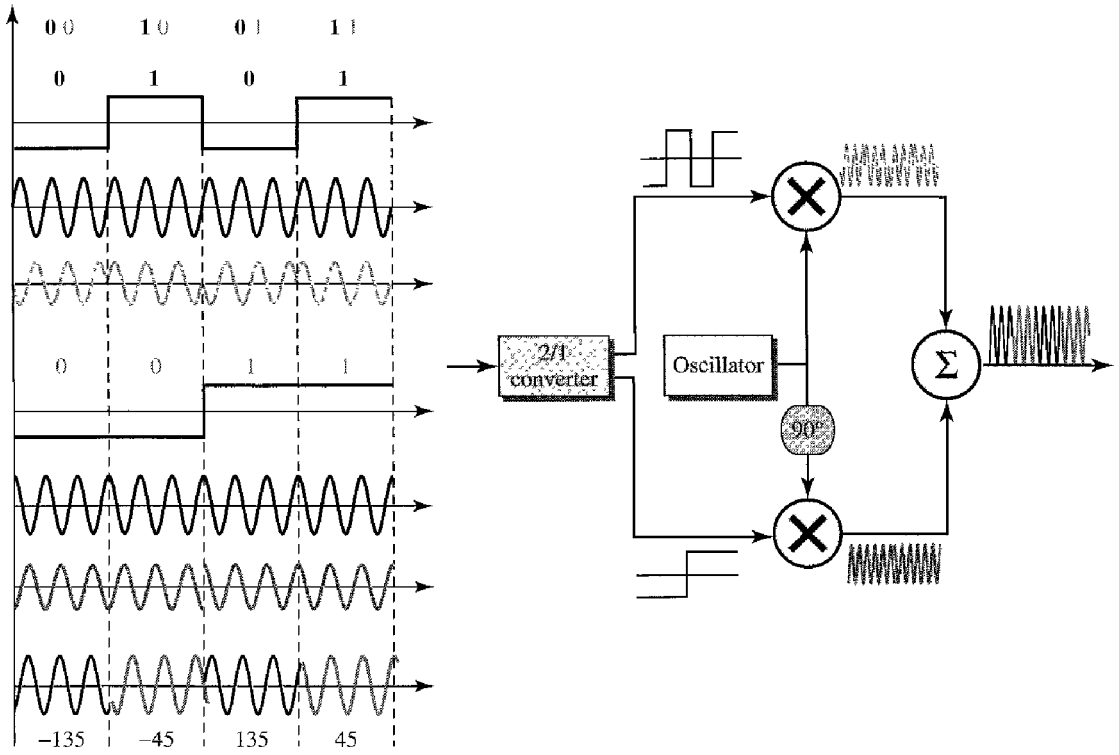


Quadrature PSK (QPSK)

The simplicity of BPSK enticed designers to use 2 bits at a time in each signal element, thereby decreasing the baud rate and eventually the required bandwidth. The scheme is called quadrature PSK or QPSK because it uses two separate BPSK modulations; one is in-phase, the other quadrature (out-of-phase). The incoming bits are first passed through a serial-to-parallel conversion that sends one bit to one modulator and the next bit to the other modulator. If the duration of each bit in the incoming signal is T, the duration of each bit sent to the corresponding BPSK signal is $2T$. This means that the bit to each BPSK signal has one-half the frequency of the original signal. Figure 5.11 shows the idea.

The two composite signals created by each multiplier are sine waves with the same frequency, but different phases. When they are added, the result is another sine wave, with one of four possible phases: 45°, –45°, 135°, and –135°. There are four kinds of signal elements in the output signal ($L = 4$), so we can send 2 bits per signal element ($r = 2$).

Figure 5.11 *QPSK and its implementation*



Example 5.7

Find the bandwidth for a signal transmitting at 12 Mbps for QPSK. The value of $d = 0$.

Solution

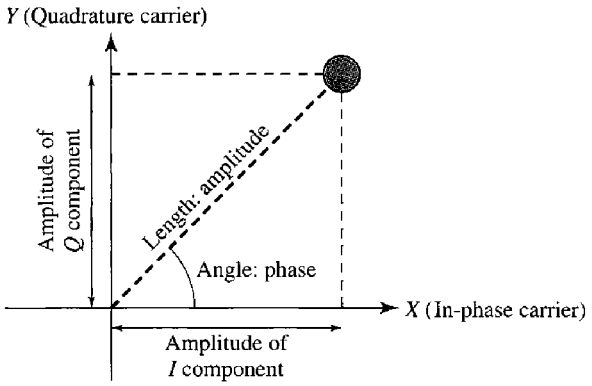
For QPSK, 2 bits is carried by one signal element. This means that $r = 2$. So the signal rate (baud rate) is $S = N \times (1/r) = 6$ Mbaud. With a value of $d = 0$, we have $B = S = 6$ MHz.

Constellation Diagram

A **constellation diagram** can help us define the amplitude and phase of a signal element, particularly when we are using two carriers (one in-phase and one quadrature). The diagram is useful when we are dealing with multilevel ASK, PSK, or QAM (see next section). In a constellation diagram, a signal element type is represented as a dot. The bit or combination of bits it can carry is often written next to it.

The diagram has two axes. The horizontal X axis is related to the in-phase carrier; the vertical Y axis is related to the quadrature carrier. For each point on the diagram, four pieces of information can be deduced. The projection of the point on the X axis defines the peak amplitude of the in-phase component; the projection of the point on the Y axis defines the peak amplitude of the quadrature component. The length of the line (vector) that connects the point to the origin is the peak amplitude of the signal element (combination of the X and Y components); the angle the line makes with the X axis is the phase of the signal element. All the information we need, can easily be found on a constellation diagram. Figure 5.12 shows a constellation diagram.

Figure 5.12 *Concept of a constellation diagram*



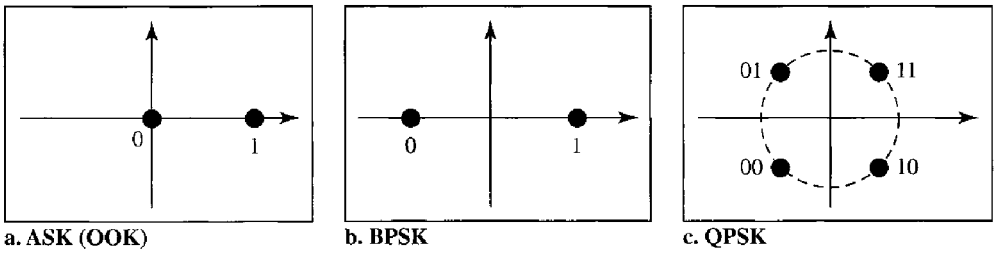
Example 5.8

Show the constellation diagrams for an ASK (OOK), BPSK, and QPSK signals.

Solution

Figure 5.13 shows the three constellation diagrams.

Figure 5.13 *Three constellation diagrams*



Let us analyze each case separately:

a. For ASK, we are using only an in-phase carrier. Therefore, the two points should be on the X axis. Binary 0 has an amplitude of 0 V; binary 1 has an amplitude of 1 V (for example). The points are located at the origin and at 1 unit.

b. BPSK also uses only an in-phase carrier. However, we use a polar NRZ signal for modulation. It creates two types of signal elements, one with amplitude 1 and the other with amplitude −1. This can be stated in other words: BPSK creates two different signal elements, one with amplitude 1 V and in phase and the other with amplitude 1 V and 180° out of phase.

c. QPSK uses two carriers, one in-phase and the other quadrature. The point representing 11 is made of two combined signal elements, both with an amplitude of 1 V. One element is represented by an in-phase carrier, the other element by a quadrature carrier. The amplitude of the final signal element sent for this 2-bit data element is $2^{1/2}$, and the phase is 45°. The argument is similar for the other three points. All signal elements have an amplitude of $2^{1/2}$, but their phases are different (45°, 135°, −135°, and −45°). Of course, we could have chosen the amplitude of the carrier to be $1/(2^{1/2})$ to make the final amplitudes 1 V.

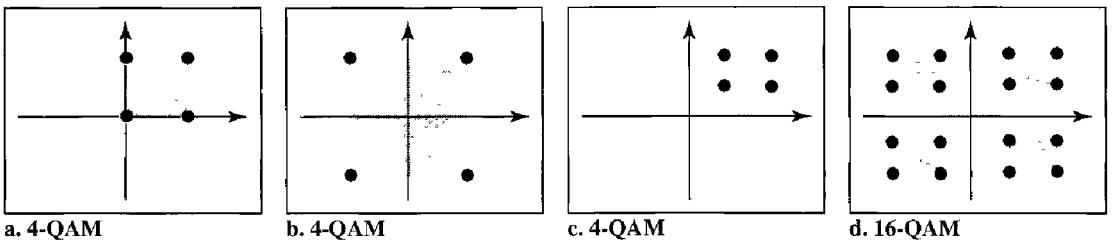
Quadrature Amplitude Modulation

PSK is limited by the ability of the equipment to distinguish small differences in phase. This factor limits its potential bit rate. So far, we have been altering only one of the three characteristics of a sine wave at a time; but what if we alter two? Why not combine ASK and PSK? The idea of using two carriers, one in-phase and the other quadrature, with different amplitude levels for each carrier is the concept behind **quadrature amplitude modulation (QAM).**

Quadrature amplitude modulation is a combination of ASK and PSK.

The possible variations of QAM are numerous. Figure 5.14 shows some of these schemes. Figure 5.14a shows the simplest 4-QAM scheme (four different signal element types) using a unipolar NRZ signal to modulate each carrier. This is the same mechanism we used for ASK (OOK). Part b shows another 4-QAM using polar NRZ, but this is exactly the same as QPSK. Part c shows another QAM-4 in which we used a signal with two positive levels to modulate each of the two carriers. Finally, Figure 5.14d shows a 16-QAM constellation of a signal with eight levels, four positive and four negative.

Figure 5.14 *Constellation diagrams for some QAMs*



a. 4-QAM b. 4-QAM c. 4-QAM d. 16-QAM

Bandwidth for QAM

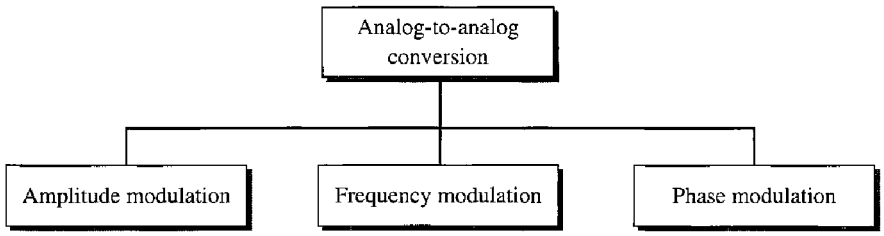
The minimum bandwidth required for QAM transmission is the same as that required for ASK and PSK transmission. QAM has the same advantages as PSK over ASK.

5.2 ANALOG-TO-ANALOG CONVERSION

Analog-to-analog conversion, or analog modulation, is the representation of analog information by an analog signal. One may ask why we need to modulate an analog signal; it is already analog. Modulation is needed if the medium is bandpass in nature or if only a bandpass channel is available to us. An example is radio. The government assigns a narrow bandwidth to each radio station. The analog signal produced by each station is a low-pass signal, all in the same range. To be able to listen to different stations, the low-pass signals need to be shifted, each to a different range.

Analog-to-analog conversion can be accomplished in three ways: **amplitude modulation (AM), frequency modulation (FM),** and **phase modulation (PM).** FM and PM are usually categorized together. See Figure 5.15.

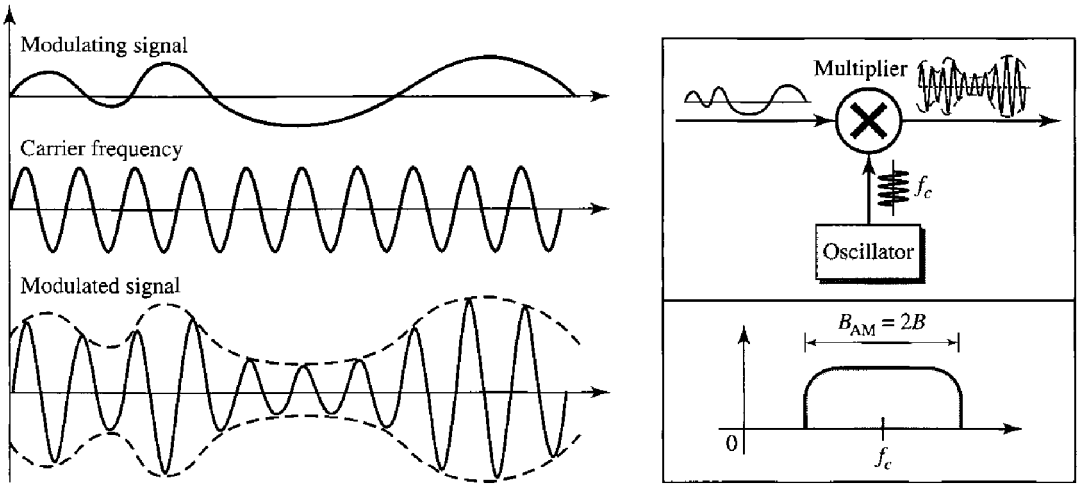
Figure 5.15 *Types of analog-to-analog modulation*



Amplitude Modulation

In AM transmission, the carrier signal is modulated so that its amplitude varies with the changing amplitudes of the modulating signal. The frequency and phase of the carrier remain the same; only the amplitude changes to follow variations in the information. Figure 5.16 shows how this concept works. The modulating signal is the envelope of the carrier.

Figure 5.16 *Amplitude modulation*



As Figure 5.16 shows, AM is normally implemented by using a simple multiplier because the amplitude of the carrier signal needs to be changed according to the amplitude of the modulating signal.

AM Bandwidth

Figure 5.16 also shows the bandwidth of an AM signal. The modulation creates a bandwidth that is twice the bandwidth of the modulating signal and covers a range centered on the carrier frequency. However, the signal components above and below the carrier

frequency carry exactly the same information. For this reason, some implementations discard one-half of the signals and cut the bandwidth in half.

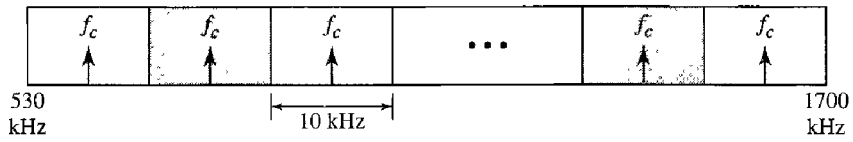
The total bandwidth required for AM can be determined from the bandwidth of the audio signal: $B_{AM} = 2B$.

Standard Bandwidth Allocation for AM Radio

The bandwidth of an audio signal (speech and music) is usually 5 kHz. Therefore, an AM radio station needs a bandwidth of 10 kHz. In fact, the Federal Communications Commission (FCC) allows 10 kHz for each AM station.

AM stations are allowed carrier frequencies anywhere between 530 and 1700 kHz (1.7 MHz). However, each station's carrier frequency must be separated from those on either side of it by at least 10 kHz (one AM bandwidth) to avoid interference. If one station uses a carrier frequency of 1100 kHz, the next station's carrier frequency cannot be lower than 1110 kHz (see Figure 5.17).

Figure 5.17 *AM band allocation*



Frequency Modulation

In FM transmission, the frequency of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal. The peak amplitude and phase of the carrier signal remain constant, but as the amplitude of the information signal changes, the frequency of the carrier changes correspondingly. Figure 5.18 shows the relationships of the modulating signal, the carrier signal, and the resultant FM signal.

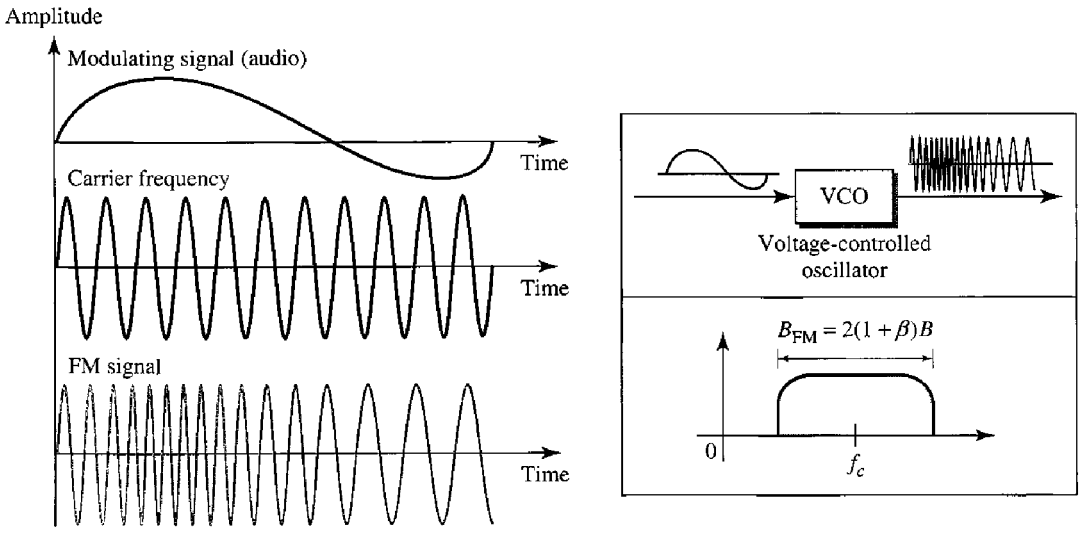
As Figure 5.18 shows, FM is normally implemented by using a voltage-controlled oscillator as with FSK. The frequency of the oscillator changes according to the input voltage which is the amplitude of the modulating signal.

FM Bandwidth

Figure 5.18 also shows the bandwidth of an FM signal. The actual bandwidth is difficult to determine exactly, but it can be shown empirically that it is several times that of the analog signal or $2(1 + \beta)B$ where β is a factor depends on modulation technique with a common value of 4.

The total bandwidth required for FM can be determined from the bandwidth of the audio signal: $B_{FM} = 2(1 + \beta)B$.

Figure 5.18 *Frequency modulation*



Standard Bandwidth Allocation for FM Radio

The bandwidth of an audio signal (speech and music) broadcast in stereo is almost 15 kHz. The FCC allows 200 kHz (0.2 MHz) for each station. This mean $\beta = 4$ with some extra guard band. FM stations are allowed carrier frequencies anywhere between 88 and 108 MHz. Stations must be separated by at least 200 kHz to keep their bandwidths from overlapping. To create even more privacy, the FCC requires that in a given area, only alternate bandwidth allocations may be used. The others remain unused to prevent any possibility of two stations interfering with each other. Given 88 to 108 MHz as a range, there are 100 potential FM bandwidths in an area, of which 50 can operate at any one time. Figure 5.19 illustrates this concept.

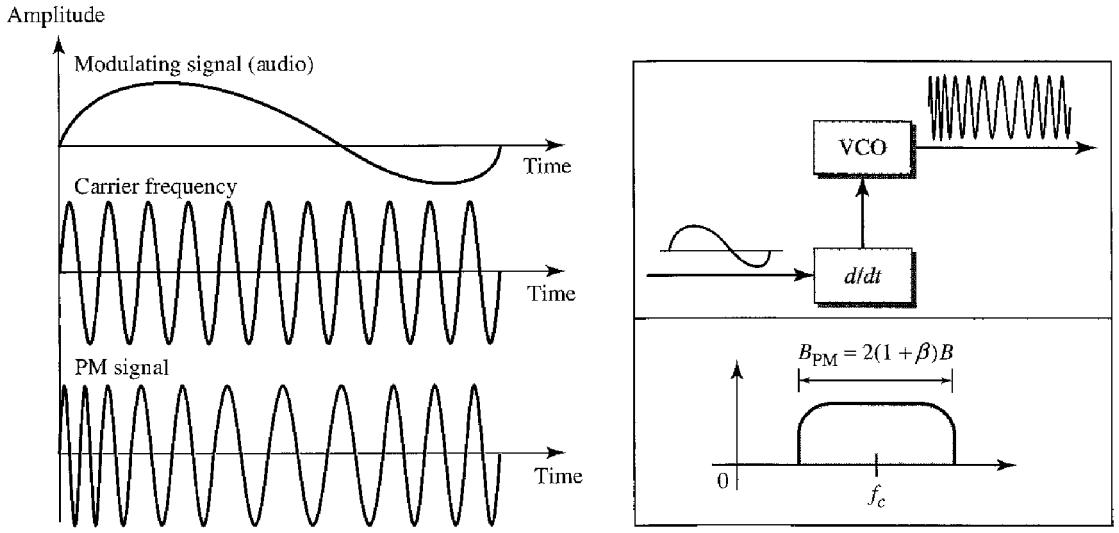
Figure 5.19 *FM band allocation*

Phase Modulation

In PM transmission, the phase of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal. The peak amplitude and frequency of the carrier signal remain constant, but as the amplitude of the information signal changes, the phase of the carrier changes correspondingly. It can proved mathematically (see Appendix C) that PM is the same as FM with one difference. In FM, the instantaneous change in the carrier frequency is proportional to the amplitude of the

modulating signal; in PM the instantaneous change in the carrier frequency is proportional to the derivative of the amplitude of the modulating signal. Figure 5.20 shows the relationships of the modulating signal, the carrier signal, and the resultant PM signal.

Figure 5.20 *Phase modulation*



As Figure 5.20 shows, PM is normally implemented by using a voltage-controlled oscillator along with a derivative. The frequency of the oscillator changes according to the derivative of the input voltage which is the amplitude of the modulating signal.

PM Bandwidth

Figure 5.20 also shows the bandwidth of a PM signal. The actual bandwidth is difficult to determine exactly, but it can be shown empirically that it is several times that of the analog signal. Although, the formula shows the same bandwidth for FM and PM, the value of β is lower in the case of PM (around 1 for narrowband and 3 for wideband).

The total bandwidth required for PM can be determined from the bandwidth and maximum amplitude of the modulating signal: $B_{PM} = 2(1 + \beta)B$.

5.3 RECOMMENDED READING

For more details about subjects discussed in this chapter, we recommend the following books. The items in brackets [. . .] refer to the reference list at the end of the text.

Books

Digital-to-analog conversion is discussed in Chapter 14 of [Pea92], Chapter 5 of [Cou01], and Section 5.2 of [Sta04]. Analog-to-analog conversion is discussed in Chapters 8 to 13 of [Pea92], Chapter 5 of [Cou01], and Section 5.4 of [Sta04]. [Hsu03]

gives a good mathematical approach to all materials discussed in this chapter. More advanced materials can be found in [Ber96].

5.4 KEY TERMS

amplitude modulation (AM)
amplitude shift keying (ASK)
analog-to-analog conversion
carrier signal
constellation diagram
digital-to-analog conversion
frequency modulation (FM)
frequency shift keying (FSK)
phase modulation (PM)
phase shift keying (PSK)
quadrature amplitude modulation (QAM)

5.5 SUMMARY

- Digital-to-analog conversion is the process of changing one of the characteristics of an analog signal based on the information in the digital data.
- Digital-to-analog conversion can be accomplished in several ways: amplitude shift keying (ASK), frequency shift keying (FSK), and phase shift keying (PSK). Quadrature amplitude modulation (QAM) combines ASK and PSK.
- In amplitude shift keying, the amplitude of the carrier signal is varied to create signal elements. Both frequency and phase remain constant while the amplitude changes.
- In frequency shift keying, the frequency of the carrier signal is varied to represent data. The frequency of the modulated signal is constant for the duration of one signal element, but changes for the next signal element if the data element changes. Both peak amplitude and phase remain constant for all signal elements.
- In phase shift keying, the phase of the carrier is varied to represent two or more different signal elements. Both peak amplitude and frequency remain constant as the phase changes.
- A constellation diagram shows us the amplitude and phase of a signal element, particularly when we are using two carriers (one in-phase and one quadrature).
- Quadrature amplitude modulation (QAM) is a combination of ASK and PSK. QAM uses two carriers, one in-phase and the other quadrature, with different amplitude levels for each carrier.
- Analog-to-analog conversion is the representation of analog information by an analog signal. Conversion is needed if the medium is bandpass in nature or if only a bandpass bandwidth is available to us.
- Analog-to-analog conversion can be accomplished in three ways: amplitude modulation (AM), frequency modulation (FM), and phase modulation (PM).
- In AM transmission, the carrier signal is modulated so that its amplitude varies with the changing amplitudes of the modulating signal. The frequency and phase of the carrier remain the same; only the amplitude changes to follow variations in the information.
- In FM transmission, the frequency of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal. The peak amplitude

and phase of the carrier signal remain constant, but as the amplitude of the information signal changes, the frequency of the carrier changes correspondingly.

- In PM transmission, the phase of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal. The peak amplitude and frequency of the carrier signal remain constant, but as the amplitude of the information signal changes, the phase of the carrier changes correspondingly.

5.6 PRACTICE SET

Review Questions

1. Define analog transmission.
2. Define carrier signal and its role in analog transmission.
3. Define digital-to-analog conversion.
4. Which characteristics of an analog signal are changed to represent the digital signal in each of the following digital-to-analog conversion?
   a. ASK
   b. FSK
   c. PSK
   d. QAM
5. Which of the four digital-to-analog conversion techniques (ASK, FSK, PSK or QAM) is the most susceptible to noise? Defend your answer.
6. Define constellation diagram and its role in analog transmission.
7. What are the two components of a signal when the signal is represented on a constellation diagram? Which component is shown on the horizontal axis? Which is shown on the vertical axis?
8. Define analog-to-analog conversion?
9. Which characteristics of an analog signal are changed to represent the lowpass analog signal in each of the following analog-to-analog conversions?
   a. AM
   b. FM
   c. PM
10. Which of the three analog-to-analog conversion techniques (AM, FM, or PM) is the most susceptible to noise? Defend your answer.

Exercises

11. Calculate the baud rate for the given bit rate and type of modulation.
    a. 2000 bps, FSK
    b. 4000 bps, ASK
    c. 6000 bps, QPSK
    d. 36,000 bps, 64-QAM

12. Calculate the bit rate for the given baud rate and type of modulation.
    a. 1000 baud, FSK
    b. 1000 baud, ASK
    c. 1000 baud, BPSK
    d. 1000 baud, 16-QAM
13. What is the number of bits per baud for the following techniques?
    a. ASK with four different amplitudes
    b. FSK with 8 different frequencies
    c. PSK with four different phases
    d. QAM with a constellation of 128 points.
14. Draw the constellation diagram for the following:
    a. ASK, with peak amplitude values of 1 and 3
    b. BPSK, with a peak amplitude value of 2
    c. QPSK, with a peak amplitude value of 3
    d. 8-QAM with two different peak amplitude values, 1 and 3, and four different phases.
15. Draw the constellation diagram for the following cases. Find the peak amplitude value for each case and define the type of modulation (ASK, FSK, PSK, or QAM). The numbers in parentheses define the values of I and Q respectively.
    a. Two points at (2, 0) and (3, 0).
    b. Two points at (3, 0) and (–3, 0).
    c. Four points at (2, 2), (–2, 2), (–2, –2), and (2, –2).
    d. Two points at (0 , 2) and (0, –2).
16. How many bits per baud can we send in each of the following cases if the signal constellation has one of the following number of points?
    a. 2
    b. 4
    c. 16
    d. 1024
17. What is the required bandwidth for the following cases if we need to send 4000 bps? Let d = 1.
    a. ASK
    b. FSK with $2\Delta f = 4$ KHz
    c. QPSK
    d. 16-QAM
18. The telephone line has 4 KHz bandwidth. What is the maximum number of bits we can send using each of the following techniques? Let d = 0.
    a. ASK
    b. QPSK
    c. 16-QAM
    d. 64-QAM

19. A corporation has a medium with a 1-MHz bandwidth (lowpass). The corporation needs to create 10 separate independent channels each capable of sending at least 10 Mbps. The company has decided to use QAM technology. What is the minimum number of bits per baud for each channel? What is the number of points in the constellation diagram for each channel? Let d = 0.
20. A cable company uses one of the cable TV channels (with a bandwidth of 6 MHz) to provide digital communication for each resident. What is the available data rate for each resident if the company uses a 64-QAM technique?
21. Find the bandwidth for the following situations if we need to modulate a 5-KHz voice.
    a. AM
    b. FM (set $\beta = 5$)
    c. PM (set $\beta = 1$)
22. Find the total number of channels in the corresponding band allocated by FCC.
    a. AM
    b. FM

Down

CHAPTER 6

Bandwidth Utilization: Multiplexing and Spreading

In real life, we have links with limited bandwidths. The wise use of these bandwidths has been, and will be, one of the main challenges of electronic communications. However, the meaning of *wise* may depend on the application. Sometimes we need to combine several low-bandwidth channels to make use of one channel with a larger bandwidth. Sometimes we need to expand the bandwidth of a channel to achieve goals such as privacy and antijamming. In this chapter, we explore these two broad categories of bandwidth utilization: multiplexing and spreading. In multiplexing, our goal is efficiency; we combine several channels into one. In spreading, our goals are privacy and antijamming; we expand the bandwidth of a channel to insert redundancy, which is necessary to achieve these goals.

Bandwidth utilization is the wise use of available bandwidth to achieve specific goals.

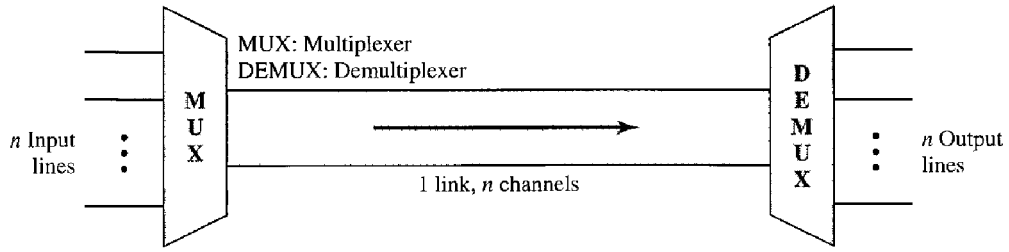
Efficiency can be achieved by multiplexing; privacy and antijamming can be achieved by spreading.

6.1 MULTIPLEXING

Whenever the bandwidth of a medium linking two devices is greater than the bandwidth needs of the devices, the link can be shared. **Multiplexing** is the set of techniques that allows the simultaneous transmission of multiple signals across a single data link. As data and telecommunications use increases, so does traffic. We can accommodate this increase by continuing to add individual links each time a new channel is needed; or we can install higher-bandwidth links and use each to carry multiple signals. As described in Chapter 7, today's technology includes high-bandwidth media such as optical fiber and terrestrial and satellite microwaves. Each has a bandwidth far in excess of that needed for the average transmission signal. If the bandwidth of a link is greater than the bandwidth needs of the devices connected to it, the bandwidth is wasted. An efficient system maximizes the utilization of all resources; bandwidth is one of the most precious resources we have in data communications.

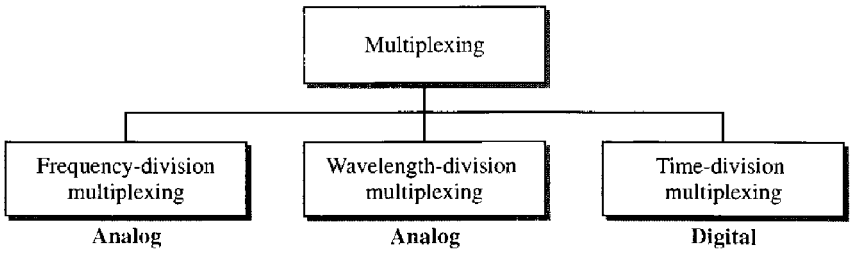
In a multiplexed system, n lines share the bandwidth of one link. Figure 6.1 shows the basic format of a multiplexed system. The lines on the left direct their transmission streams to a **multiplexer (MUX),** which combines them into a single stream (many-to-one). At the receiving end, that stream is fed into a **demultiplexer (DEMUX),** which separates the stream back into its component transmissions (one-to-many) and directs them to their corresponding lines. In the figure, the word **link** refers to the physical path. The word **channel** refers to the portion of a link that carries a transmission between a given pair of lines. One link can have many (n) channels.

Figure 6.1 *Dividing a link into channels*



There are three basic multiplexing techniques: frequency-division multiplexing, wavelength-division multiplexing, and time-division multiplexing. The first two are techniques designed for analog signals, the third, for digital signals (see Figure 6.2).

Figure 6.2 *Categories of multiplexing*



Although some textbooks consider *carrier division multiple access* (CDMA) as a fourth multiplexing category, we discuss CDMA as an access method (see Chapter 12).

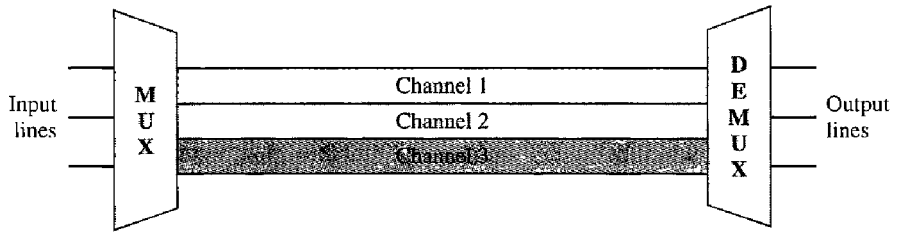
Frequency-Division Multiplexing

Frequency-division multiplexing (FDM) is an analog technique that can be applied when the bandwidth of a link (in hertz) is greater than the combined bandwidths of the signals to be transmitted. In FDM, signals generated by each sending device modulate different carrier frequencies. These modulated signals are then combined into a single composite signal that can be transported by the link. Carrier frequencies are separated by sufficient bandwidth to accommodate the modulated signal. These bandwidth ranges are the channels through which the various signals travel. Channels can be separated by

strips of unused bandwidth—**guard bands**—to prevent signals from overlapping. In addition, carrier frequencies must not interfere with the original data frequencies.

Figure 6.3 gives a conceptual view of FDM. In this illustration, the transmission path is divided into three parts, each representing a channel that carries one transmission.

Figure 6.3 *Frequency-division multiplexing*



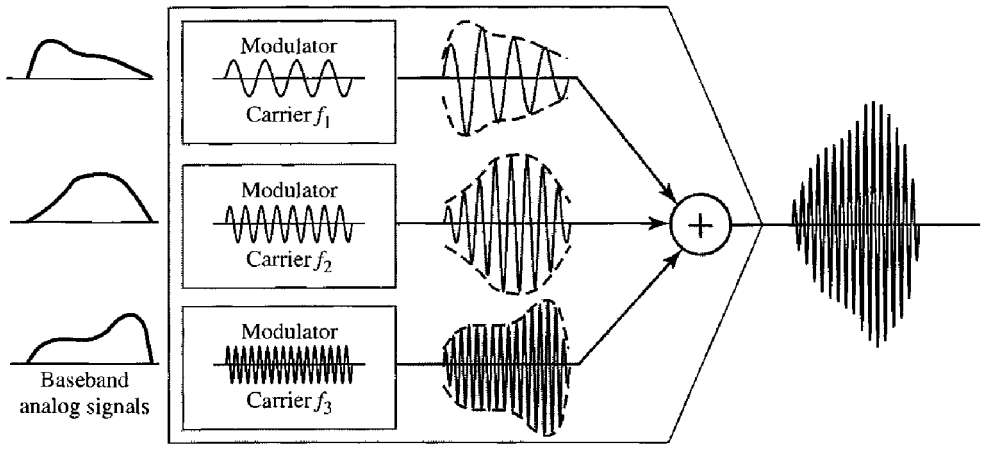
We consider FDM to be an analog multiplexing technique; however, this does not mean that FDM cannot be used to combine sources sending digital signals. A digital signal can be converted to an analog signal (with the techniques discussed in Chapter 5) before FDM is used to multiplex them.

FDM is an analog multiplexing technique that combines analog signals.

Multiplexing Process

Figure 6.4 is a conceptual illustration of the multiplexing process. Each source generates a signal of a similar frequency range. Inside the multiplexer, these similar signals modulates different carrier frequencies (f_1, f_2, and f_3). The resulting modulated signals are then combined into a single composite signal that is sent out over a media link that has enough bandwidth to accommodate it.

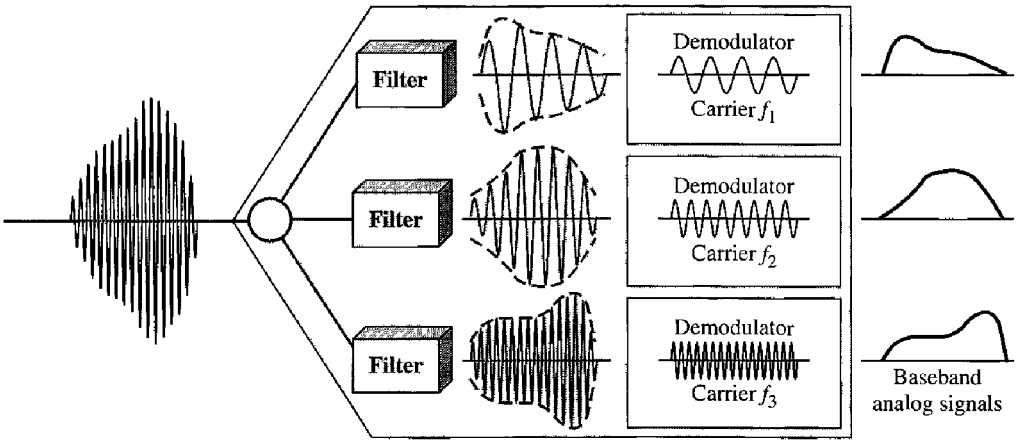
Figure 6.4 *FDM process*



Demultiplexing Process

The demultiplexer uses a series of filters to decompose the multiplexed signal into its constituent component signals. The individual signals are then passed to a demodulator that separates them from their carriers and passes them to the output lines. Figure 6.5 is a conceptual illustration of demultiplexing process.

Figure 6.5 *FDM demultiplexing example*



Example 6.1

Assume that a voice channel occupies a bandwidth of 4 kHz. We need to combine three voice channels into a link with a bandwidth of 12 kHz, from 20 to 32 kHz. Show the configuration, using the frequency domain. Assume there are no guard bands.

Solution

We shift (modulate) each of the three voice channels to a different bandwidth, as shown in Figure 6.6. We use the 20- to 24-kHz bandwidth for the first channel, the 24- to 28-kHz bandwidth for the second channel, and the 28- to 32-kHz bandwidth for the third one. Then we combine them as shown in Figure 6.6. At the receiver, each channel receives the entire signal, using a filter to separate out its own signal. The first channel uses a filter that passes frequencies between 20 and 24 kHz and filters out (discards) any other frequencies. The second channel uses a filter that passes frequencies between 24 and 28 kHz, and the third channel uses a filter that passes frequencies between 28 and 32 kHz. Each channel then shifts the frequency to start from zero.

Example 6.2

Five channels, each with a 100-kHz bandwidth, are to be multiplexed together. What is the minimum bandwidth of the link if there is a need for a guard band of 10 kHz between the channels to prevent interference?

Solution

For five channels, we need at least four guard bands. This means that the required bandwidth is at least $5 \times 100 + 4 \times 10 = 540$ kHz, as shown in Figure 6.7.

Figure 6.6 *Example 6.1*

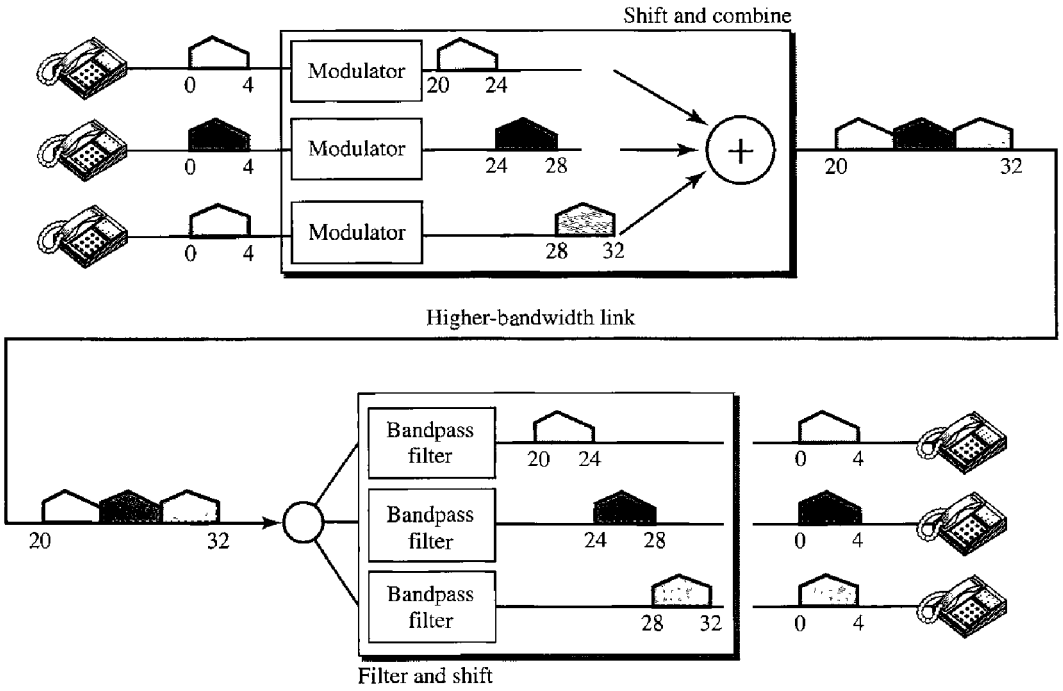
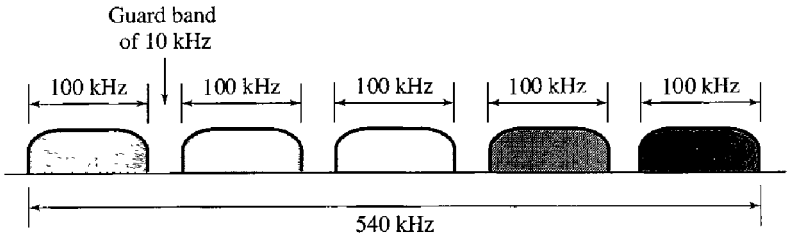


Figure 6.7 *Example 6.2*



Example 6.3

Four data channels (digital), each transmitting at 1 Mbps, use a satellite channel of 1 MHz. Design an appropriate configuration, using FDM.

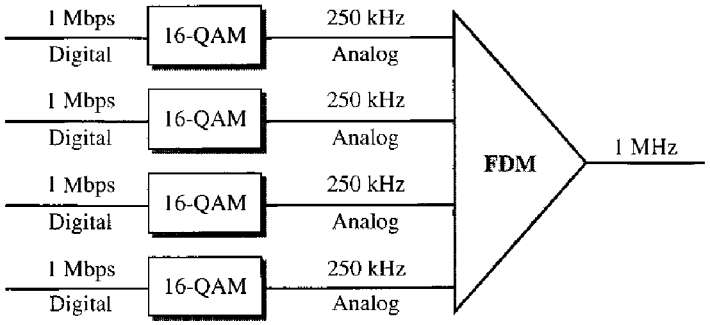
Solution

The satellite channel is analog. We divide it into four channels, each channel having a 250-kHz bandwidth. Each digital channel of 1 Mbps is modulated such that each 4 bits is modulated to 1 Hz. One solution is 16-QAM modulation. Figure 6.8 shows one possible configuration.

The Analog Carrier System

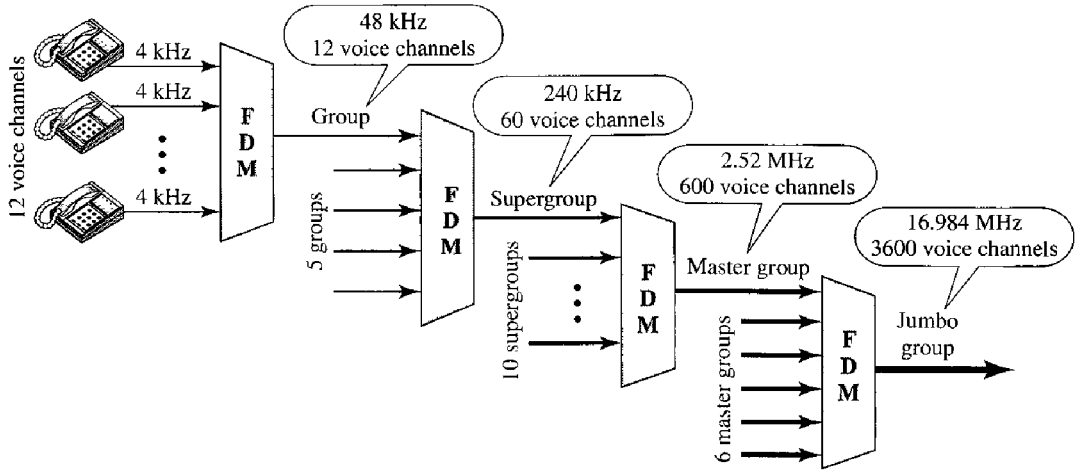
To maximize the efficiency of their infrastructure, telephone companies have traditionally multiplexed signals from lower-bandwidth lines onto higher-bandwidth lines. In this way, many switched or leased lines can be combined into fewer but bigger channels. For analog lines, FDM is used.

Figure 6.8 *Example 6.3*



One of these hierarchical systems used by AT&T is made up of groups, supergroups, master groups, and jumbo groups (see Figure 6.9).

Figure 6.9 *Analog hierarchy*



In this **analog hierarchy,** 12 voice channels are multiplexed onto a higher-bandwidth line to create a **group.** A group has 48 kHz of bandwidth and supports 12 voice channels.

At the next level, up to five groups can be multiplexed to create a composite signal called a **supergroup.** A supergroup has a bandwidth of 240 kHz and supports up to 60 voice channels. Supergroups can be made up of either five groups or 60 independent voice channels.

At the next level, 10 supergroups are multiplexed to create a **master group.** A master group must have 2.40 MHz of bandwidth, but the need for guard bands between the supergroups increases the necessary bandwidth to 2.52 MHz. Master groups support up to 600 voice channels.

Finally, six master groups can be combined into a **jumbo group.** A jumbo group must have 15.12 MHz (6×2.52 MHz) but is augmented to 16.984 MHz to allow for guard bands between the master groups.

Other Applications of FDM

A very common application of FDM is AM and FM radio broadcasting. Radio uses the air as the transmission medium. A special band from 530 to 1700 kHz is assigned to AM radio. All radio stations need to share this band. As discussed in Chapter 5, each AM station needs 10 kHz of bandwidth. Each station uses a different carrier frequency, which means it is shifting its signal and multiplexing. The signal that goes to the air is a combination of signals. A receiver receives all these signals, but filters (by tuning) only the one which is desired. Without multiplexing, only one AM station could broadcast to the common link, the air. However, we need to know that there is physical multiplexer or demultiplexer here. As we will see in Chapter 12 multiplexing is done at the data link layer.

The situation is similar in FM broadcasting. However, FM has a wider band of 88 to 108 MHz because each station needs a bandwidth of 200 kHz.

Another common use of FDM is in television broadcasting. Each TV channel has its own bandwidth of 6 MHz.

The first generation of cellular telephones (still in operation) also uses FDM. Each user is assigned two 30-kHz channels, one for sending voice and the other for receiving. The voice signal, which has a bandwidth of 3 kHz (from 300 to 3300 Hz), is modulated by using FM. Remember that an FM signal has a bandwidth 10 times that of the modulating signal, which means each channel has 30 kHz (10×3) of bandwidth. Therefore, each user is given, by the base station, a 60-kHz bandwidth in a range available at the time of the call.

Example 6.4

The *Advanced Mobile Phone System* (AMPS) uses two bands. The first band of 824 to 849 MHz is used for sending, and 869 to 894 MHz is used for receiving. Each user has a bandwidth of 30 kHz in each direction. The 3-kHz voice is modulated using FM, creating 30 kHz of modulated signal. How many people can use their cellular phones simultaneously?

Solution

Each band is 25 MHz. If we divide 25 MHz by 30 kHz, we get 833.33. In reality, the band is divided into 832 channels. Of these, 42 channels are used for control, which means only 790 channels are available for cellular phone users. We discuss AMPS in greater detail in Chapter 16.

Implementation

FDM can be implemented very easily. In many cases, such as radio and television broadcasting, there is no need for a physical multiplexer or demultiplexer. As long as the stations agree to send their broadcasts to the air using different carrier frequencies, multiplexing is achieved. In other cases, such as the cellular telephone system, a base station needs to assign a carrier frequency to the telephone user. There is not enough bandwidth in a cell to permanently assign a bandwidth range to every telephone user. When a user hangs up, her or his bandwidth is assigned to another caller.

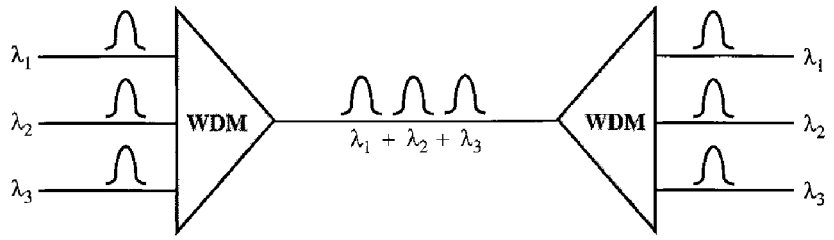
Wavelength-Division Multiplexing

Wavelength-division multiplexing (WDM) is designed to use the high-data-rate capability of fiber-optic cable. The optical fiber data rate is higher than the data rate of metallic transmission cable. Using a fiber-optic cable for one single line wastes the available bandwidth. Multiplexing allows us to combine several lines into one.

WDM is conceptually the same as FDM, except that the multiplexing and demultiplexing involve optical signals transmitted through fiber-optic channels. The idea is the same: We are combining different signals of different frequencies. The difference is that the frequencies are very high.

Figure 6.10 gives a conceptual view of a WDM multiplexer and demultiplexer. Very narrow bands of light from different sources are combined to make a wider band of light. At the receiver, the signals are separated by the demultiplexer.

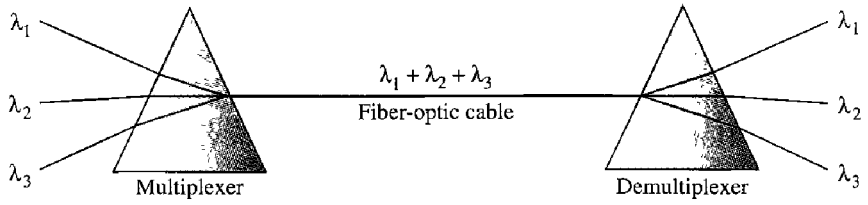
Figure 6.10 *Wavelength-division multiplexing*



WDM is an analog multiplexing technique to combine optical signals.

Although WDM technology is very complex, the basic idea is very simple. We want to combine multiple light sources into one single light at the multiplexer and do the reverse at the demultiplexer. The combining and splitting of light sources are easily handled by a prism. Recall from basic physics that a prism bends a beam of light based on the angle of incidence and the frequency. Using this technique, a multiplexer can be made to combine several input beams of light, each containing a narrow band of frequencies, into one output beam of a wider band of frequencies. A demultiplexer can also be made to reverse the process. Figure 6.11 shows the concept.

Figure 6.11 *Prisms in wavelength-division multiplexing and demultiplexing*



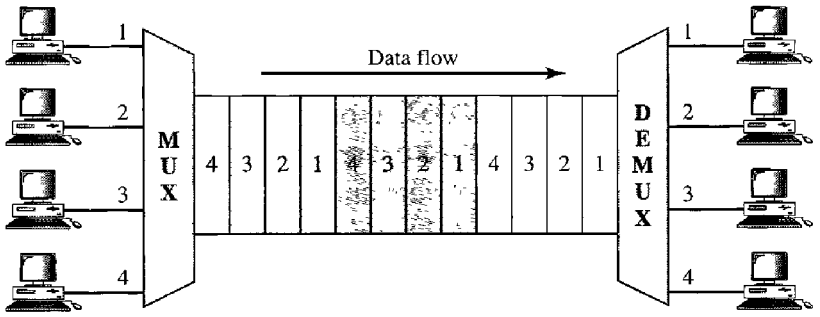
One application of WDM is the SONET network in which multiple optical fiber lines are multiplexed and demultiplexed. We discuss SONET in Chapter 17.

A new method, called **dense WDM (DWDM),** can multiplex a very large number of channels by spacing channels very close to one another. It achieves even greater efficiency.

Synchronous Time-Division Multiplexing

Time-division multiplexing (TDM) is a digital process that allows several connections to share the high bandwidth of a link. Instead of sharing a portion of the bandwidth as in FDM, time is shared. Each connection occupies a portion of time in the link. Figure 6.12 gives a conceptual view of TDM. Note that the same link is used as in FDM; here, however, the link is shown sectioned by time rather than by frequency. In the figure, portions of signals 1, 2, 3, and 4 occupy the link sequentially.

Figure 6.12 *TDM*



Note that in Figure 6.12 we are concerned with only multiplexing, not switching. This means that all the data in a message from source 1 always go to one specific destination, be it 1, 2, 3, or 4. The delivery is fixed and unvarying, unlike switching.

We also need to remember that TDM is, in principle, a digital multiplexing technique. Digital data from different sources are combined into one timeshared link. However, this does not mean that the sources cannot produce analog data; analog data can be sampled, changed to digital data, and then multiplexed by using TDM.

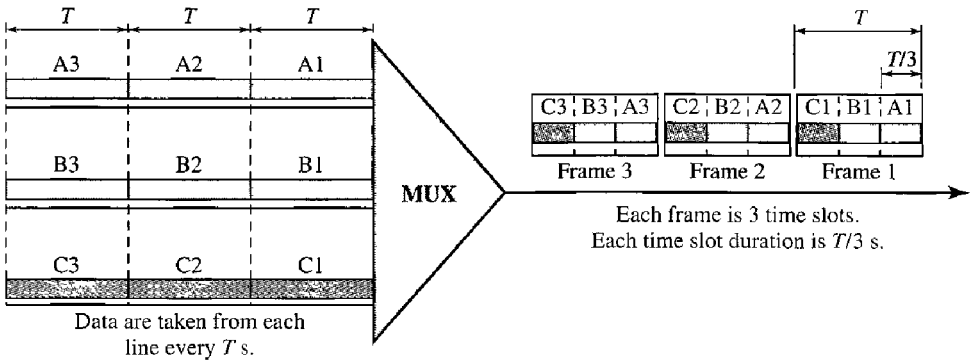
TDM is a digital multiplexing technique for combining several low-rate channels into one high-rate one.

We can divide TDM into two different schemes: synchronous and statistical. We first discuss **synchronous TDM** and then show how **statistical TDM** differs. In synchronous TDM, each input connection has an allotment in the output even if it is not sending data.

Time Slots and Frames

In synchronous TDM, the data flow of each input connection is divided into units, where each input occupies one input time slot. A unit can be 1 bit, one character, or one block of data. Each input unit becomes one output unit and occupies one output time slot. However, the duration of an output time slot is n times shorter than the duration of an input time slot. If an input time slot is T s, the output time slot is T/n s, where n is the number of connections. In other words, a unit in the output connection has a shorter duration; it travels faster. Figure 6.13 shows an example of synchronous TDM where n is 3.

Figure 6.13 *Synchronous time-division multiplexing*



In synchronous TDM, a round of data units from each input connection is collected into a frame (we will see the reason for this shortly). If we have n connections, a frame is divided into n time slots and one slot is allocated for each unit, one for each input line. If the duration of the input unit is T, the duration of each slot is T/n and the duration of each frame is T (unless a frame carries some other information, as we will see shortly).

The data rate of the output link must be n times the data rate of a connection to guarantee the flow of data. In Figure 6.13, the data rate of the link is 3 times the data rate of a connection; likewise, the duration of a unit on a connection is 3 times that of the time slot (duration of a unit on the link). In the figure we represent the data prior to multiplexing as 3 times the size of the data after multiplexing. This is just to convey the idea that each unit is 3 times longer in duration before multiplexing than after.

In synchronous TDM, the data rate of the link is n times faster, and the unit duration is n times shorter.

Time slots are grouped into frames. A frame consists of one complete cycle of time slots, with one slot dedicated to each sending device. In a system with n input lines, each frame has n slots, with each slot allocated to carrying data from a specific input line.

Example 6.5

In Figure 6.13, the data rate for each input connection is 3 kbps. If 1 bit at a time is multiplexed (a unit is 1 bit), what is the duration of (a) each input slot, (b) each output slot, and (c) each frame?

Solution

We can answer the questions as follows:

a. The data rate of each input connection is 1 kbps. This means that the bit duration is 1/1000 s or 1 ms. The duration of the input time slot is 1 ms (same as bit duration).

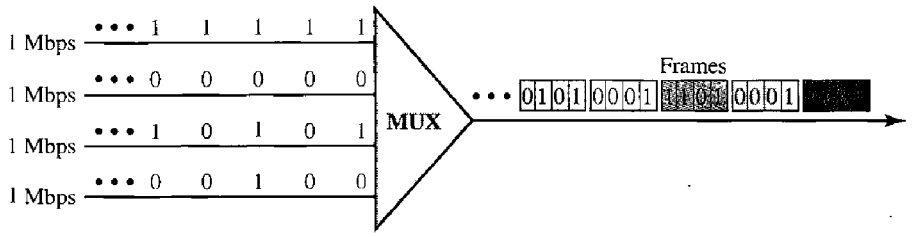
b. The duration of each output time slot is one-third of the input time slot. This means that the duration of the output time slot is 1/3 ms.

c. Each frame carries three output time slots. So the duration of a frame is $3 \times 1/3$ ms, or 1 ms. The duration of a frame is the same as the duration of an input unit.

Example 6.6

Figure 6.14 shows synchronous TDM with a data stream for each input and one data stream for the output. The unit of data is 1 bit. Find (a) the input bit duration, (b) the output bit duration, (c) the output bit rate, and (d) the output frame rate.

Figure 6.14 *Example 6.6*



Solution

We can answer the questions as follows:

a. The input bit duration is the inverse of the bit rate: 1/1 Mbps = 1 μs.
b. The output bit duration is one-fourth of the input bit duration, or 1/4 μs.
c. The output bit rate is the inverse of the output bit duration or 1/4 μs, or 4 Mbps. This can also be deduced from the fact that the output rate is 4 times as fast as any input rate; so the output rate = 4 × 1 Mbps = 4 Mbps.
d. The frame rate is always the same as any input rate. So the frame rate is 1,000,000 frames per second. Because we are sending 4 bits in each frame, we can verify the result of the previous question by multiplying the frame rate by the number of bits per frame.

Example 6.7

Four 1-kbps connections are multiplexed together. A unit is 1 bit. Find (a) the duration of 1 bit before multiplexing, (b) the transmission rate of the link, (c) the duration of a time slot, and (d) the duration of a frame.

Solution

We can answer the questions as follows:

a. The duration of 1 bit before multiplexing is 1/1 kbps, or 0.001 s (1 ms).
b. The rate of the link is 4 times the rate of a connection, or 4 kbps.
c. The duration of each time slot is one-fourth of the duration of each bit before multiplexing, or 1/4 ms or 250 μs. Note that we can also calculate this from the data rate of the link, 4 kbps. The bit duration is the inverse of the data rate, or 1/4 kbps or 250 μs.
d. The duration of a frame is always the same as the duration of a unit before multiplexing, or 1 ms. We can also calculate this in another way. Each frame in this case has four time slots. So the duration of a frame is 4 times 250 μs, or 1 ms.

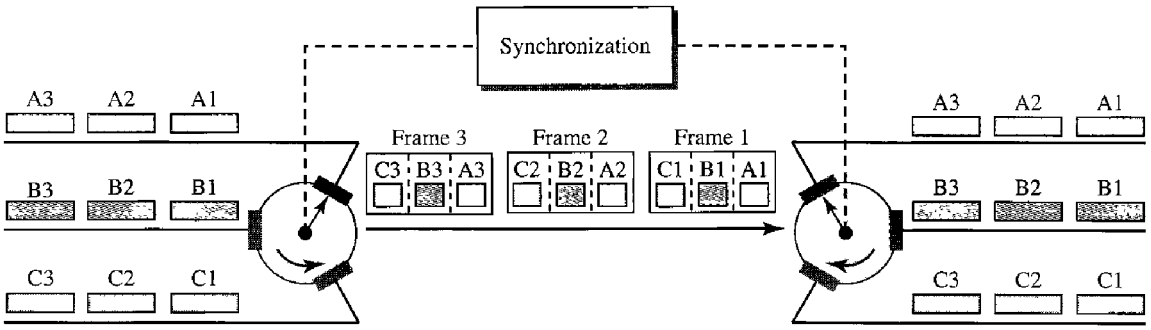
Interleaving

TDM can be visualized as two fast-rotating switches, one on the multiplexing side and the other on the demultiplexing side. The switches are synchronized and rotate at the same speed, but in opposite directions. On the multiplexing side, as the switch opens

in front of a connection, that connection has the opportunity to send a unit onto the path. This process is called **interleaving.** On the demultiplexing side, as the switch opens in front of a connection, that connection has the opportunity to receive a unit from the path.

Figure 6.15 shows the interleaving process for the connection shown in Figure 6.13. In this figure, we assume that no switching is involved and that the data from the first connection at the multiplexer site go to the first connection at the demultiplexer. We discuss switching in Chapter 8.

Figure 6.15 *Interleaving*



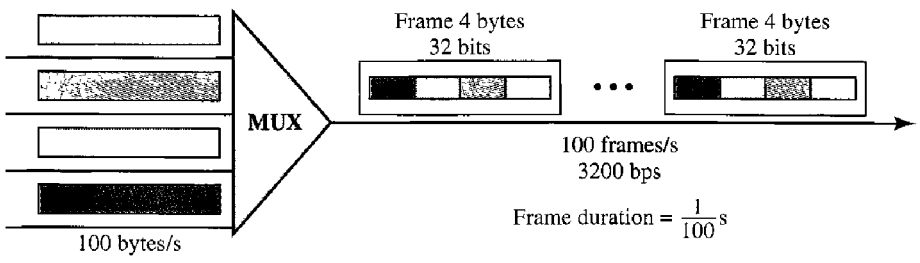
Example 6.8

Four channels are multiplexed using TDM. If each channel sends 100 bytes/s and we multiplex 1 byte per channel, show the frame traveling on the link, the size of the frame, the duration of a frame, the frame rate, and the bit rate for the link.

Solution

The multiplexer is shown in Figure 6.16. Each frame carries 1 byte from each channel; the size of each frame, therefore, is 4 bytes, or 32 bits. Because each channel is sending 100 bytes/s and a frame carries 1 byte from each channel, the frame rate must be 100 frames per second. The duration of a frame is therefore 1/100 s. The link is carrying 100 frames per second, and since each frame contains 32 bits, the bit rate is 100×32, or 3200 bps. This is actually 4 times the bit rate of each channel, which is $100 \times 8 = 800$ bps.

Figure 6.16 *Example 6.8*



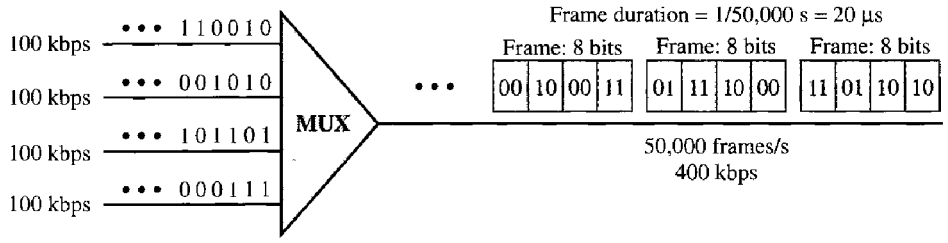
Example 6.9

A multiplexer combines four 100-kbps channels using a time slot of 2 bits. Show the output with four arbitrary inputs. What is the frame rate? What is the frame duration? What is the bit rate? What is the bit duration?

Solution

Figure 6.17 shows the output for four arbitrary inputs. The link carries 50,000 frames per second since each frame contains 2 bits per channel. The frame duration is therefore 1/50,000 s or 20 μs. The frame rate is 50,000 frames per second, and each frame carries 8 bits; the bit rate is 50,000 × 8 = 400,000 bits or 400 kbps. The bit duration is 1/400,000 s, or 2.5 μs. Note that the frame duration is 8 times the bit duration because each frame is carrying 8 bits.

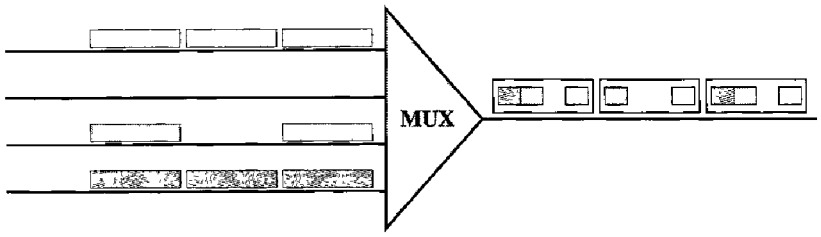
Figure 6.17 *Example 6.9*



Empty Slots

Synchronous TDM is not as efficient as it could be. If a source does not have data to send, the corresponding slot in the output frame is empty. Figure 6.18 shows a case in which one of the input lines has no data to send and one slot in another input line has discontinuous data.

Figure 6.18 *Empty slots*



The first output frame has three slots filled, the second frame has two slots filled, and the third frame has three slots filled. No frame is full. We learn in the next section that statistical TDM can improve the efficiency by removing the empty slots from the frame.

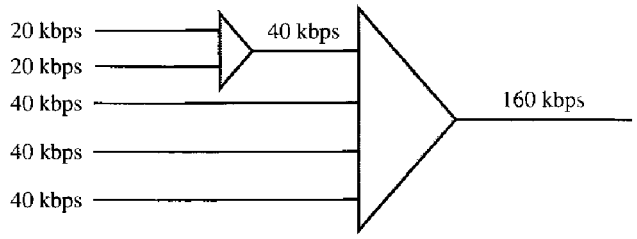
Data Rate Management

One problem with TDM is how to handle a disparity in the input data rates. In all our discussion so far, we assumed that the data rates of all input lines were the same. However,

if data rates are not the same, three strategies, or a combination of them, can be used. We call these three strategies **multilevel multiplexing, multiple-slot allocation,** and **pulse stuffing.**

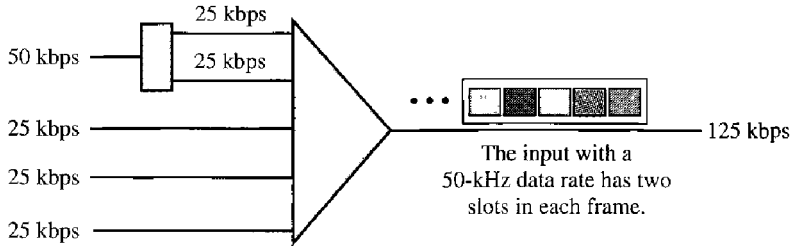
Multilevel Multiplexing Multilevel multiplexing is a technique used when the data rate of an input line is a multiple of others. For example, in Figure 6.19, we have two inputs of 20 kbps and three inputs of 40 kbps. The first two input lines can be multiplexed together to provide a data rate equal to the last three. A second level of multiplexing can create an output of 160 kbps.

Figure 6.19 *Multilevel multiplexing*



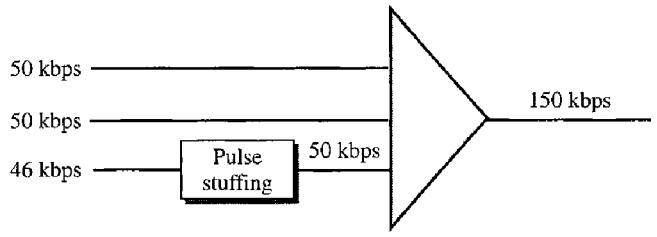
Multiple-Slot Allocation Sometimes it is more efficient to allot more than one slot in a frame to a single input line. For example, we might have an input line that has a data rate that is a multiple of another input. In Figure 6.20, the input line with a 50-kbps data rate can be given two slots in the output. We insert a serial-to-parallel converter in the line to make two inputs out of one.

Figure 6.20 *Multiple-slot multiplexing*



Pulse Stuffing Sometimes the bit rates of sources are not multiple integers of each other. Therefore, neither of the above two techniques can be applied. One solution is to make the highest input data rate the dominant data rate and then add dummy bits to the input lines with lower rates. This will increase their rates. This technique is called pulse stuffing, bit padding, or bit stuffing. The idea is shown in Figure 6.21. The input with a data rate of 46 is pulse-stuffed to increase the rate to 50 kbps. Now multiplexing can take place.

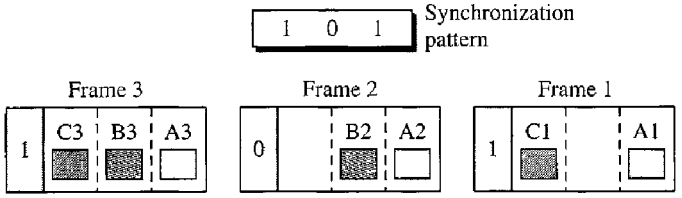
Figure 6.21 *Pulse stuffing*



Frame Synchronizing

The implementation of TDM is not as simple as that of FDM. Synchronization between the multiplexer and demultiplexer is a major issue. If the multiplexer and the demultiplexer are not synchronized, a bit belonging to one channel may be received by the wrong channel. For this reason, one or more synchronization bits are usually added to the beginning of each frame. These bits, called **framing bits,** follow a pattern, frame to frame, that allows the demultiplexer to synchronize with the incoming stream so that it can separate the time slots accurately. In most cases, this synchronization information consists of 1 bit per frame, alternating between 0 and 1, as shown in Figure 6.22.

Figure 6.22 *Framing bits*



Example 6.10

We have four sources, each creating 250 characters per second. If the interleaved unit is a character and 1 synchronizing bit is added to each frame, find (a) the data rate of each source, (b) the duration of each character in each source, (c) the frame rate, (d) the duration of each frame, (e) the number of bits in each frame, and (f) the data rate of the link.

Solution

We can answer the questions as follows:

a. The data rate of each source is $250 \times 8 = 2000$ bps = 2 kbps.

b. Each source sends 250 characters per second; therefore, the duration of a character is 1/250 s, or 4 ms.

c. Each frame has one character from each source, which means the link needs to send 250 frames per second to keep the transmission rate of each source.

d. The duration of each frame is 1/250 s, or 4 ms. Note that the duration of each frame is the same as the duration of each character coming from each source.

e. Each frame carries 4 characters and 1 extra synchronizing bit. This means that each frame is $4 \times 8 + 1 = 33$ bits.

f. The link sends 250 frames per second, and each frame contains 33 bits. This means that the data rate of the link is 250 × 33, or 8250 bps. Note that the bit rate of the link is greater than the combined bit rates of the four channels. If we add the bit rates of four channels, we get 8000 bps. Because 250 frames are traveling per second and each contains 1 extra bit for synchronizing, we need to add 250 to the sum to get 8250 bps.

Example 6.11

Two channels, one with a bit rate of 100 kbps and another with a bit rate of 200 kbps, are to be multiplexed. How this can be achieved? What is the frame rate? What is the frame duration? What is the bit rate of the link?

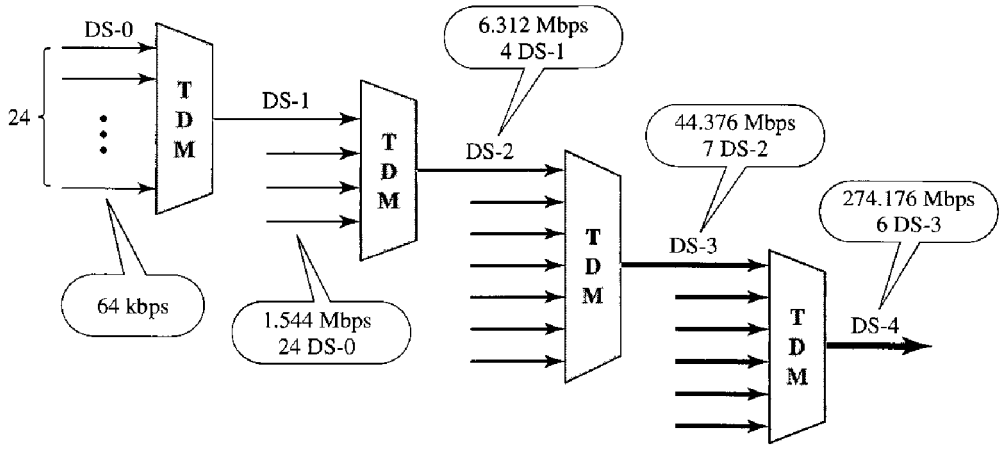
Solution

We can allocate one slot to the first channel and two slots to the second channel. Each frame carries 3 bits. The frame rate is 100,000 frames per second because it carries 1 bit from the first channel. The frame duration is 1/100,000 s, or 10 ms. The bit rate is 100,000 frames/s × 3 bits per frame, or 300 kbps. Note that because each frame carries 1 bit from the first channel, the bit rate for the first channel is preserved. The bit rate for the second channel is also preserved because each frame carries 2 bits from the second channel.

Digital Signal Service

Telephone companies implement TDM through a hierarchy of digital signals, called **digital signal (DS) service** or **digital hierarchy.** Figure 6.23 shows the data rates supported by each level.

Figure 6.23 *Digital hierarchy*



- A **DS-0** service is a single digital channel of 64 kbps.
- **DS-1** is a 1.544-Mbps service; 1.544 Mbps is 24 times 64 kbps plus 8 kbps of overhead. It can be used as a single service for 1.544-Mbps transmissions, or it can be used to multiplex 24 DS-0 channels or to carry any other combination desired by the user that can fit within its 1.544-Mbps capacity.
- **DS-2** is a 6.312-Mbps service; 6.312 Mbps is 96 times 64 kbps plus 168 kbps of overhead. It can be used as a single service for 6.312-Mbps transmissions; or it can

be used to multiplex 4 DS-1 channels, 96 DS-0 channels, or a combination of these service types.

- **DS-3** is a 44.376-Mbps service; 44.376 Mbps is 672 times 64 kbps plus 1.368 Mbps of overhead. It can be used as a single service for 44.376-Mbps transmissions; or it can be used to multiplex 7 DS-2 channels, 28 DS-1 channels, 672 DS-0 channels, or a combination of these service types.
- **DS-4** is a 274.176-Mbps service; 274.176 is 4032 times 64 kbps plus 16.128 Mbps of overhead. It can be used to multiplex 6 DS-3 channels, 42 DS-2 channels, 168 DS-1 channels, 4032 DS-0 channels, or a combination of these service types.

T Lines

DS-0, DS-1, and so on are the names of services. To implement those services, the telephone companies use **T lines** (T-1 to T-4). These are lines with capacities precisely matched to the data rates of the DS-1 to DS-4 services (see Table 6.1). So far only T-1 and T-3 lines are commercially available.

Table 6.1 *DS and T line rates*

Service	*Line*	*Rate (Mbps)*	*Voice Channels*
DS-1	T-1	1.544	24
DS-2	T-2	6.312	96
DS-3	T-3	44.736	672
DS-4	T-4	274.176	4032

The T-1 line is used to implement DS-1; T-2 is used to implement DS-2; and so on. As you can see from Table 6.1, DS-0 is not actually offered as a service, but it has been defined as a basis for reference purposes.

T Lines for Analog Transmission

T lines are digital lines designed for the transmission of digital data, audio, or video. However, they also can be used for analog transmission (regular telephone connections), provided the analog signals are first sampled, then time-division multiplexed.

The possibility of using T lines as analog carriers opened up a new generation of services for the telephone companies. Earlier, when an organization wanted 24 separate telephone lines, it needed to run 24 twisted-pair cables from the company to the central exchange. (Remember those old movies showing a busy executive with 10 telephones lined up on his desk? Or the old office telephones with a big fat cable running from them? Those cables contained a bundle of separate lines.) Today, that same organization can combine the 24 lines into one T-1 line and run only the T-1 line to the exchange. Figure 6.24 shows how 24 voice channels can be multiplexed onto one T-1 line. (Refer to Chapter 5 for PCM encoding.)

The T-1 Frame As noted above, DS-1 requires 8 kbps of overhead. To understand how this overhead is calculated, we must examine the format of a 24-voice-channel frame.

The frame used on a T-1 line is usually 193 bits divided into 24 slots of 8 bits each plus 1 extra bit for synchronization ($24 \times 8 + 1 = 193$); see Figure 6.25. In other words,

Figure 6.24 *T-1 line for multiplexing telephone lines*

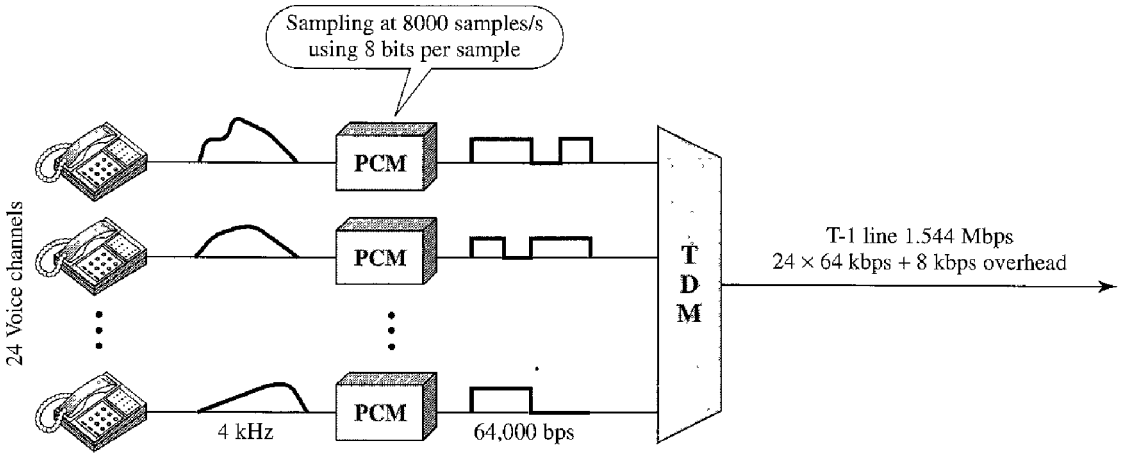
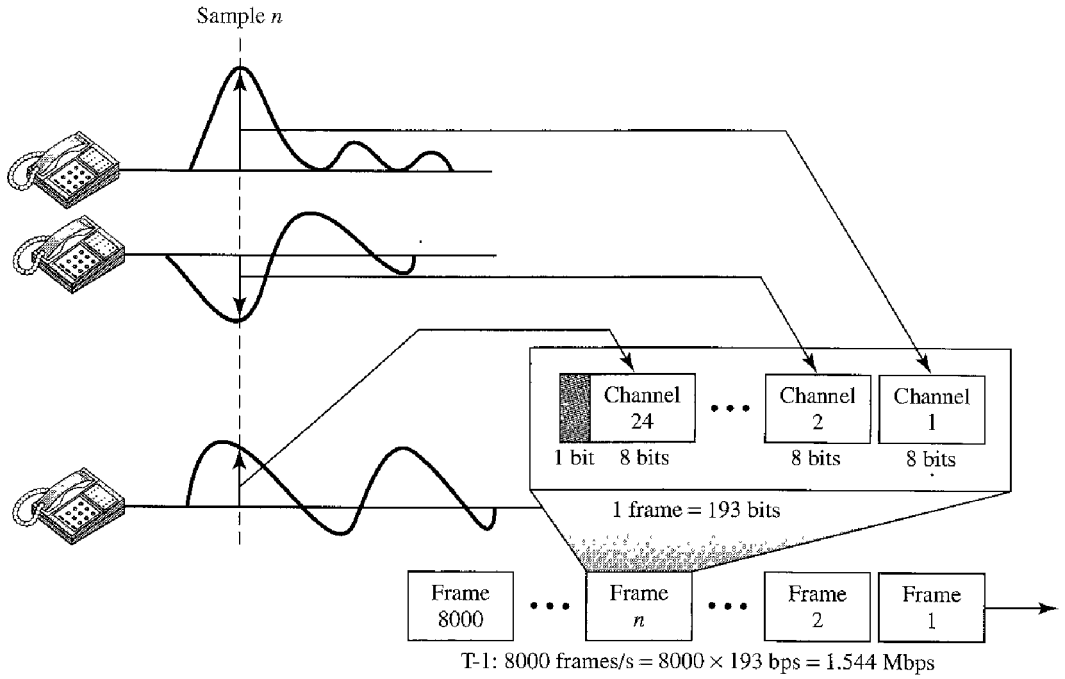


Figure 6.25 *T-1 frame structure*



each slot contains one signal segment from each channel; 24 segments are interleaved in one frame. If a T-1 line carries 8000 frames, the data rate is 1.544 Mbps (193 × 8000 = 1.544 Mbps)—the capacity of the line.

E Lines

Europeans use a version of T lines called **E lines.** The two systems are conceptually identical, but their capacities differ. Table 6.2 shows the E lines and their capacities.

Table 6.2 *E line rates*

Line	*Rate (Mbps)*	*Voice Channels*
E-1	2.048	30
E-2	8.448	120
E-3	34.368	480
E-4	139.264	1920

More Synchronous TDM Applications

Some second-generation cellular telephone companies use synchronous TDM. For example, the digital version of cellular telephony divides the available bandwidth into 30-kHz bands. For each band, TDM is applied so that six users can share the band. This means that each 30-kHz band is now made of six time slots, and the digitized voice signals of the users are inserted in the slots. Using TDM, the number of telephone users in each area is now 6 times greater. We discuss second-generation cellular telephony in Chapter 16.

Statistical Time-Division Multiplexing

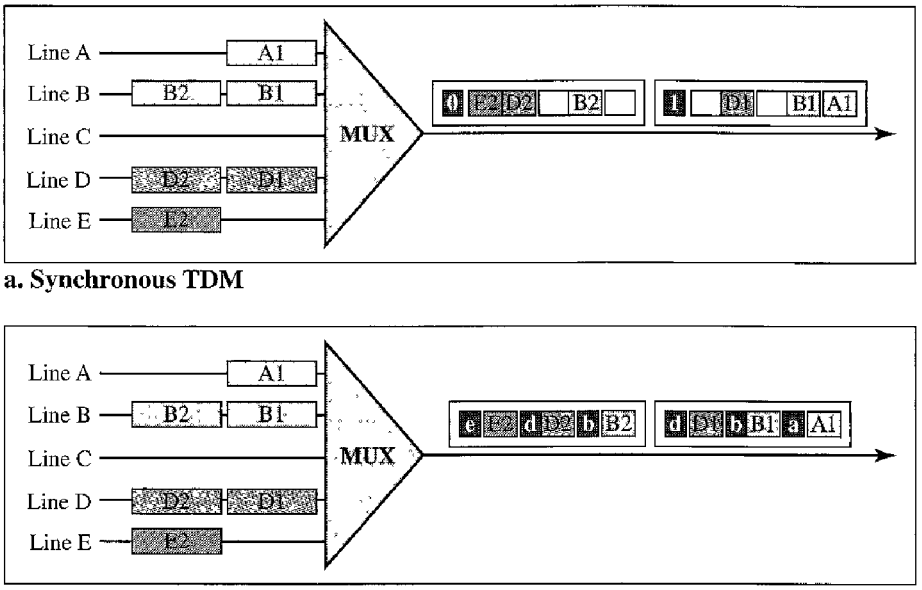
As we saw in the previous section, in synchronous TDM, each input has a reserved slot in the output frame. This can be inefficient if some input lines have no data to send. In statistical time-division multiplexing, slots are dynamically allocated to improve bandwidth efficiency. Only when an input line has a slot's worth of data to send is it given a slot in the output frame. In statistical multiplexing, the number of slots in each frame is less than the number of input lines. The multiplexer checks each input line in round-robin fashion; it allocates a slot for an input line if the line has data to send; otherwise, it skips the line and checks the next line.

Figure 6.26 shows a synchronous and a statistical TDM example. In the former, some slots are empty because the corresponding line does not have data to send. In the latter, however, no slot is left empty as long as there are data to be sent by any input line.

Addressing

Figure 6.26 also shows a major difference between slots in synchronous TDM and statistical TDM. An output slot in synchronous TDM is totally occupied by data; in statistical TDM, a slot needs to carry data as well as the address of the destination. In synchronous TDM, there is no need for addressing; synchronization and preassigned relationships between the inputs and outputs serve as an address. We know, for example, that input 1 always goes to input 2. If the multiplexer and the demultiplexer are synchronized, this is guaranteed. In statistical multiplexing, there is no fixed relationship between the inputs and outputs because there are no preassigned or reserved slots. We need to include the address of the receiver inside each slot to show where it is to be delivered. The addressing in its simplest form can be n bits to define N different output lines with $n = \log_2 N$. For example, for eight different output lines, we need a 3-bit address.

Figure 6.26 *TDM slot comparison*



a. Synchronous TDM

b. Statistical TDM

Slot Size

Since a slot carries both data and an address in statistical TDM, the ratio of the data size to address size must be reasonable to make transmission efficient. For example, it would be inefficient to send 1 bit per slot as data when the address is 3 bits. This would mean an overhead of 300 percent. In statistical TDM, a block of data is usually many bytes while the address is just a few bytes.

No Synchronization Bit

There is another difference between synchronous and statistical TDM, but this time it is at the frame level. The frames in statistical TDM need not be synchronized, so we do not need synchronization bits.

Bandwidth

In statistical TDM, the capacity of the link is normally less than the sum of the capacities of each channel. The designers of statistical TDM define the capacity of the link based on the statistics of the load for each channel. If on average only x percent of the input slots are filled, the capacity of the link reflects this. Of course, during peak times, some slots need to wait.

6.2 SPREAD SPECTRUM

Multiplexing combines signals from several sources to achieve bandwidth efficiency; the available bandwidth of a link is divided between the sources. In **spread spectrum (SS),** we also combine signals from different sources to fit into a larger bandwidth, but our goals

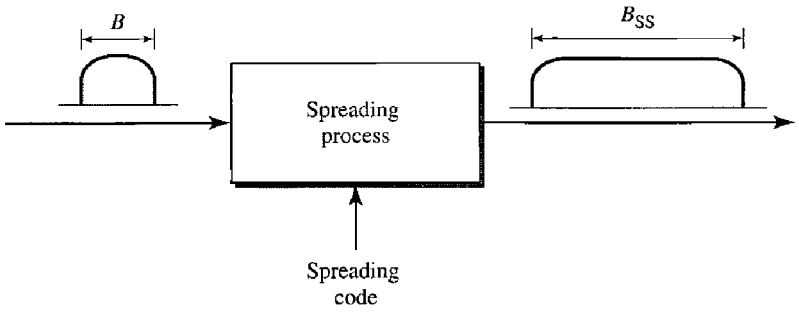
are somewhat different. Spread spectrum is designed to be used in wireless applications (LANs and WANs). In these types of applications, we have some concerns that outweigh bandwidth efficiency. In wireless applications, all stations use air (or a vacuum) as the medium for communication. Stations must be able to share this medium without interception by an eavesdropper and without being subject to jamming from a malicious intruder (in military operations, for example).

To achieve these goals, spread spectrum techniques add redundancy; they spread the original spectrum needed for each station. If the required bandwidth for each station is B, spread spectrum expands it to B_{ss}, such that $B_{ss} >> B$. The expanded bandwidth allows the source to wrap its message in a protective envelope for a more secure transmission. An analogy is the sending of a delicate, expensive gift. We can insert the gift in a special box to prevent it from being damaged during transportation, and we can use a superior delivery service to guarantee the safety of the package.

Figure 6.27 shows the idea of spread spectrum. Spread spectrum achieves its goals through two principles:

1. The bandwidth allocated to each station needs to be, by far, larger than what is needed. This allows redundancy.
2. The expanding of the original bandwidth B to the bandwidth B_{ss} must be done by a process that is independent of the original signal. In other words, the spreading process occurs after the signal is created by the source.

Figure 6.27 *Spread spectrum*



After the signal is created by the source, the spreading process uses a spreading code and spreads the bandwidth. The figure shows the original bandwidth B and the spreaded bandwidth B_{SS}. The spreading code is a series of numbers that look random, but are actually a pattern.

There are two techniques to spread the bandwidth: frequency hopping spread spectrum (FHSS) and direct sequence spread spectrum (DSSS).

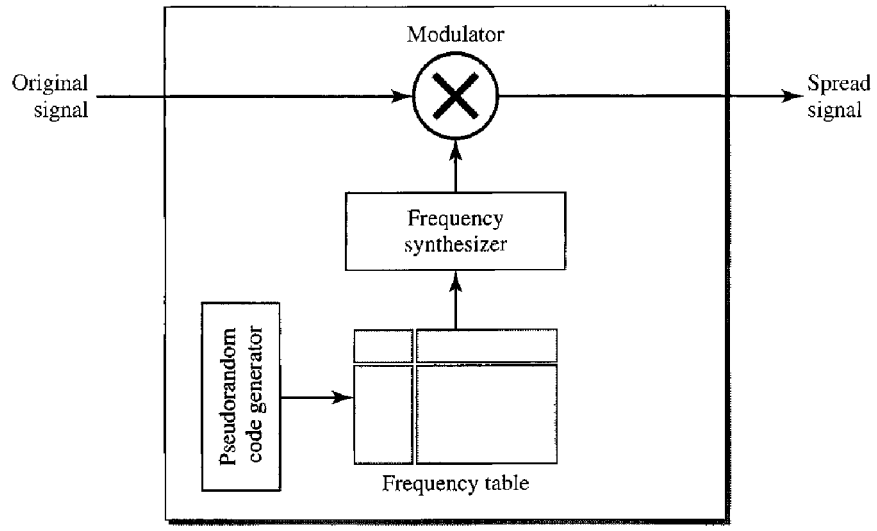
Frequency Hopping Spread Spectrum (FHSS)

The **frequency hopping spread spectrum (FHSS)** technique uses M different carrier frequencies that are modulated by the source signal. At one moment, the signal modulates one carrier frequency; at the next moment, the signal modulates another carrier

frequency. Although the modulation is done using one carrier frequency at a time, M frequencies are used in the long run. The bandwidth occupied by a source after spreading is $B_{FHSS} >> B$.

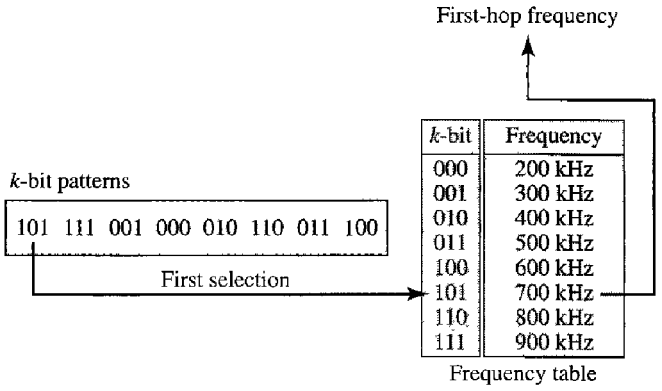
Figure 6.28 shows the general layout for FHSS. A **pseudorandom code generator,** called **pseudorandom noise (PN),** creates a k-bit pattern for every **hopping period T_h.** The frequency table uses the pattern to find the frequency to be used for this hopping period and passes it to the frequency synthesizer. The frequency synthesizer creates a carrier signal of that frequency, and the source signal modulates the carrier signal.

Figure 6.28 *Frequency hopping spread spectrum (FHSS)*



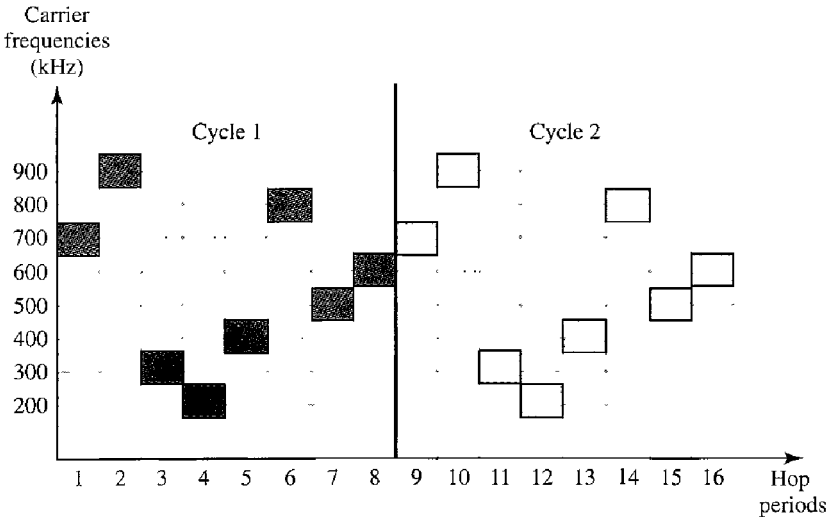
Suppose we have decided to have eight hopping frequencies. This is extremely low for real applications and is just for illustration. In this case, M is 8 and k is 3. The pseudorandom code generator will create eight different 3-bit patterns. These are mapped to eight different frequencies in the frequency table (see Figure 6.29).

Figure 6.29 *Frequency selection in FHSS*



The pattern for this station is 101, 111, 001, 000, 010, 011, 100. Note that the pattern is pseudorandom it is repeated after eight hoppings. This means that at hopping period 1, the pattern is 101. The frequency selected is 700 kHz; the source signal modulates this carrier frequency. The second k-bit pattern selected is 111, which selects the 900-kHz carrier; the eighth pattern is 100, the frequency is 600 kHz. After eight hoppings, the pattern repeats, starting from 101 again. Figure 6.30 shows how the signal hops around from carrier to carrier. We assume the required bandwidth of the original signal is 100 kHz.

Figure 6.30 *FHSS cycles*



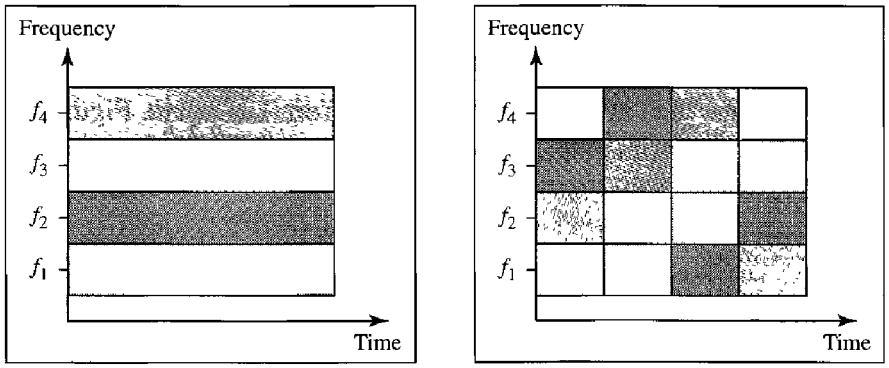
It can be shown that this scheme can accomplish the previously mentioned goals. If there are many k-bit patterns and the hopping period is short, a sender and receiver can have privacy. If an intruder tries to intercept the transmitted signal, she can only access a small piece of data because she does not know the spreading sequence to quickly adapt herself to the next hop. The scheme has also an antijamming effect. A malicious sender may be able to send noise to jam the signal for one hopping period (randomly), but not for the whole period.

Bandwidth Sharing

If the number of hopping frequencies is M, we can multiplex M channels into one by using the same B_{ss} bandwidth. This is possible because a station uses just one frequency in each hopping period; $M - 1$ other frequencies can be used by other $M - 1$ stations. In other words, M different stations can use the same B_{ss} if an appropriate modulation technique such as multiple FSK (MFSK) is used. FHSS is similar to FDM, as shown in Figure 6.31.

Figure 6.31 shows an example of four channels using FDM and four channels using FHSS. In FDM, each station uses $1/M$ of the bandwidth, but the allocation is fixed; in FHSS, each station uses $1/M$ of the bandwidth, but the allocation changes hop to hop.

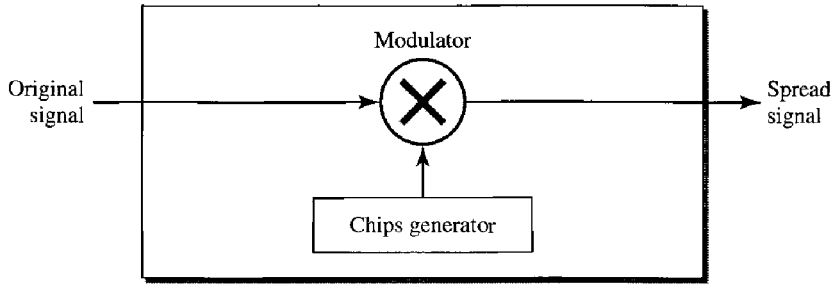
Figure 6.31 *Bandwidth sharing*



Direct Sequence Spread Spectrum

The **direct sequence spread spectrum (DSSS)** technique also expands the bandwidth of the original signal, but the process is different. In DSSS, we replace each data bit with n bits using a spreading code. In other words, each bit is assigned a code of n bits, called chips, where the chip rate is n times that of the data bit. Figure 6.32 shows the concept of DSSS.

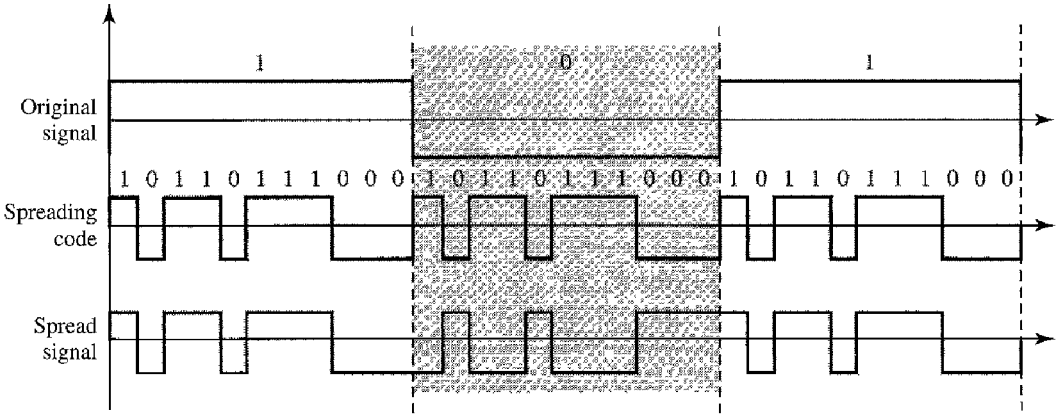
Figure 6.32 *DSSS*



As an example, let us consider the sequence used in a wireless LAN, the famous **Barker sequence** where n is 11. We assume that the original signal and the chips in the chip generator use polar NRZ encoding. Figure 6.33 shows the chips and the result of multiplying the original data by the chips to get the spread signal.

In Figure 6.33, the spreading code is 11 chips having the pattern 10110111000 (in this case). If the original signal rate is N, the rate of the spread signal is $11N$. This means that the required bandwidth for the spread signal is 11 times larger than the bandwidth of the original signal. The spread signal can provide privacy if the intruder does not know the code. It can also provide immunity against interference if each station uses a different code.

Figure 6.33 *DSSS example*



Bandwidth Sharing

Can we share a bandwidth in DSSS as we did in FHSS? The answer is no and yes. If we use a spreading code that spreads signals (from different stations) that cannot be combined and separated, we cannot share a bandwidth. For example, as we will see in Chapter 14, some wireless LANs use DSSS and the spread bandwidth cannot be shared. However, if we use a special type of sequence code that allows the combining and separating of spread signals, we can share the bandwidth. As we will see in Chapter 16, a special spreading code allows us to use DSSS in cellular telephony and share a bandwidth between several users.

6.3 RECOMMENDED READING

For more details about subjects discussed in this chapter, we recommend the following books. The items in brackets [. . .] refer to the reference list at the end of the text.

Books

Multiplexing is elegantly discussed in Chapters 19 of [Pea92]. [Cou01] gives excellent coverage of TDM and FDM in Sections 3.9 to 3.11. More advanced materials can be found in [Ber96]. Multiplexing is discussed in Chapter 8 of [Sta04]. A good coverage of spread spectrum can be found in Section 5.13 of [Cou01] and Chapter 9 of [Sta04].

6.4 KEY TERMS

analog hierarchy
Barker sequence
channel
chip
demultiplexer (DEMUX)
dense WDM (DWDM)
digital signal (DS) service
direct sequence spread spectrum (DSSS)
E line
framing bit
frequency hopping spread spectrum (FSSS)

frequency-division multiplexing (FDM)
group
guard band
hopping period
interleaving
jumbo group
link
master group
multilevel multiplexing
multiple-slot multiplexing
multiplexer (MUX)
multiplexing
pseudorandom code generator
pseudorandom noise (PN)
pulse stuffing
spread spectrum (SS)
statistical TDM
supergroup
synchronous TDM
T line
time-division multiplexing (TDM)
wavelength-division multiplexing (WDM)

6.5 SUMMARY

- Bandwidth utilization is the use of available bandwidth to achieve specific goals. Efficiency can be achieved by using multiplexing; privacy and antijamming can be achieved by using spreading.
- Multiplexing is the set of techniques that allows the simultaneous transmission of multiple signals across a single data link. In a multiplexed system, n lines share the bandwidth of one link. The word link refers to the physical path. The word channel refers to the portion of a link that carries a transmission.
- There are three basic multiplexing techniques: frequency-division multiplexing, wavelength-division multiplexing, and time-division multiplexing. The first two are techniques designed for analog signals, the third, for digital signals
- Frequency-division multiplexing (FDM) is an analog technique that can be applied when the bandwidth of a link (in hertz) is greater than the combined bandwidths of the signals to be transmitted.
- Wavelength-division multiplexing (WDM) is designed to use the high bandwidth capability of fiber-optic cable. WDM is an analog multiplexing technique to combine optical signals.
- Time-division multiplexing (TDM) is a digital process that allows several connections to share the high bandwidth of a link. TDM is a digital multiplexing technique for combining several low-rate channels into one high-rate one.
- We can divide TDM into two different schemes: synchronous or statistical. In synchronous TDM, each input connection has an allotment in the output even if it is not sending data. In statistical TDM, slots are dynamically allocated to improve bandwidth efficiency.
- In spread spectrum (SS), we combine signals from different sources to fit into a larger bandwidth. Spread spectrum is designed to be used in wireless applications in which stations must be able to share the medium without interception by an eavesdropper and without being subject to jamming from a malicious intruder.
- The frequency hopping spread spectrum (FHSS) technique uses M different carrier frequencies that are modulated by the source signal. At one moment, the signal

modulates one carrier frequency; at the next moment, the signal modulates another carrier frequency.

- The direct sequence spread spectrum (DSSS) technique expands the bandwidth of a signal by replacing each data bit with n bits using a spreading code. In other words, each bit is assigned a code of n bits, called chips.

6.6 PRACTICE SET

Review Questions

1. Describe the goals of multiplexing.
2. List three main multiplexing techniques mentioned in this chapter.
3. Distinguish between a link and a channel in multiplexing.
4. Which of the three multiplexing techniques is (are) used to combine analog signals? Which of the three multiplexing techniques is (are) used to combine digital signals?
5. Define the analog hierarchy used by telephone companies and list different levels of the hierarchy.
6. Define the digital hierarchy used by telephone companies and list different levels of the hierarchy.
7. Which of the three multiplexing techniques is common for fiber optic links? Explain the reason.
8. Distinguish between multilevel TDM, multiple slot TDM, and pulse-stuffed TDM.
9. Distinguish between synchronous and statistical TDM.
10. Define spread spectrum and its goal. List the two spread spectrum techniques discussed in this chapter.
11. Define FHSS and explain how it achieves bandwidth spreading.
12. Define DSSS and explain how it achieves bandwidth spreading.

Exercises

13. Assume that a voice channel occupies a bandwidth of 4 kHz. We need to multiplex 10 voice channels with guard bands of 500 Hz using FDM. Calculate the required bandwidth.
14. We need to transmit 100 digitized voice channels using a pass-band channel of 20 KHz. What should be the ratio of bits/Hz if we use no guard band?
15. In the analog hierarchy of Figure 6.9, find the overhead (extra bandwidth for guard band or control) in each hierarchy level (group, supergroup, master group, and jumbo group).
16. We need to use synchronous TDM and combine 20 digital sources, each of 100 Kbps. Each output slot carries 1 bit from each digital source, but one extra bit is added to each frame for synchronization. Answer the following questions:
    a. What is the size of an output frame in bits?
    b. What is the output frame rate?

c. What is the duration of an output frame?

d. What is the output data rate?

e. What is the efficiency of the system (ratio of useful bits to the total bits).

17. Repeat Exercise 16 if each output slot carries 2 bits from each source.

18. We have 14 sources, each creating 500 8-bit characters per second. Since only some of these sources are active at any moment, we use statistical TDM to combine these sources using character interleaving. Each frame carries 6 slots at a time, but we need to add four-bit addresses to each slot. Answer the following questions:

    a. What is the size of an output frame in bits?

    b. What is the output frame rate?

    c. What is the duration of an output frame?

    d. What is the output data rate?

19. Ten sources, six with a bit rate of 200 kbps and four with a bit rate of 400 kbps are to be combined using multilevel TDM with no synchronizing bits. Answer the following questions about the final stage of the multiplexing:

    a. What is the size of a frame in bits?

    b. What is the frame rate?

    c. What is the duration of a frame?

    d. What is the data rate?

20. Four channels, two with a bit rate of 200 kbps and two with a bit rate of 150 kbps, are to be multiplexed using multiple slot TDM with no synchronization bits. Answer the following questions:

    a. What is the size of a frame in bits?

    b. What is the frame rate?

    c. What is the duration of a frame?

    d. What is the data rate?

21. Two channels, one with a bit rate of 190 kbps and another with a bit rate of 180 kbps, are to be multiplexed using pulse stuffing TDM with no synchronization bits. Answer the following questions:

    a. What is the size of a frame in bits?

    b. What is the frame rate?

    c. What is the duration of a frame?

    d. What is the data rate?

22. Answer the following questions about a T-1 line:

    a. What is the duration of a frame?

    b. What is the overhead (number of extra bits per second)?

23. Show the contents of the five output frames for a synchronous TDM multiplexer that combines four sources sending the following characters. Note that the characters are sent in the same order that they are typed. The third source is silent.

    a. Source 1 message: HELLO

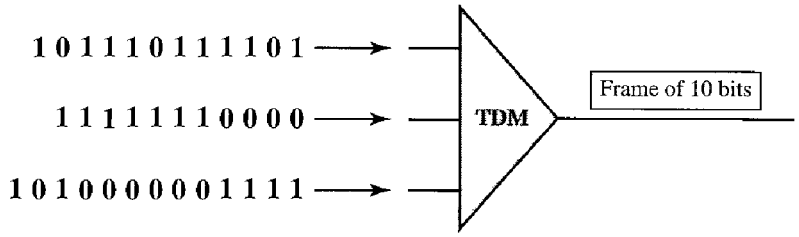
    b. Source 2 message: HI

c. Source 3 message:

d. Source 4 message: BYE

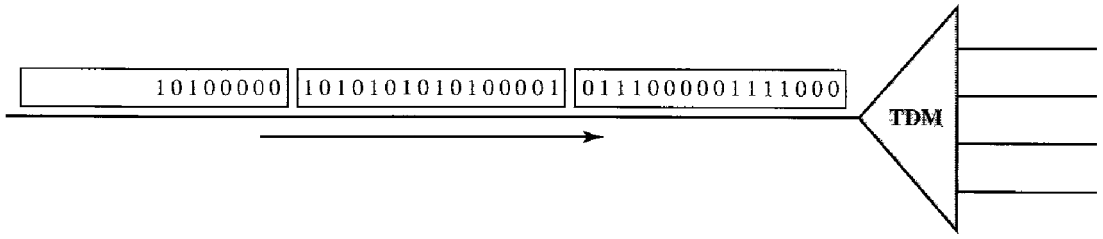
24. Figure 6.34 shows a multiplexer in a synchronous TDM system. Each output slot is only 10 bits long (3 bits taken from each input plus 1 framing bit). What is the output stream? The bits arrive at the multiplexer as shown by the arrows.

Figure 6.34 *Exercise 24*



25. Figure 6.35 shows a demultiplexer in a synchronous TDM. If the input slot is 16 bits long (no framing bits), what is the bit stream in each output? The bits arrive at the demultiplexer as shown by the arrows.

Figure 6.35 *Exercise 25*



26. Answer the following questions about the digital hierarchy in Figure 6.23:
    a. What is the overhead (number of extra bits) in the DS-1 service?
    b. What is the overhead (number of extra bits) in the DS-2 service?
    c. What is the overhead (number of extra bits) in the DS-3 service?
    d. What is the overhead (number of extra bits) in the DS-4 service?
27. What is the minimum number of bits in a PN sequence if we use FHSS with a channel bandwidth of B = 4 KHz and B_{SS} = 100 KHz?
28. An FHSS system uses a 4-bit PN sequence. If the bit rate of the PN is 64 bits per second, answer the following questions:
    a. What is the total number of possible hops?
    b. What is the time needed to finish a complete cycle of PN?

29. A pseudorandom number generator uses the following formula to create a random series:

$$N_{i+1} = (5 + 7N_i) \bmod 17 - 1$$

In which N_i defines the current random number and N_{i+1} defines the next random number. The term *mod* means the value of the remainder when dividing $(5 + 7N_i)$ by 17.

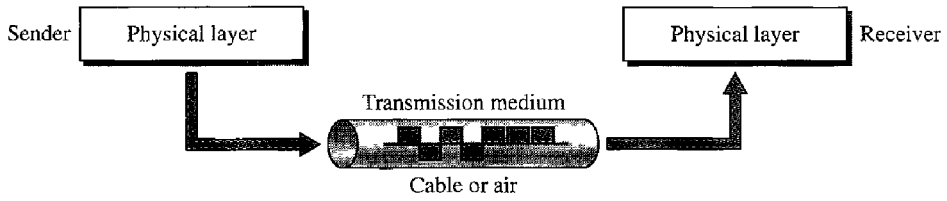
30. We have a digital medium with a data rate of 10 Mbps. How many 64-kbps voice channels can be carried by this medium if we use DSSS with the Barker sequence?

CHAPTER 7

Transmission Media

We discussed many issues related to the physical layer in Chapters 3 through 6. In this chapter, we discuss transmission media. Transmission media are actually located below the physical layer and are directly controlled by the physical layer. You could say that transmission media belong to layer zero. Figure 7.1 shows the position of transmission media in relation to the physical layer.

Figure 7.1 *Transmission medium and physical layer*



A **transmission medium** can be broadly defined as anything that can carry information from a source to a destination. For example, the transmission medium for two people having a dinner conversation is the air. The air can also be used to convey the message in a smoke signal or semaphore. For a written message, the transmission medium might be a mail carrier, a truck, or an airplane.

In data communications the definition of the information and the transmission medium is more specific. The transmission medium is usually free space, metallic cable, or fiber-optic cable. The information is usually a signal that is the result of a conversion of data from another form.

The use of long-distance communication using electric signals started with the invention of the telegraph by Morse in the 19th century. Communication by telegraph was slow and dependent on a metallic medium.

Extending the range of the human voice became possible when the telephone was invented in 1869. Telephone communication at that time also needed a metallic medium to carry the electric signals that were the result of a conversion from the human voice.

The communication was, however, unreliable due to the poor quality of the wires. The lines were often noisy and the technology was unsophisticated.

Wireless communication started in 1895 when Hertz was able to send high-frequency signals. Later, Marconi devised a method to send telegraph-type messages over the Atlantic Ocean.

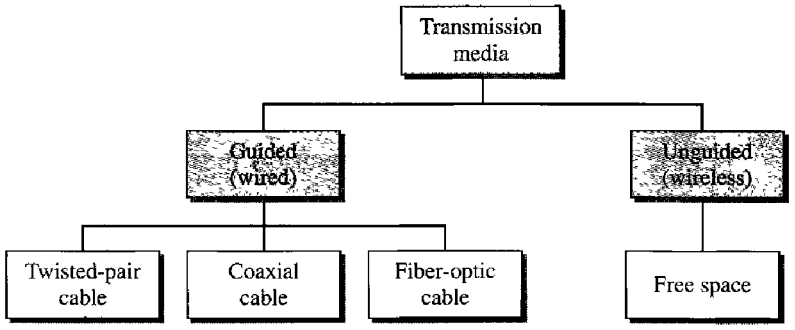
We have come a long way. Better metallic media have been invented (twisted-pair and coaxial cables, for example). The use of optical fibers has increased the data rate incredibly. Free space (air, vacuum, and water) is used more efficiently, in part due to the technologies (such as modulation and multiplexing) discussed in the previous chapters.

As discussed in Chapter 3, computers and other telecommunication devices use signals to represent data. These signals are transmitted from one device to another in the form of electromagnetic energy, which is propagated through transmission media.

Electromagnetic energy, a combination of electric and magnetic fields vibrating in relation to each other, includes power, radio waves, infrared light, visible light, ultraviolet light, and X, gamma, and cosmic rays. Each of these constitutes a portion of the **electromagnetic spectrum.** Not all portions of the spectrum are currently usable for telecommunications, however. The media to harness those that are usable are also limited to a few types.

In telecommunications, transmission media can be divided into two broad categories: guided and unguided. Guided media include twisted-pair cable, coaxial cable, and fiber-optic cable. Unguided medium is free space. Figure 7.2 shows this taxonomy.

Figure 7.2 *Classes of transmission media*



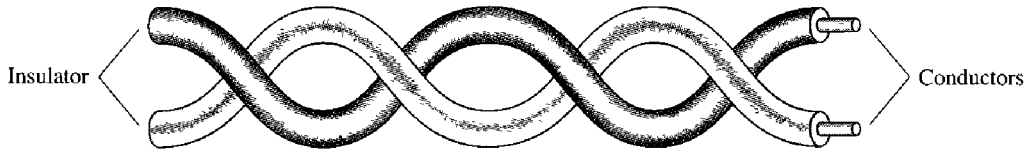
7.1 GUIDED MEDIA

Guided media, which are those that provide a conduit from one device to another, include **twisted-pair cable, coaxial cable,** and **fiber-optic cable.** A signal traveling along any of these media is directed and contained by the physical limits of the medium. Twisted-pair and coaxial cable use metallic (copper) conductors that accept and transport signals in the form of electric current. **Optical fiber** is a cable that accepts and transports signals in the form of light.

Twisted-Pair Cable

A twisted pair consists of two conductors (normally copper), each with its own plastic insulation, twisted together, as shown in Figure 7.3.

Figure 7.3 *Twisted-pair cable*



One of the wires is used to carry signals to the receiver, and the other is used only as a ground reference. The receiver uses the difference between the two.

In addition to the signal sent by the sender on one of the wires, interference (noise) and crosstalk may affect both wires and create unwanted signals.

If the two wires are parallel, the effect of these unwanted signals is not the same in both wires because they are at different locations relative to the noise or crosstalk sources (e.g., one is closer and the other is farther). This results in a difference at the receiver. By twisting the pairs, a balance is maintained. For example, suppose in one twist, one wire is closer to the noise source and the other is farther; in the next twist, the reverse is true. Twisting makes it probable that both wires are equally affected by external influences (noise or crosstalk). This means that the receiver, which calculates the difference between the two, receives no unwanted signals. The unwanted signals are mostly canceled out. From the above discussion, it is clear that the number of twists per unit of length (e.g., inch) has some effect on the quality of the cable.

Unshielded Versus Shielded Twisted-Pair Cable

The most common twisted-pair cable used in communications is referred to as **unshielded twisted-pair (UTP).** IBM has also produced a version of twisted-pair cable for its use called **shielded twisted-pair (STP).** STP cable has a metal foil or braided-mesh covering that encases each pair of insulated conductors. Although metal casing improves the quality of cable by preventing the penetration of noise or crosstalk, it is bulkier and more expensive. Figure 7.4 shows the difference between UTP and STP. Our discussion focuses primarily on UTP because STP is seldom used outside of IBM.

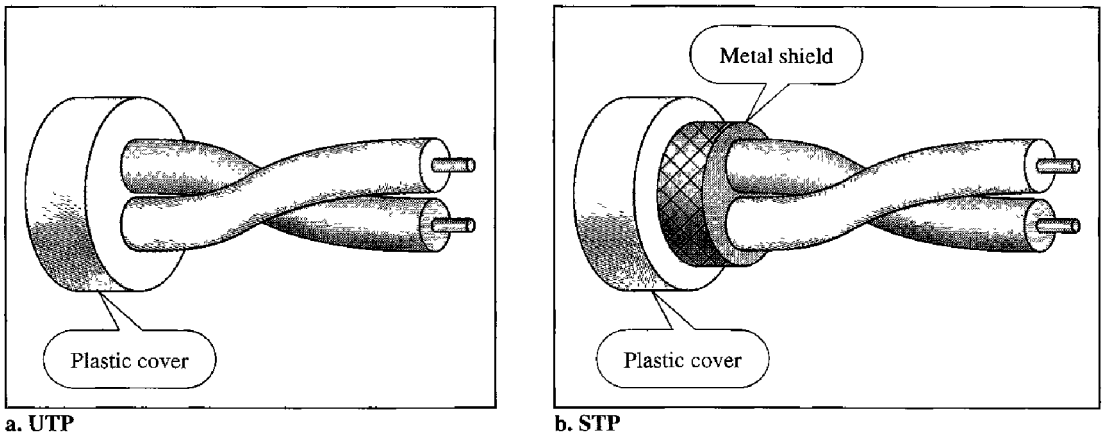
Categories

The Electronic Industries Association (EIA) has developed standards to classify unshielded twisted-pair cable into seven categories. Categories are determined by cable quality, with 1 as the lowest and 7 as the highest. Each EIA category is suitable for specific uses. Table 7.1 shows these categories.

Connectors

The most common UTP connector is **RJ45** (RJ stands for registered jack), as shown in Figure 7.5. The RJ45 is a keyed connector, meaning the connector can be inserted in only one way.

Figure 7.4 *UTP and STP cables*



a. UTP b. STP

Table 7.1 *Categories of unshielded twisted-pair cables*

Category	*Specification*	*Data Rate (Mbps)*	*Use*
1	Unshielded twisted-pair used in telephone	< 0.1	Telephone
2	Unshielded twisted-pair originally used in T-lines	2	T-1 lines
3	Improved CAT 2 used in LANs	10	LANs
4	Improved CAT 3 used in Token Ring networks	20	LANs
5	Cable wire is normally 24 AWG with a jacket and outside sheath	100	LANs
5E	An extension to category 5 that includes extra features to minimize the crosstalk and electromagnetic interference	125	LANs
6	A new category with matched components coming from the same manufacturer. The cable must be tested at a 200-Mbps data rate.	200	LANs
7	Sometimes called SSTP (shielded screen twisted-pair). Each pair is individually wrapped in a helical metallic foil followed by a metallic foil shield in addition to the outside sheath. The shield decreases the effect of crosstalk and increases the data rate.	600	LANs

Performance

One way to measure the performance of twisted-pair cable is to compare attenuation versus frequency and distance. A twisted-pair cable can pass a wide range of frequencies. However, Figure 7.6 shows that with increasing frequency, the attenuation, measured in decibels per kilometer (dB/km), sharply increases with frequencies above 100 kHz. Note that ***gauge*** is a measure of the thickness of the wire.

Figure 7.5 *UTP connector*

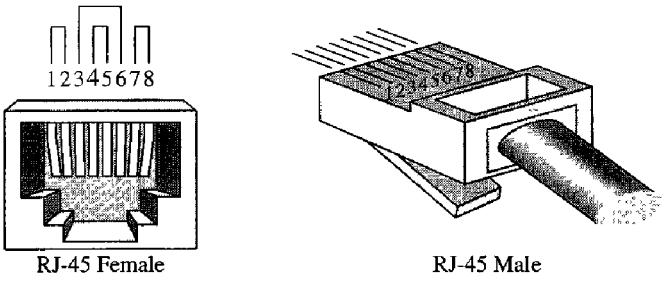
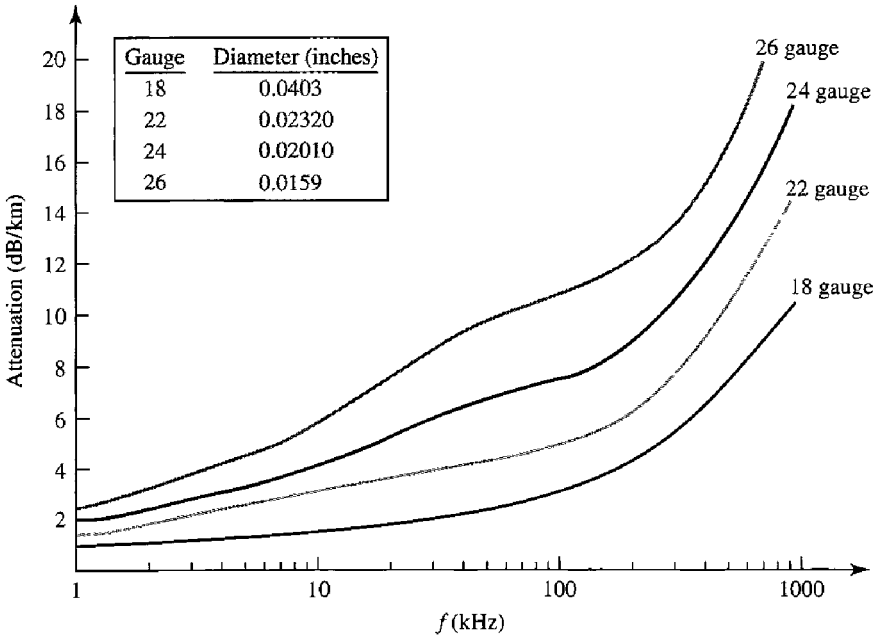


Figure 7.6 *UTP performance*



Applications

Twisted-pair cables are used in telephone lines to provide voice and data channels. The local loop—the line that connects subscribers to the central telephone office—commonly consists of unshielded twisted-pair cables. We discuss telephone networks in Chapter 9.

The DSL lines that are used by the telephone companies to provide high-data-rate connections also use the high-bandwidth capability of unshielded twisted-pair cables. We discuss DSL technology in Chapter 9.

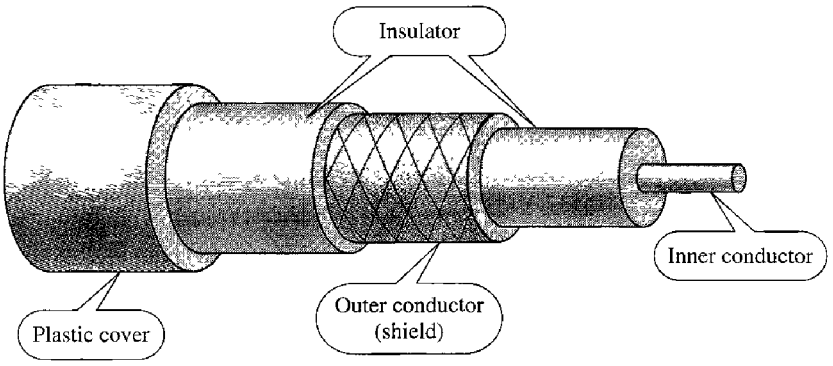
Local-area networks, such as 10Base-T and 100Base-T, also use twisted-pair cables. We discuss these networks in Chapter 13.

Coaxial Cable

Coaxial cable (or *coax*) carries signals of higher frequency ranges than those in twisted-pair cable, in part because the two media are constructed quite differently. Instead of

having two wires, coax has a central core conductor of solid or stranded wire (usually copper) enclosed in an insulating sheath, which is, in turn, encased in an outer conductor of metal foil, braid, or a combination of the two. The outer metallic wrapping serves both as a shield against noise and as the second conductor, which completes the circuit. This outer conductor is also enclosed in an insulating sheath, and the whole cable is protected by a plastic cover (see Figure 7.7).

Figure 7.7 *Coaxial cable*



Coaxial Cable Standards

Coaxial cables are categorized by their **radio government (RG)** ratings. Each RG number denotes a unique set of physical specifications, including the wire gauge of the inner conductor, the thickness and type of the inner insulator, the construction of the shield, and the size and type of the outer casing. Each cable defined by an RG rating is adapted for a specialized function, as shown in Table 7.2.

Table 7.2 *Categories of coaxial cables*

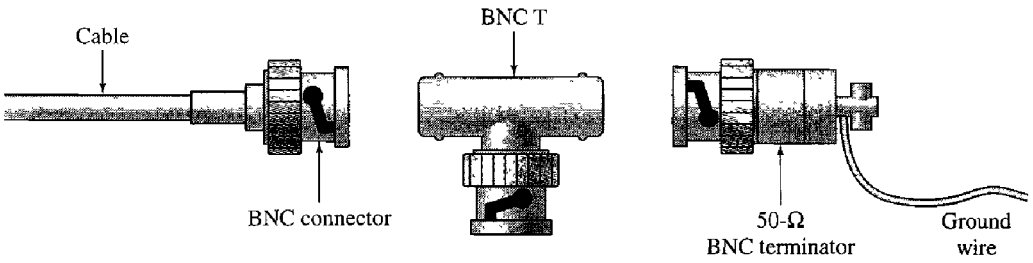
Category	*Impedance*	*Use*
RG-59	75 Ω	Cable TV
RG-58	50 Ω	Thin Ethernet
RG-11	50 Ω	Thick Ethernet

Coaxial Cable Connectors

To connect coaxial cable to devices, we need coaxial connectors. The most common type of connector used today is the **Bayone-Neill-Concelman (BNC),** connector. Figure 7.8 shows three popular types of these connectors: the BNC connector, the BNC T connector, and the BNC terminator.

The BNC connector is used to connect the end of the cable to a device, such as a TV set. The BNC T connector is used in Ethernet networks (see Chapter 13) to branch out to a connection to a computer or other device. The BNC terminator is used at the end of the cable to prevent the reflection of the signal.

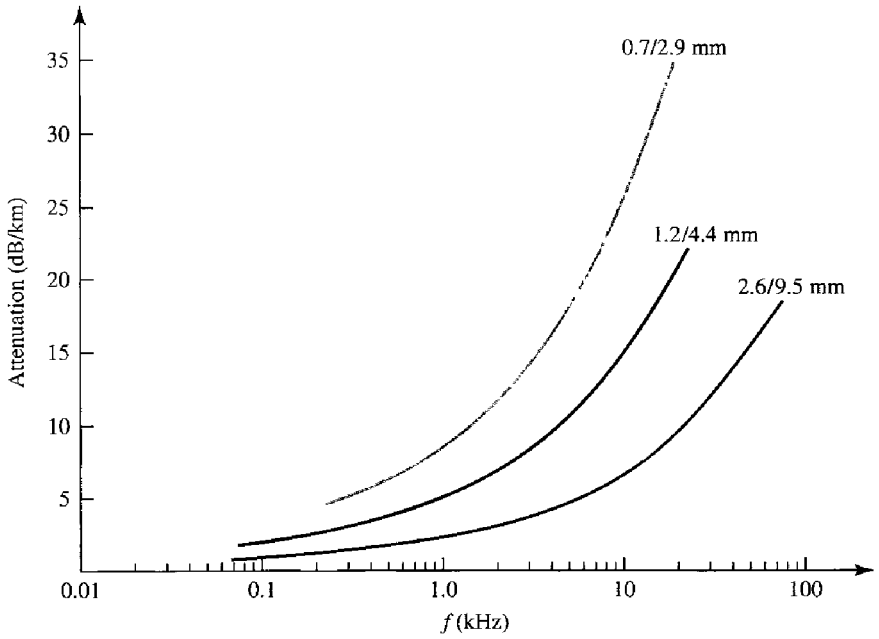
Figure 7.8 *BNC connectors*



Performance

As we did with twisted-pair cables, we can measure the performance of a coaxial cable. We notice in Figure 7.9 that the attenuation is much higher in coaxial cables than in twisted-pair cable. In other words, although coaxial cable has a much higher bandwidth, the signal weakens rapidly and requires the frequent use of repeaters.

Figure 7.9 *Coaxial cable performance*



Applications

Coaxial cable was widely used in analog telephone networks where a single coaxial network could carry 10,000 voice signals. Later it was used in digital telephone networks where a single coaxial cable could carry digital data up to 600 Mbps. However, coaxial cable in telephone networks has largely been replaced today with fiber-optic cable.

Cable TV networks (see Chapter 9) also use coaxial cables. In the traditional cable TV network, the entire network used coaxial cable. Later, however, cable TV providers

replaced most of the media with fiber-optic cable; hybrid networks use coaxial cable only at the network boundaries, near the consumer premises. Cable TV uses RG-59 coaxial cable.

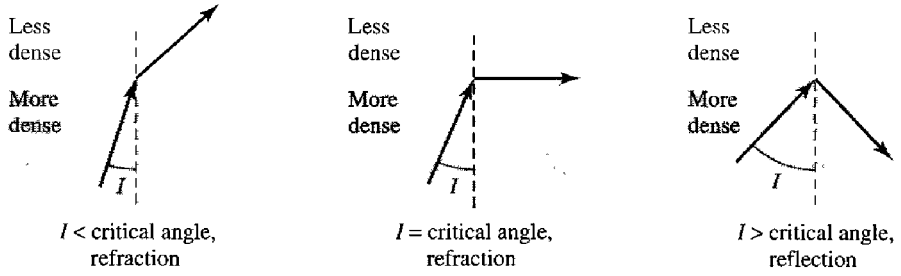
Another common application of coaxial cable is in traditional Ethernet LANs (see Chapter 13). Because of its high bandwidth, and consequently high data rate, coaxial cable was chosen for digital transmission in early Ethernet LANs. The 10Base-2, or Thin Ethernet, uses RG-58 coaxial cable with BNC connectors to transmit data at 10 Mbps with a range of 185 m. The 10Base5, or Thick Ethernet, uses RG-11 (thick coaxial cable) to transmit 10 Mbps with a range of 5000 m. Thick Ethernet has specialized connectors.

Fiber-Optic Cable

A fiber-optic cable is made of glass or plastic and transmits signals in the form of light. To understand optical fiber, we first need to explore several aspects of the nature of light.

Light travels in a straight line as long as it is moving through a single uniform substance. If a ray of light traveling through one substance suddenly enters another substance (of a different density), the ray changes direction. Figure 7.10 shows how a ray of light changes direction when going from a more dense to a less dense substance.

Figure 7.10 *Bending of light ray*



As the figure shows, if the **angle of incidence** *I* (the angle the ray makes with the line perpendicular to the interface between the two substances) is less than the **critical angle,** the ray **refracts** and moves closer to the surface. If the angle of incidence is equal to the critical angle, the light bends along the interface. If the angle is greater than the critical angle, the ray **reflects** (makes a turn) and travels again in the denser substance. Note that the critical angle is a property of the substance, and its value differs from one substance to another.

Optical fibers use reflection to guide light through a channel. A glass or plastic **core** is surrounded by a **cladding** of less dense glass or plastic. The difference in density of the two materials must be such that a beam of light moving through the core is reflected off the cladding instead of being refracted into it. See Figure 7.11.

Propagation Modes

Current technology supports two modes (multimode and single mode) for propagating light along optical channels, each requiring fiber with different physical characteristics. Multimode can be implemented in two forms: step-index or graded-index (see Figure 7.12).

Figure 7.11 *Optical fiber*

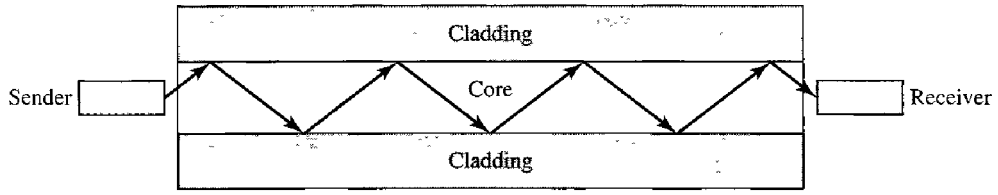
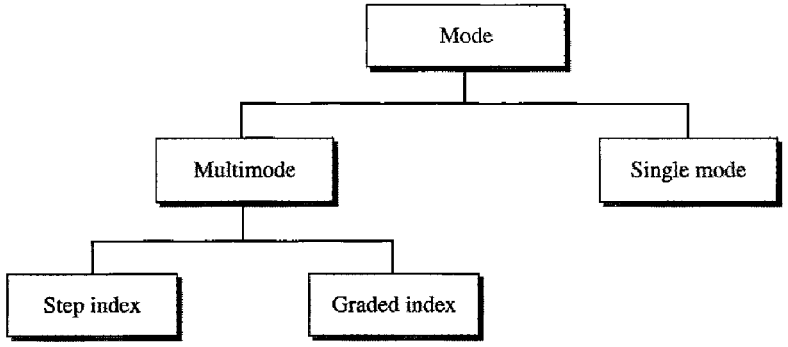


Figure 7.12 *Propagation modes*



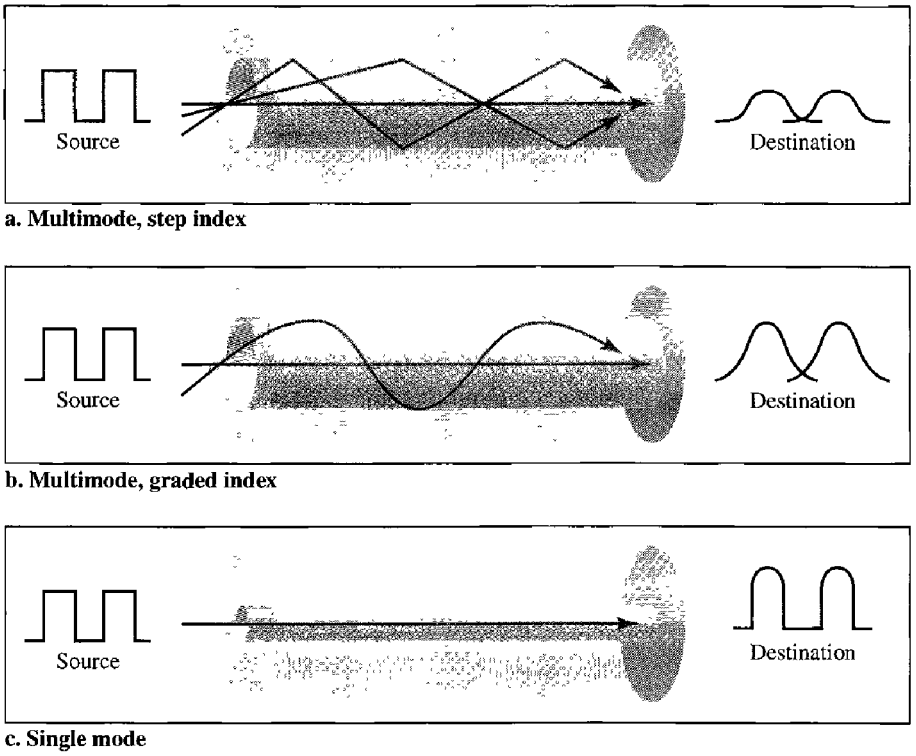
Multimode Multimode is so named because multiple beams from a light source move through the core in different paths. How these beams move within the cable depends on the structure of the core, as shown in Figure 7.13.

In **multimode step-index fiber,** the density of the core remains constant from the center to the edges. A beam of light moves through this constant density in a straight line until it reaches the interface of the core and the cladding. At the interface, there is an abrupt change due to a lower density; this alters the angle of the beam's motion. The term *step index* refers to the suddenness of this change, which contributes to the distortion of the signal as it passes through the fiber.

A second type of fiber, called **multimode graded-index fiber,** decreases this distortion of the signal through the cable. The word *index* here refers to the index of refraction. As we saw above, the index of refraction is related to density. A graded-index fiber, therefore, is one with varying densities. Density is highest at the center of the core and decreases gradually to its lowest at the edge. Figure 7.13 shows the impact of this variable density on the propagation of light beams.

Single-Mode Single-mode uses step-index fiber and a highly focused source of light that limits beams to a small range of angles, all close to the horizontal. The **single-mode fiber** itself is manufactured with a much smaller diameter than that of multimode fiber, and with substantially lower density (index of refraction). The decrease in density results in a critical angle that is close enough to 90° to make the propagation of beams almost horizontal. In this case, propagation of different beams is almost identical, and delays are negligible. All the beams arrive at the destination "together" and can be recombined with little distortion to the signal (see Figure 7.13).

Figure 7.13 *Modes*



a. Multimode, step index

b. Multimode, graded index

c. Single mode

Fiber Sizes

Optical fibers are defined by the ratio of the diameter of their core to the diameter of their cladding, both expressed in micrometers. The common sizes are shown in Table 7.3. Note that the last size listed is for single-mode only.

Table 7.3 *Fiber types*

Type	*Core (μm)*	*Cladding (μm)*	*Mode*
50/125	50.0	125	Multimode, graded index
62.5/125	62.5	125	Multimode, graded index
100/125	100.0	125	Multimode, graded index
7/125	7.0	125	Single mode

Cable Composition

Figure 7.14 shows the composition of a typical fiber-optic cable. The outer jacket is made of either PVC or Teflon. Inside the jacket are Kevlar strands to strengthen the cable. Kevlar is a strong material used in the fabrication of bulletproof vests. Below the Kevlar is another plastic coating to cushion the fiber. The fiber is at the center of the cable, and it consists of cladding and core.

Fiber-Optic Cable Connectors

There are three types of connectors for fiber-optic cables, as shown in Figure 7.15.

Figure 7.14 *Fiber construction*

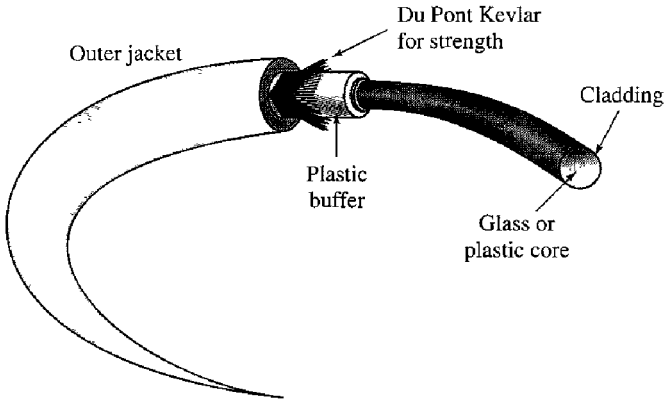
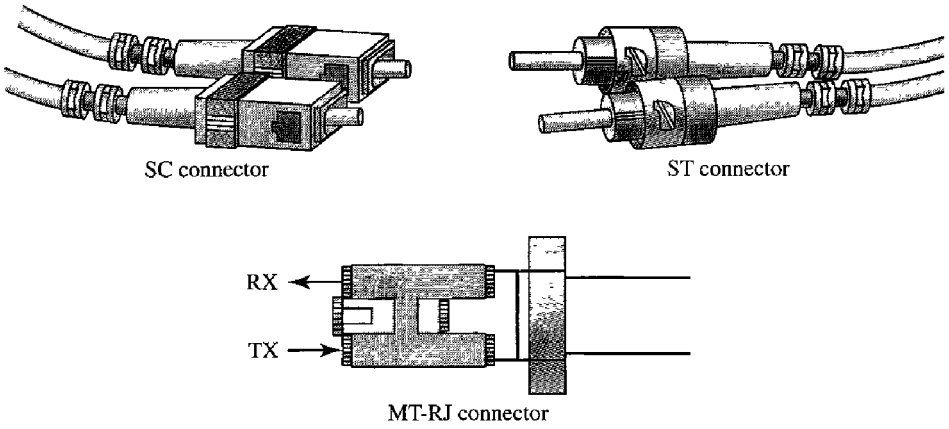


Figure 7.15 *Fiber-optic cable connectors*



The **subscriber channel (SC) connector** is used for cable TV. It uses a push/pull locking system. The **straight-tip (ST) connector** is used for connecting cable to networking devices. It uses a bayonet locking system and is more reliable than SC. **MT-RJ** is a connector that is the same size as RJ45.

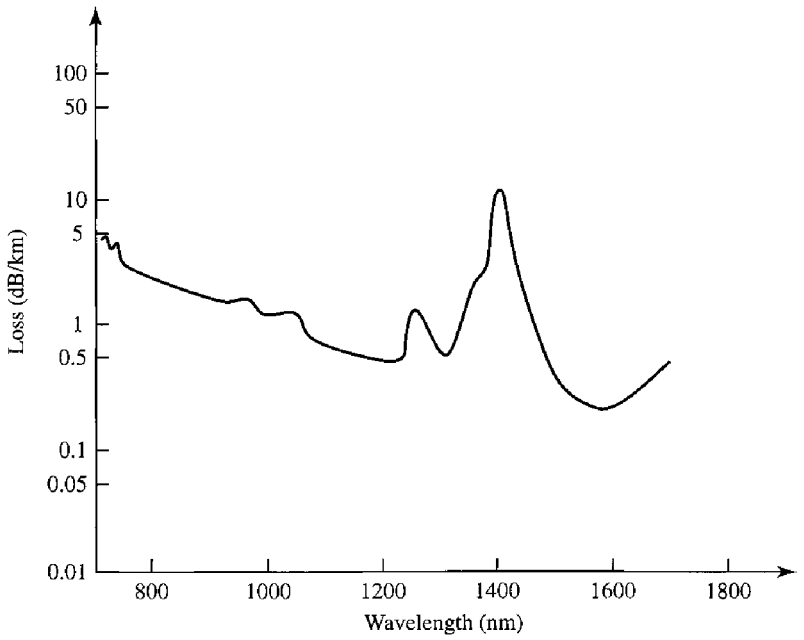
Performance

The plot of attenuation versus wavelength in Figure 7.16 shows a very interesting phenomenon in fiber-optic cable. Attenuation is flatter than in the case of twisted-pair cable and coaxial cable. The performance is such that we need fewer (actually 10 times less) repeaters when we use fiber-optic cable.

Applications

Fiber-optic cable is often found in backbone networks because its wide bandwidth is cost-effective. Today, with wavelength-division multiplexing (WDM), we can transfer

Figure 7.16 *Optical fiber performance*



data at a rate of 1600 Gbps. The SONET network that we discuss in Chapter 17 provides such a backbone.

Some cable TV companies use a combination of optical fiber and coaxial cable, thus creating a hybrid network. Optical fiber provides the backbone structure while coaxial cable provides the connection to the user premises. This is a cost-effective configuration since the narrow bandwidth requirement at the user end does not justify the use of optical fiber.

Local-area networks such as 100Base-FX network (Fast Ethernet) and 1000Base-X also use fiber-optic cable.

Advantages and Disadvantages of Optical Fiber

Advantages Fiber-optic cable has several advantages over metallic cable (twisted-pair or coaxial).

- **Higher bandwidth.** Fiber-optic cable can support dramatically higher bandwidths (and hence data rates) than either twisted-pair or coaxial cable. Currently, data rates and bandwidth utilization over fiber-optic cable are limited not by the medium but by the signal generation and reception technology available.
- **Less signal attenuation.** Fiber-optic transmission distance is significantly greater than that of other guided media. A signal can run for 50 km without requiring regeneration. We need repeaters every 5 km for coaxial or twisted-pair cable.
- **Immunity to electromagnetic interference.** Electromagnetic noise cannot affect fiber-optic cables.
- **Resistance to corrosive materials.** Glass is more resistant to corrosive materials than copper.

- **Light weight.** Fiber-optic cables are much lighter than copper cables.
- **Greater immunity to tapping.** Fiber-optic cables are more immune to tapping than copper cables. Copper cables create antenna effects that can easily be tapped.

Disadvantages There are some disadvantages in the use of optical fiber.

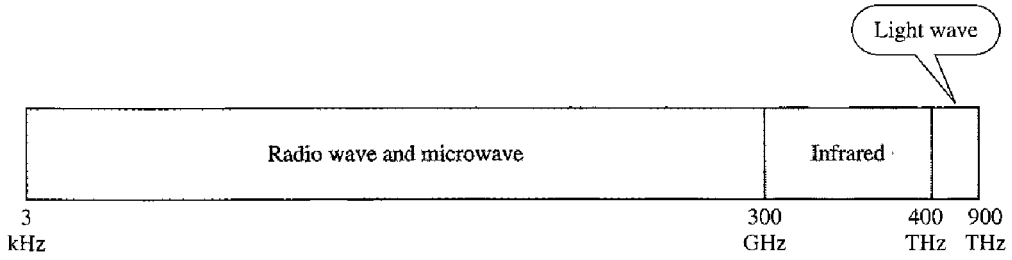
- **Installation and maintenance.** Fiber-optic cable is a relatively new technology. Its installation and maintenance require expertise that is not yet available everywhere.
- **Unidirectional light propagation.** Propagation of light is unidirectional. If we need bidirectional communication, two fibers are needed.
- **Cost.** The cable and the interfaces are relatively more expensive than those of other guided media. If the demand for bandwidth is not high, often the use of optical fiber cannot be justified.

7.2 UNGUIDED MEDIA: WIRELESS

Unguided media transport electromagnetic waves without using a physical conductor. This type of communication is often referred to as **wireless communication.** Signals are normally broadcast through free space and thus are available to anyone who has a device capable of receiving them.

Figure 7.17 shows the part of the electromagnetic spectrum, ranging from 3 kHz to 900 THz, used for wireless communication.

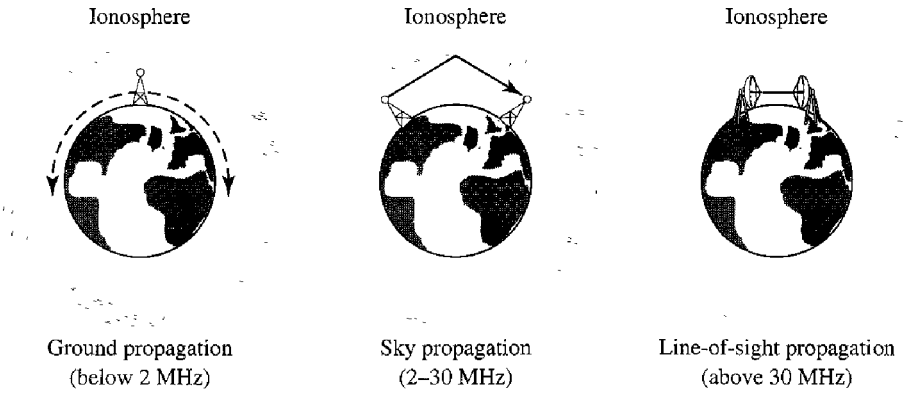
Figure 7.17 *Electromagnetic spectrum for wireless communication*



Unguided signals can travel from the source to destination in several ways: ground propagation, sky propagation, and line-of-sight propagation, as shown in Figure 7.18.

In **ground propagation,** radio waves travel through the lowest portion of the atmosphere, hugging the earth. These low-frequency signals emanate in all directions from the transmitting antenna and follow the curvature of the planet. Distance depends on the amount of power in the signal: The greater the power, the greater the distance. In **sky propagation,** higher-frequency radio waves radiate upward into the ionosphere (the layer of atmosphere where particles exist as ions) where they are reflected back to earth. This type of transmission allows for greater distances with lower output power. In **line-of-sight propagation,** very high-frequency signals are transmitted in straight lines directly from antenna to antenna. Antennas must be directional, facing each other,

Figure 7.18 *Propagation methods*



and either tall enough or close enough together not to be affected by the curvature of the earth. Line-of-sight propagation is tricky because radio transmissions cannot be completely focused.

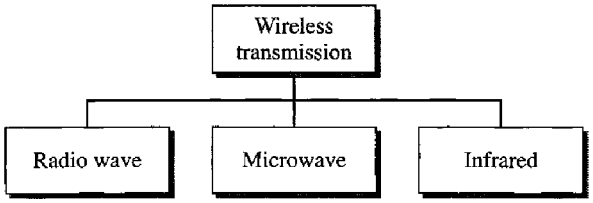
The section of the electromagnetic spectrum defined as radio waves and microwaves is divided into eight ranges, called *bands,* each regulated by government authorities. These bands are rated from *very low frequency* (VLF) to *extremely high frequency* (EHF). Table 7.4 lists these bands, their ranges, propagation methods, and some applications.

Table 7.4 *Bands*

Band	*Range*	*Propagation*	*Application*
VLF (very low frequency)	3–30 kHz	Ground	Long-range radio navigation
LF (low frequency)	30–300 kHz	Ground	Radio beacons and navigational locators
MF (middle frequency)	300 kHz–3 MHz	Sky	AM radio
HF (high frequency)	3–30 MHz	Sky	Citizens band (CB), ship/aircraft communication
VHF (very high frequency)	30–300 MHz	Sky and line-of-sight	VHF TV, FM radio
UHF (ultrahigh frequency)	300 MHz–3 GHz	Line-of-sight	UHF TV, cellular phones, paging, satellite
SHF (superhigh frequency)	3–30 GHz	Line-of-sight	Satellite communication
EHF (extremely high frequency)	30–300 GHz	Line-of-sight	Radar, satellite

We can divide wireless transmission into three broad groups: radio waves, microwaves, and infrared waves. See Figure 7.19.

Figure 7.19 *Wireless transmission waves*



Radio Waves

Although there is no clear-cut demarcation between radio waves and microwaves, electromagnetic waves ranging in frequencies between 3 kHz and 1 GHz are normally called **radio waves;** waves ranging in frequencies between 1 and 300 GHz are called **microwaves.** However, the behavior of the waves, rather than the frequencies, is a better criterion for classification.

Radio waves, for the most part, are omnidirectional. When an antenna transmits radio waves, they are propagated in all directions. This means that the sending and receiving antennas do not have to be aligned. A sending antenna sends waves that can be received by any receiving antenna. The omnidirectional property has a disadvantage, too. The radio waves transmitted by one antenna are susceptible to interference by another antenna that may send signals using the same frequency or band.

Radio waves, particularly those waves that propagate in the sky mode, can travel long distances. This makes radio waves a good candidate for long-distance broadcasting such as AM radio.

Radio waves, particularly those of low and medium frequencies, can penetrate walls. This characteristic can be both an advantage and a disadvantage. It is an advantage because, for example, an AM radio can receive signals inside a building. It is a disadvantage because we cannot isolate a communication to just inside or outside a building. The radio wave band is relatively narrow, just under 1 GHz, compared to the microwave band. When this band is divided into subbands, the subbands are also narrow, leading to a low data rate for digital communications.

Almost the entire band is regulated by authorities (e.g., the FCC in the United States). Using any part of the band requires permission from the authorities.

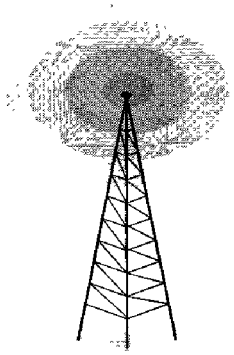
Omnidirectional Antenna

Radio waves use **omnidirectional antennas** that send out signals in all directions. Based on the wavelength, strength, and the purpose of transmission, we can have several types of antennas. Figure 7.20 shows an omnidirectional antenna.

Applications

The omnidirectional characteristics of radio waves make them useful for multicasting, in which there is one sender but many receivers. AM and FM radio, television, maritime radio, cordless phones, and paging are examples of multicasting.

Figure 7.20 *Omnidirectional antenna*



Radio waves are used for multicast communications, such as radio and television, and paging systems.

Microwaves

Electromagnetic waves having frequencies between 1 and 300 GHz are called microwaves.

Microwaves are unidirectional. When an antenna transmits microwave waves, they can be narrowly focused. This means that the sending and receiving antennas need to be aligned. The unidirectional property has an obvious advantage. A pair of antennas can be aligned without interfering with another pair of aligned antennas. The following describes some characteristics of microwave propagation:

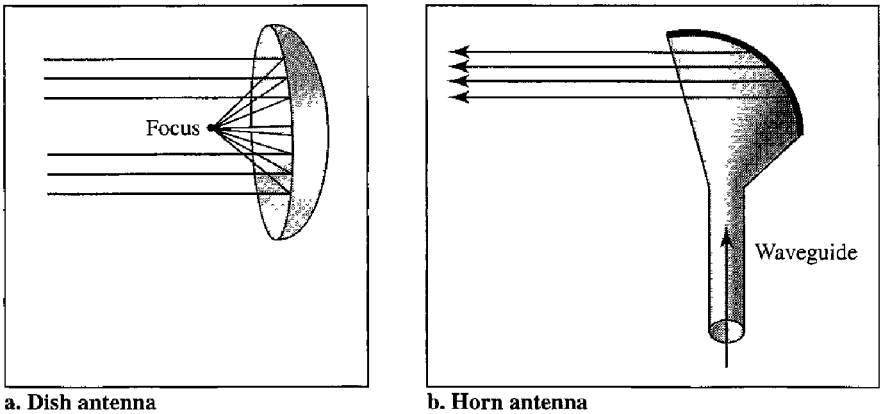
- Microwave propagation is line-of-sight. Since the towers with the mounted antennas need to be in direct sight of each other, towers that are far apart need to be very tall. The curvature of the earth as well as other blocking obstacles do not allow two short towers to communicate by using microwaves. Repeaters are often needed for long-distance communication.
- Very high-frequency microwaves cannot penetrate walls. This characteristic can be a disadvantage if receivers are inside buildings.
- The microwave band is relatively wide, almost 299 GHz. Therefore wider subbands can be assigned, and a high data rate is possible
- Use of certain portions of the band requires permission from authorities.

Unidirectional Antenna

Microwaves need **unidirectional antennas** that send out signals in one direction. Two types of antennas are used for microwave communications: the parabolic dish and the horn (see Figure 7.21).

A **parabolic dish antenna** is based on the geometry of a parabola: Every line parallel to the line of symmetry (line of sight) reflects off the curve at angles such that all the lines intersect in a common point called the focus. The parabolic dish works as a

Figure 7.21 *Unidirectional antennas*



a. Dish antenna b. Horn antenna

funnel, catching a wide range of waves and directing them to a common point. In this way, more of the signal is recovered than would be possible with a single-point receiver.

Outgoing transmissions are broadcast through a horn aimed at the dish. The microwaves hit the dish and are deflected outward in a reversal of the receipt path.

A **horn antenna** looks like a gigantic scoop. Outgoing transmissions are broadcast up a stem (resembling a handle) and deflected outward in a series of narrow parallel beams by the curved head. Received transmissions are collected by the scooped shape of the horn, in a manner similar to the parabolic dish, and are deflected down into the stem.

Applications

Microwaves, due to their unidirectional properties, are very useful when unicast (one-to-one) communication is needed between the sender and the receiver. They are used in cellular phones (Chapter 16), satellite networks (Chapter 16), and wireless LANs (Chapter 14).

Microwaves are used for unicast communication such as cellular telephones, satellite networks, and wireless LANs.

Infrared

Infrared waves, with frequencies from 300 GHz to 400 THz (wavelengths from 1 mm to 770 nm), can be used for short-range communication. Infrared waves, having high frequencies, cannot penetrate walls. This advantageous characteristic prevents interference between one system and another; a short-range communication system in one room cannot be affected by another system in the next room. When we use our infrared remote control, we do not interfere with the use of the remote by our neighbors. However, this same characteristic makes infrared signals useless for long-range communication. In addition, we cannot use infrared waves outside a building because the sun's rays contain infrared waves that can interfere with the communication.

Applications

The infrared band, almost 400 THz, has an excellent potential for data transmission. Such a wide bandwidth can be used to transmit digital data with a very high data rate. The *Infrared Data Association* (IrDA), an association for sponsoring the use of infrared waves, has established standards for using these signals for communication between devices such as keyboards, mice, PCs, and printers. For example, some manufacturers provide a special port called the **IrDA port** that allows a wireless keyboard to communicate with a PC. The standard originally defined a data rate of 75 kbps for a distance up to 8 m. The recent standard defines a data rate of 4 Mbps.

Infrared signals defined by IrDA transmit through line of sight; the IrDA port on the keyboard needs to point to the PC for transmission to occur.

Infrared signals can be used for short-range communication in a closed area using line-of-sight propagation.

7.3 RECOMMENDED READING

For more details about subjects discussed in this chapter, we recommend the following books. The items in brackets [. . .] refer to the reference list at the end of the text.

Books

Transmission media is discussed in Section 3.8 of [GW04], Chapter 4 of [Sta04], Section 2.2 and 2.3 of [Tan03]. [SSS05] gives a full coverage of transmission media.

7.4 KEY TERMS

angle of incidence
Bayone-Neil-Concelman (BNC) connector
cladding
coaxial cable
core
critical angle
electromagnetic spectrum
fiber-optic cable
gauge
ground propagation
guided media
horn antenna
infrared wave
IrDA port
line-of-sight propagation
microwave
MT-RJ
multimode graded-index fiber
multimode step-index fiber
omnidirectional antenna
optical fiber
parabolic dish antenna
Radio Government (RG) number
radio wave
reflection
refraction
RJ45

shielded twisted-pair (STP)
single-mode fiber
sky propagation
straight-tip (ST) connector
subscriber channel (SC) connector
transmission medium
twisted-pair cable
unguided medium
unidirectional antenna
unshielded twisted-pair (UTP)
wireless communication

7.5 SUMMARY

- Transmission media lie below the physical layer.
- A guided medium provides a physical conduit from one device to another. Twisted-pair cable, coaxial cable, and optical fiber are the most popular types of guided media.
- Twisted-pair cable consists of two insulated copper wires twisted together. Twisted-pair cable is used for voice and data communications.
- Coaxial cable consists of a central conductor and a shield. Coaxial cable can carry signals of higher frequency ranges than twisted-pair cable. Coaxial cable is used in cable TV networks and traditional Ethernet LANs.
- Fiber-optic cables are composed of a glass or plastic inner core surrounded by cladding, all encased in an outside jacket. Fiber-optic cables carry data signals in the form of light. The signal is propagated along the inner core by reflection. Fiber-optic transmission is becoming increasingly popular due to its noise resistance, low attenuation, and high-bandwidth capabilities. Fiber-optic cable is used in backbone networks, cable TV networks, and Fast Ethernet networks.
- Unguided media (free space) transport electromagnetic waves without the use of a physical conductor.
- Wireless data are transmitted through ground propagation, sky propagation, and line-of-sight propagation.Wireless waves can be classified as radio waves, microwaves, or infrared waves. Radio waves are omnidirectional; microwaves are unidirectional. Microwaves are used for cellular phone, satellite, and wireless LAN communications.
- Infrared waves are used for short-range communications such as those between a PC and a peripheral device. It can also be used for indoor LANs.

7.6 PRACTICE SET

Review Questions

1. What is the position of the transmission media in the OSI or the Internet model?
2. Name the two major categories of transmission media.
3. How do guided media differ from unguided media?
4. What are the three major classes of guided media?
5. What is the significance of the twisting in twisted-pair cable?

6. What is refraction? What is reflection?
7. What is the purpose of cladding in an optical fiber?
8. Name the advantages of optical fiber over twisted-pair and coaxial cable.
9. How does sky propagation differ from line-of-sight propagation?
10. What is the difference between omnidirectional waves and unidirectional waves?

Exercises

11. Using Figure 7.6, tabulate the attenuation (in dB) of a 18-gauge UTP for the indicated frequencies and distances.

Table 7.5 *Attenuation for 18-gauge UTP*

Distance	*dB at 1 KHz*	*dB at 10 KHz*	*dB at 100 KHz*
1 Km			
10 Km			
15 Km			
20 Km			

12. Use the result of Exercise 11 to infer that the bandwidth of a UTP cable decreases with an increase in distance.
13. If the power at the beginning of a 1 Km 18-gauge UTP is 200 mw, what is the power at the end for frequencies 1 KHz, 10 KHz, and 100 KHz? Use the result of Exercise 11.
14. Using Figure 7.9, tabulate the attenuation (in dB) of a 2.6/9.5 mm coaxial cable for the indicated frequencies and distances.

Table 7.6 *Attenuation for 2.6/9.5 mm coaxial cable*

Distance	*dB at 1 KHz*	*dB at 10 KHz*	*dB at 100 KHz*
1 Km			
10 Km			
15 Km			
20 Km			

15. Use the result of Exercise 14 to infer that the bandwidth of a coaxial cable decreases with the increase in distance.
16. If the power at the beginning of a 1 Km 2.6/9.5 mm coaxial cable is 200 mw, what is the power at the end for frequencies 1 KHz, 10 KHz, and 100 KHz? Use the result of Exercise 14.
17. Calculate the bandwidth of the light for the following wavelength ranges (assume a propagation speed of 2×10^8 m):
    a. 1000 to 1200 nm
    b. 1000 to 1400 nm

18. The horizontal axes in Figure 7.6 and 7.9 represent frequencies. The horizontal axis in Figure 7.16 represents wavelength. Can you explain the reason? If the propagation speed in an optical fiber is 2×10^8 m, can you change the units in the horizontal axis to frequency? Should the vertical-axis units be changed too? Should the curve be changed too?
19. Using Figure 7.16, tabulate the attenuation (in dB) of an optical fiber for the indicated wavelength and distances.

Table 7.7 *Attenuation for optical fiber*

Distance	*dB at 800 nm*	*dB at 1000 nm*	*dB at 1200 nm*
1 Km			
10 Km			
15 Km			
20 Km			

20. A light signal is travelling through a fiber. What is the delay in the signal if the length of the fiber-optic cable is 10 m, 100 m, and 1 Km (assume a propagation speed of 2×10^8 m)?
21. A beam of light moves from one medium to another medium with less density. The critical angle is 60°. Do we have refraction or reflection for each of the following incident angles? Show the bending of the light ray in each case.
    a. 40°
    b. 60°
    c. 80°

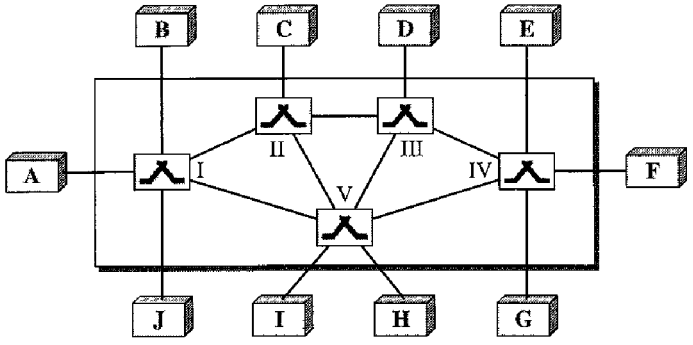
CHAPTER 8

Switching

A network is a set of connected devices. Whenever we have multiple devices, we have the problem of how to connect them to make one-to-one communication possible. One solution is to make a point-to-point connection between each pair of devices (a mesh topology) or between a central device and every other device (a star topology). These methods, however, are impractical and wasteful when applied to very large networks. The number and length of the links require too much infrastructure to be cost-efficient, and the majority of those links would be idle most of the time. Other topologies employing multipoint connections, such as a bus, are ruled out because the distances between devices and the total number of devices increase beyond the capacities of the media and equipment.

A better solution is **switching.** A switched network consists of a series of interlinked nodes, called **switches.** Switches are devices capable of creating temporary connections between two or more devices linked to the switch. In a switched network, some of these nodes are connected to the end systems (computers or telephones, for example). Others are used only for routing. Figure 8.1 shows a switched network.

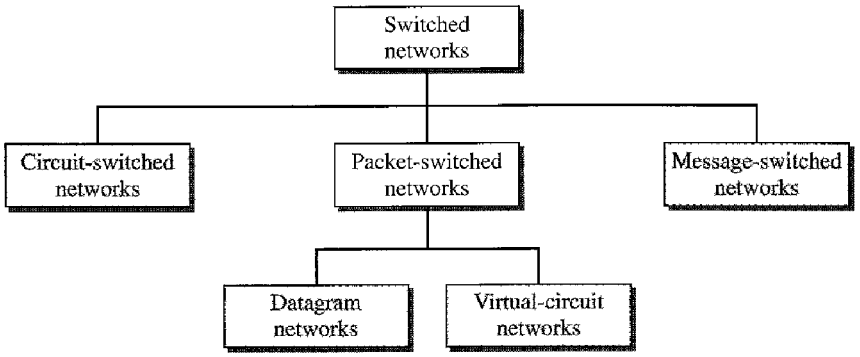
Figure 8.1 *Switched network*



The **end systems** (communicating devices) are labeled A, B, C, D, and so on, and the switches are labeled I, II, III, IV, and V. Each switch is connected to multiple links.

Traditionally, three methods of switching have been important: circuit switching, packet switching, and message switching. The first two are commonly used today. The third has been phased out in general communications but still has networking applications. We can then divide today's networks into three broad categories: circuit-switched networks, packet-switched networks, and message-switched. Packet-switched networks can further be divided into two subcategories—virtual-circuit networks and datagram networks—as shown in Figure 8.2.

Figure 8.2 *Taxonomy of switched networks*



We can say that the virtual-circuit networks have some common characteristics with circuit-switched and datagram networks. Thus, we first discuss circuit-switched networks, then datagram networks, and finally virtual-circuit networks.

Today the tendency in packet switching is to combine datagram networks and virtual-circuit networks. Networks route the first packet based on the datagram addressing idea, but then create a virtual-circuit network for the rest of the packets coming from the same source and going to the same destination. We will see some of these networks in future chapters.

In message switching, each switch stores the whole message and forwards it to the next switch. Although, we don't see message switching at lower layers, it is still used in some applications like electronic mail (e-mail). We will not discuss this topic in this book.

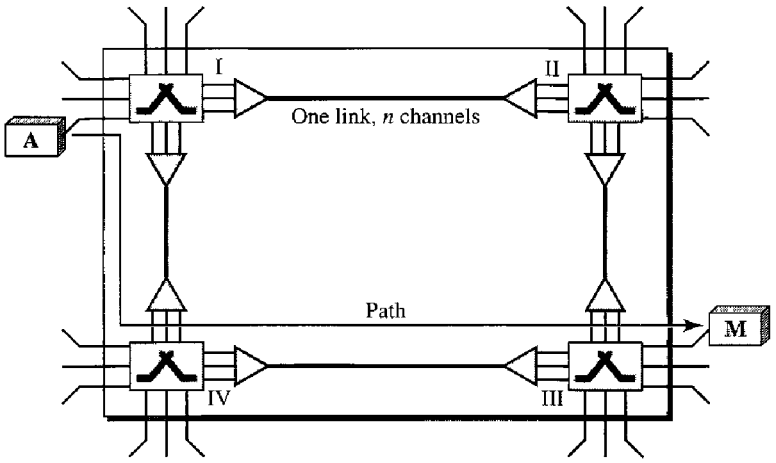
8.1 CIRCUIT-SWITCHED NETWORKS

A **circuit-switched network** consists of a set of switches connected by physical links. A connection between two stations is a dedicated path made of one or more links. However, each connection uses only one dedicated channel on each link. Each link is normally divided into n channels by using FDM or TDM as discussed in Chapter 6.

A circuit-switched network is made of a set of switches connected by physical links, in which each link is divided into n channels.

Figure 8.3 shows a trivial circuit-switched network with four switches and four links. Each link is divided into n (n is 3 in the figure) channels by using FDM or TDM.

Figure 8.3 *A trivial circuit-switched network*



We have explicitly shown the multiplexing symbols to emphasize the division of the link into channels even though multiplexing can be implicitly included in the switch fabric.

The end systems, such as computers or telephones, are directly connected to a switch. We have shown only two end systems for simplicity. When end system A needs to communicate with end system M, system A needs to request a connection to M that must be accepted by all switches as well as by M itself. This is called the **setup phase;** a circuit (channel) is reserved on each link, and the combination of circuits or channels defines the dedicated path. After the dedicated path made of connected circuits (channels) is established, **data transfer** can take place. After all data have been transferred, the circuits are torn down.

We need to emphasize several points here:

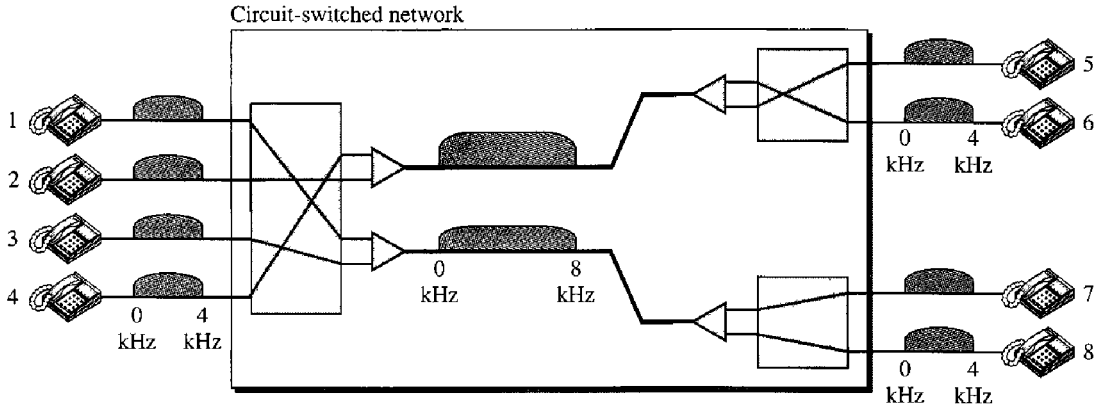
- Circuit switching takes place at the physical layer.
- Before starting communication, the stations must make a reservation for the resources to be used during the communication. These resources, such as channels (bandwidth in FDM and time slots in TDM), switch buffers, switch processing time, and switch input/output ports, must remain dedicated during the entire duration of data transfer until the **teardown phase.**
- Data transferred between the two stations are not packetized (physical layer transfer of the signal). The data are a continuous flow sent by the source station and received by the destination station, although there may be periods of silence.
- There is no addressing involved during data transfer. The switches route the data based on their occupied band (FDM) or time slot (TDM). Of course, there is end-to-end addressing used during the setup phase, as we will see shortly.

In circuit switching, the resources need to be reserved during the setup phase; the resources remain dedicated for the entire duration of data transfer until the teardown phase.

Example 8.1

As a trivial example, let us use a circuit-switched network to connect eight telephones in a small area. Communication is through 4-kHz voice channels. We assume that each link uses FDM to connect a maximum of two voice channels. The bandwidth of each link is then 8 kHz. Figure 8.4 shows the situation. Telephone 1 is connected to telephone 7; 2 to 5; 3 to 8; and 4 to 6. Of course the situation may change when new connections are made. The switch controls the connections.

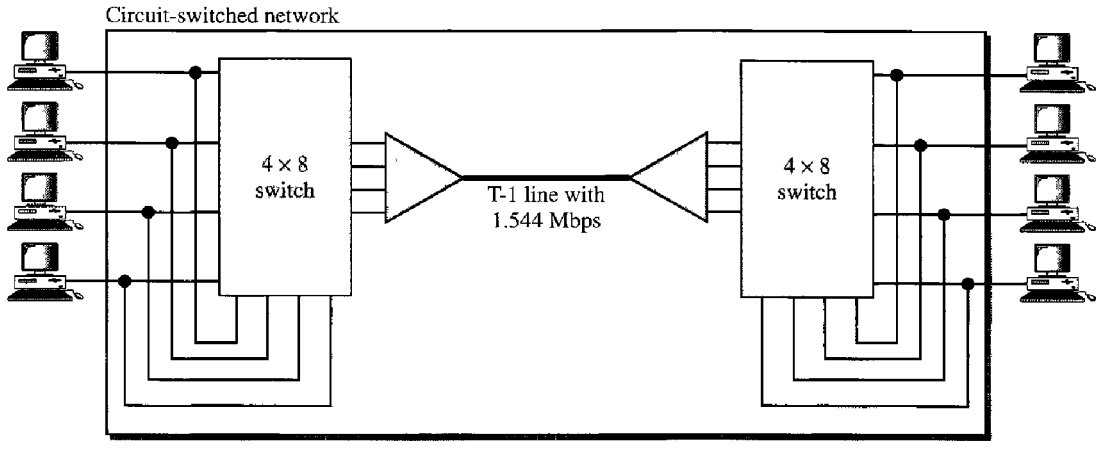
Figure 8.4 *Circuit-switched network used in Example 8.1*



Example 8.2

As another example, consider a circuit-switched network that connects computers in two remote offices of a private company. The offices are connected using a T-1 line leased from a communication service provider. There are two 4×8 (4 inputs and 8 outputs) switches in this network. For each switch, four output ports are folded into the input ports to allow communication between computers in the same office. Four other output ports allow communication between the two offices. Figure 8.5 shows the situation.

Figure 8.5 *Circuit-switched network used in Example 8.2*



Three Phases

The actual communication in a circuit-switched network requires three phases: connection setup, data transfer, and connection teardown.

Setup Phase

Before the two parties (or multiple parties in a conference call) can communicate, a dedicated circuit (combination of channels in links) needs to be established. The end systems are normally connected through dedicated lines to the switches, so connection setup means creating dedicated channels between the switches. For example, in Figure 8.3, when system A needs to connect to system M, it sends a setup request that includes the address of system M, to switch I. Switch I finds a channel between itself and switch IV that can be dedicated for this purpose. Switch I then sends the request to switch IV, which finds a dedicated channel between itself and switch III. Switch III informs system M of system A's intention at this time.

In the next step to making a connection, an acknowledgment from system M needs to be sent in the opposite direction to system A. Only after system A receives this acknowledgment is the connection established.

Note that end-to-end addressing is required for creating a connection between the two end systems. These can be, for example, the addresses of the computers assigned by the administrator in a TDM network, or telephone numbers in an FDM network.

Data Transfer Phase

After the establishment of the dedicated circuit (channels), the two parties can transfer data.

Teardown Phase

When one of the parties needs to disconnect, a signal is sent to each switch to release the resources.

Efficiency

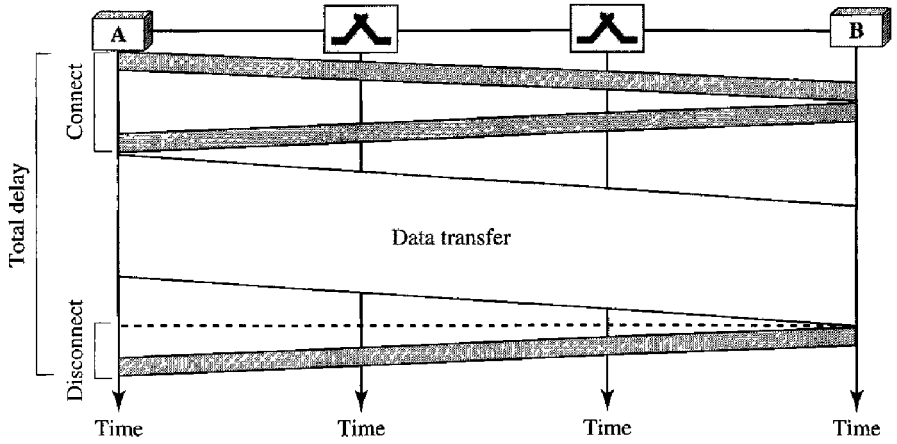
It can be argued that circuit-switched networks are not as efficient as the other two types of networks because resources are allocated during the entire duration of the connection. These resources are unavailable to other connections. In a telephone network, people normally terminate the communication when they have finished their conversation. However, in computer networks, a computer can be connected to another computer even if there is no activity for a long time. In this case, allowing resources to be dedicated means that other connections are deprived.

Delay

Although a circuit-switched network normally has low efficiency, the delay in this type of network is minimal. During data transfer the data are not delayed at each switch; the resources are allocated for the duration of the connection. Figure 8.6 shows the idea of delay in a circuit-switched network when only two switches are involved.

As Figure 8.6 shows, there is no waiting time at each switch. The total delay is due to the time needed to create the connection, transfer data, and disconnect the circuit. The

Figure 8.6 *Delay in a circuit-switched network*



delay caused by the setup is the sum of four parts: the propagation time of the source computer request (slope of the first gray box), the request signal transfer time (height of the first gray box), the propagation time of the acknowledgment from the destination computer (slope of the second gray box), and the signal transfer time of the acknowledgment (height of the second gray box). The delay due to data transfer is the sum of two parts: the propagation time (slope of the colored box) and data transfer time (height of the colored box), which can be very long. The third box shows the time needed to tear down the circuit. We have shown the case in which the receiver requests disconnection, which creates the maximum delay.

Circuit-Switched Technology in Telephone Networks

As we will see in Chapter 9, the telephone companies have previously chosen the circuit-switched approach to switching in the physical layer; today the tendency is moving toward other switching techniques. For example, the telephone number is used as the global address, and a signaling system (called SS7) is used for the setup and teardown phases.

Switching at the physical layer in the traditional telephone network uses the circuit-switching approach.

8.2 DATAGRAM NETWORKS

In data communications, we need to send messages from one end system to another. If the message is going to pass through a packet-switched network, it needs to be divided into packets of fixed or variable size. The size of the packet is determined by the network and the governing protocol.

In packet switching, there is no resource allocation for a packet. This means that there is no reserved bandwidth on the links, and there is no scheduled processing time

for each packet. Resources are allocated on demand. The allocation is done on a first-come, first-served basis. When a switch receives a packet, no matter what is the source or destination, the packet must wait if there are other packets being processed. As with other systems in our daily life, this lack of reservation may create delay. For example, if we do not have a reservation at a restaurant, we might have to wait.

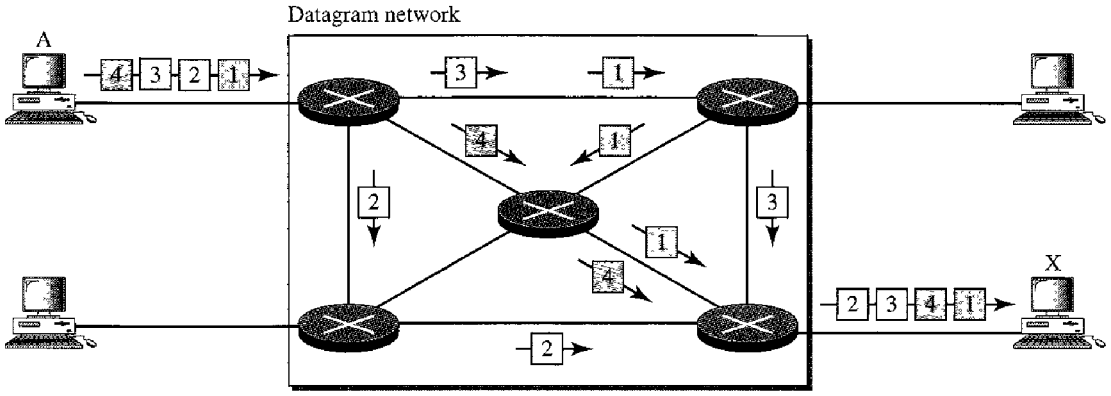
In a packet-switched network, there is no resource reservation; resources are allocated on demand.

In a **datagram network,** each packet is treated independently of all others. Even if a packet is part of a multipacket transmission, the network treats it as though it existed alone. Packets in this approach are referred to as **datagrams.**

Datagram switching is normally done at the network layer. We briefly discuss datagram networks here as a comparison with circuit-switched and virtual-circuit-switched networks. In Part 4 of this text, we go into greater detail.

Figure 8.7 shows how the datagram approach is used to deliver four packets from station A to station X. The switches in a datagram network are traditionally referred to as routers. That is why we use a different symbol for the switches in the figure.

Figure 8.7 *A datagram network with four switches (routers)*



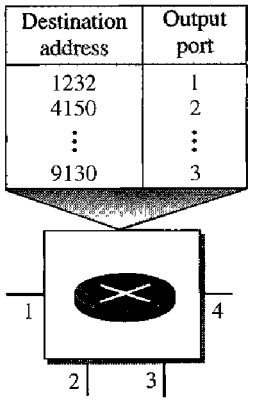
In this example, all four packets (or datagrams) belong to the same message, but may travel different paths to reach their destination. This is so because the links may be involved in carrying packets from other sources and do not have the necessary bandwidth available to carry all the packets from A to X. This approach can cause the datagrams of a transmission to arrive at their destination out of order with different delays between the packets. Packets may also be lost or dropped because of a lack of resources. In most protocols, it is the responsibility of an upper-layer protocol to reorder the datagrams or ask for lost datagrams before passing them on to the application.

The datagram networks are sometimes referred to as **connectionless networks.** The term *connectionless* here means that the switch (packet switch) does not keep information about the connection state. There are no setup or teardown phases. Each packet is treated the same by a switch regardless of its source or destination.

Routing Table

If there are no setup or teardown phases, how are the packets routed to their destinations in a datagram network? In this type of network, each switch (or packet switch) has a routing table which is based on the destination address. The routing tables are dynamic and are updated periodically. The destination addresses and the corresponding forwarding output ports are recorded in the tables. This is different from the table of a circuit-switched network in which each entry is created when the setup phase is completed and deleted when the teardown phase is over. Figure 8.8 shows the routing table for a switch.

Figure 8.8 *Routing table in a datagram network*



A switch in a datagram network uses a routing table that is based on the destination address.

Destination Address

Every packet in a datagram network carries a header that contains, among other information, the destination address of the packet. When the switch receives the packet, this destination address is examined; the routing table is consulted to find the corresponding port through which the packet should be forwarded. This address, unlike the address in a virtual-circuit-switched network, remains the same during the entire journey of the packet.

The destination address in the header of a packet in a datagram network remains the same during the entire journey of the packet.

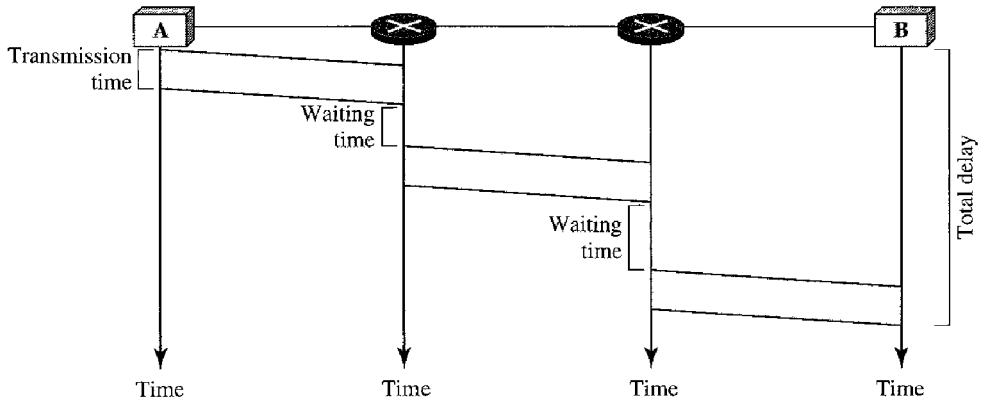
Efficiency

The efficiency of a datagram network is better than that of a circuit-switched network; resources are allocated only when there are packets to be transferred. If a source sends a packet and there is a delay of a few minutes before another packet can be sent, the resources can be reallocated during these minutes for other packets from other sources.

Delay

There may be greater delay in a datagram network than in a virtual-circuit network. Although there are no setup and teardown phases, each packet may experience a wait at a switch before it is forwarded. In addition, since not all packets in a message necessarily travel through the same switches, the delay is not uniform for the packets of a message. Figure 8.9 gives an example of delay in a datagram network for one single packet.

Figure 8.9 *Delay in a datagram network*



The packet travels through two switches. There are three transmission times ($3T$), three propagation delays (slopes 3τ of the lines), and two waiting times ($w_1 + w_2$). We ignore the processing time in each switch. The total delay is

$$\text{Total delay} = 3T + 3\tau + w_1 + w_2$$

Datagram Networks in the Internet

As we will see in future chapters, the Internet has chosen the datagram approach to switching at the network layer. It uses the universal addresses defined in the network layer to route packets from the source to the destination.

Switching in the Internet is done by using the datagram approach to packet switching at the network layer.

8.3 VIRTUAL-CIRCUIT NETWORKS

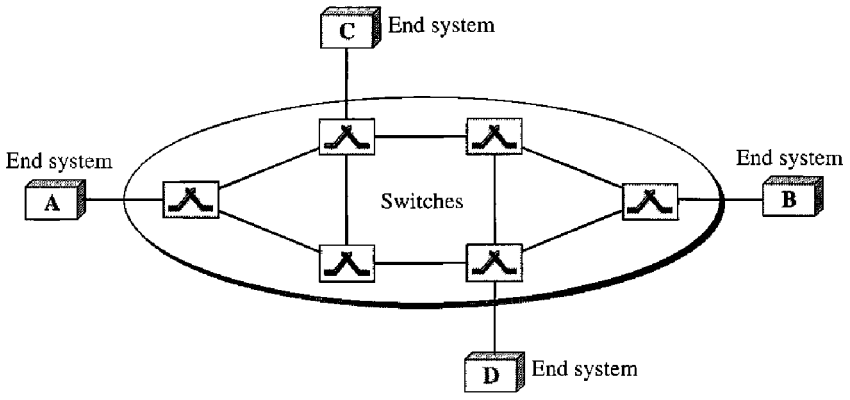
A **virtual-circuit network** is a cross between a circuit-switched network and a datagram network. It has some characteristics of both.

1. As in a circuit-switched network, there are setup and teardown phases in addition to the data transfer phase.

2. Resources can be allocated during the setup phase, as in a circuit-switched network, or on demand, as in a datagram network.
3. As in a datagram network, data are packetized and each packet carries an address in the header. However, the address in the header has local jurisdiction (it defines what should be the next switch and the channel on which the packet is being carried), not end-to-end jurisdiction. The reader may ask how the intermediate switches know where to send the packet if there is no final destination address carried by a packet. The answer will be clear when we discuss virtual-circuit identifiers in the next section.
4. As in a circuit-switched network, all packets follow the same path established during the connection.
5. A virtual-circuit network is normally implemented in the data link layer, while a circuit-switched network is implemented in the physical layer and a datagram network in the network layer. But this may change in the future.

Figure 8.10 is an example of a virtual-circuit network. The network has switches that allow traffic from sources to destinations. A source or destination can be a computer, packet switch, bridge, or any other device that connects other networks.

Figure 8.10 *Virtual-circuit network*



Addressing

In a virtual-circuit network, two types of addressing are involved: global and local (virtual-circuit identifier).

Global Addressing

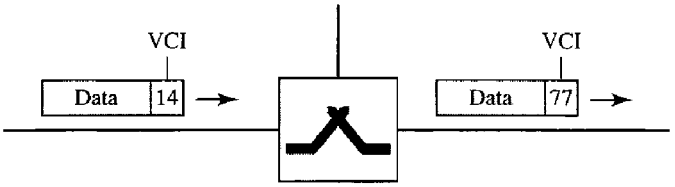
A source or a destination needs to have a global address—an address that can be unique in the scope of the network or internationally if the network is part of an international network. However, we will see that a global address in virtual-circuit networks is used only to create a virtual-circuit identifier, as discussed next.

Virtual-Circuit Identifier

The identifier that is actually used for data transfer is called the **virtual-circuit identifier (VCI).** A VCI, unlike a global address, is a small number that has only switch scope; it

is used by a frame between two switches. When a frame arrives at a switch, it has a VCI; when it leaves, it has a different VCI. Figure 8.11 shows how the VCI in a data frame changes from one switch to another. Note that a VCI does not need to be a large number since each switch can use its own unique set of VCIs.

Figure 8.11 *Virtual-circuit identifier*



Three Phases

As in a circuit-switched network, a source and destination need to go through three phases in a virtual-circuit network: setup, data transfer, and teardown. In the setup phase, the source and destination use their global addresses to help switches make table entries for the connection. In the teardown phase, the source and destination inform the switches to delete the corresponding entry. Data transfer occurs between these two phases. We first discuss the data transfer phase, which is more straightforward; we then talk about the setup and teardown phases.

Data Transfer Phase

To transfer a frame from a source to its destination, all switches need to have a table entry for this virtual circuit. The table, in its simplest form, has four columns. This means that the switch holds four pieces of information for each virtual circuit that is already set up. We show later how the switches make their table entries, but for the moment we assume that each switch has a table with entries for all active virtual circuits. Figure 8.12 shows such a switch and its corresponding table.

Figure 8.12 shows a frame arriving at port 1 with a VCI of 14. When the frame arrives, the switch looks in its table to find port 1 and a VCI of 14. When it is found, the switch knows to change the VCI to 22 and send out the frame from port 3.

Figure 8.13 shows how a frame from source A reaches destination B and how its VCI changes during the trip. Each switch changes the VCI and routes the frame.

The data transfer phase is active until the source sends all its frames to the destination. The procedure at the switch is the same for each frame of a message. The process creates a virtual circuit, not a real circuit, between the source and destination.

Setup Phase

In the setup phase, a switch creates an entry for a virtual circuit. For example, suppose source A needs to create a virtual circuit to B. Two steps are required: the setup request and the acknowledgment.

Figure 8.12 *Switch and tables in a virtual-circuit network*

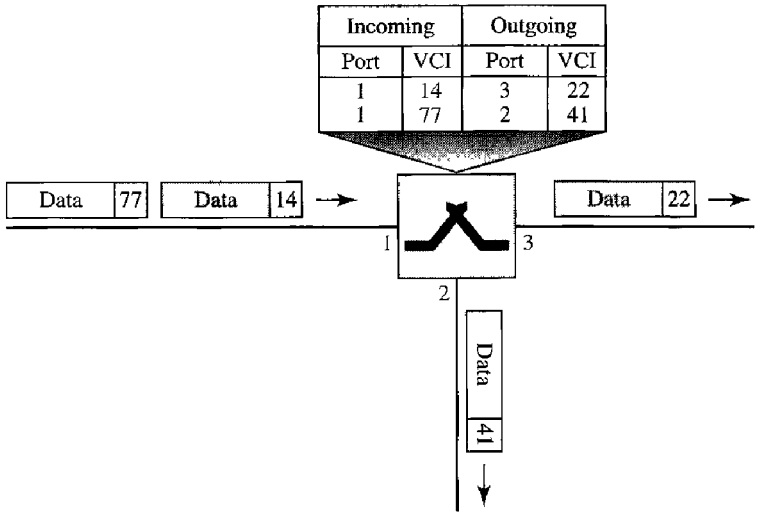
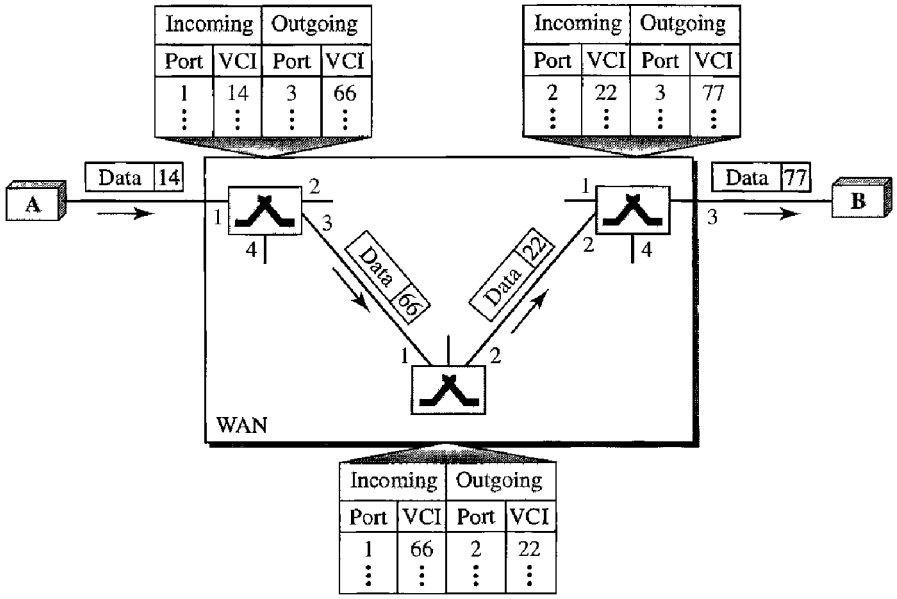


Figure 8.13 *Source-to-destination data transfer in a virtual-circuit network*

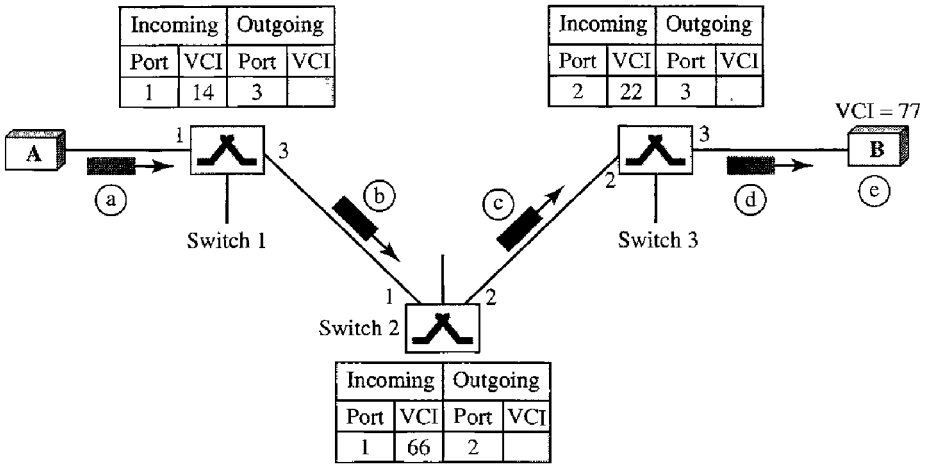


Setup Request A setup request frame is sent from the source to the destination. Figure 8.14 shows the process.

a. Source A sends a setup frame to switch 1.

b. Switch 1 receives the setup request frame. It knows that a frame going from A to B goes out through port 3. How the switch has obtained this information is a point covered in future chapters. The switch, in the setup phase, acts as a packet switch; it has a routing table which is different from the switching table. For the moment, assume that it knows the output port. The switch creates an entry in its table for

Figure 8.14 *Setup request in a virtual-circuit network*



this virtual circuit, but it is only able to fill three of the four columns. The switch assigns the incoming port (1) and chooses an available incoming VCI (14) and the outgoing port (3). It does not yet know the outgoing VCI, which will be found during the acknowledgment step. The switch then forwards the frame through port 3 to switch 2.

c. Switch 2 receives the setup request frame. The same events happen here as at switch 1; three columns of the table are completed: in this case, incoming port (1), incoming VCI (66), and outgoing port (2).

d. Switch 3 receives the setup request frame. Again, three columns are completed: incoming port (2), incoming VCI (22), and outgoing port (3).

e. Destination B receives the setup frame, and if it is ready to receive frames from A, it assigns a VCI to the incoming frames that come from A, in this case 77. This VCI lets the destination know that the frames come from A, and not other sources.

Acknowledgment A special frame, called the acknowledgment frame, completes the entries in the switching tables. Figure 8.15 shows the process.

a. The destination sends an acknowledgment to switch 3. The acknowledgment carries the global source and destination addresses so the switch knows which entry in the table is to be completed. The frame also carries VCI 77, chosen by the destination as the incoming VCI for frames from A. Switch 3 uses this VCI to complete the outgoing VCI column for this entry. Note that 77 is the incoming VCI for destination B, but the outgoing VCI for switch 3.

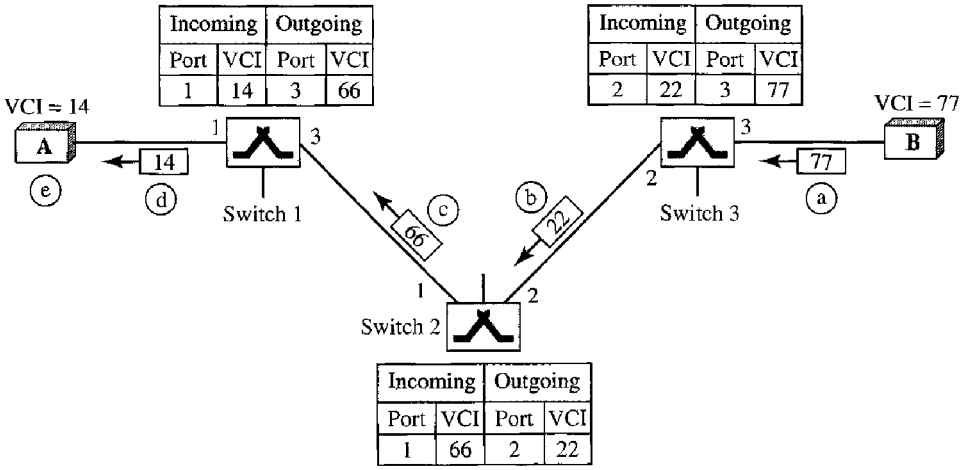
b. Switch 3 sends an acknowledgment to switch 2 that contains its incoming VCI in the table, chosen in the previous step. Switch 2 uses this as the outgoing VCI in the table.

c. Switch 2 sends an acknowledgment to switch 1 that contains its incoming VCI in the table, chosen in the previous step. Switch 1 uses this as the outgoing VCI in the table.

d. Finally switch 1 sends an acknowledgment to source A that contains its incoming VCI in the table, chosen in the previous step.

e. The source uses this as the outgoing VCI for the data frames to be sent to destination B.

Figure 8.15 *Setup acknowledgment in a virtual-circuit network*



Teardown Phase

In this phase, source A, after sending all frames to B, sends a special frame called a *teardown request*. Destination B responds with a teardown confirmation frame. All switches delete the corresponding entry from their tables.

Efficiency

As we said before, resource reservation in a virtual-circuit network can be made during the setup or can be on demand during the data transfer phase. In the first case, the delay for each packet is the same; in the second case, each packet may encounter different delays. There is one big advantage in a virtual-circuit network even if resource allocation is on demand. The source can check the availability of the resources, without actually reserving it. Consider a family that wants to dine at a restaurant. Although the restaurant may not accept reservations (allocation of the tables is on demand), the family can call and find out the waiting time. This can save the family time and effort.

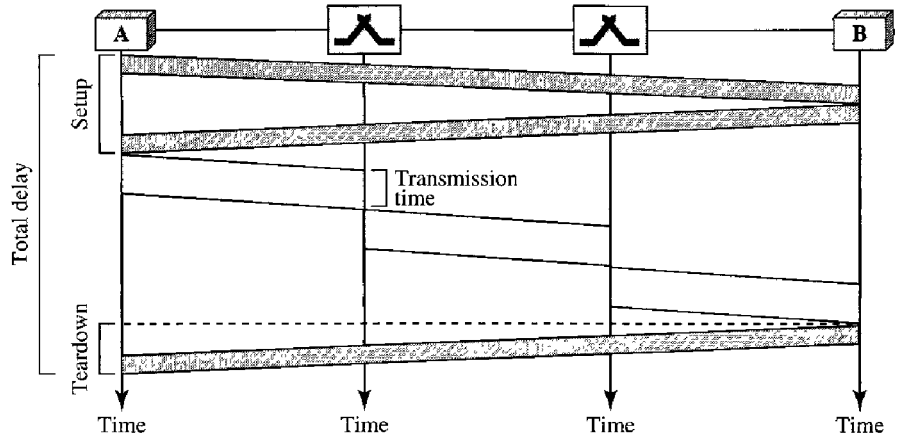
In virtual-circuit switching, all packets belonging to the same source and destination travel the same path; but the packets may arrive at the destination with different delays if resource allocation is on demand.

Delay in Virtual-Circuit Networks

In a virtual-circuit network, there is a one-time delay for setup and a one-time delay for teardown. If resources are allocated during the setup phase, there is no wait time for individual packets. Figure 8.16 shows the delay for a packet traveling through two switches in a virtual-circuit network.

The packet is traveling through two switches (routers). There are three transmission times ($3T$), three propagation times (3τ), data transfer depicted by the sloping lines, a setup delay (which includes transmission and propagation in two directions),

Figure 8.16 *Delay in a virtual-circuit network*



and a teardown delay (which includes transmission and propagation in one direction). We ignore the processing time in each switch. The total delay time is

$$\text{Total delay} = 3T + 3\tau + \text{setup delay} + \text{teardown delay}$$

Circuit-Switched Technology in WANs

As we will see in Chapter 18, virtual-circuit networks are used in switched WANs such as Frame Relay and ATM networks. The data link layer of these technologies is well suited to the virtual-circuit technology.

Switching at the data link layer in a switched WAN is normally implemented by using virtual-circuit techniques.

8.4 STRUCTURE OF A SWITCH

We use switches in circuit-switched and packet-switched networks. In this section, we discuss the structures of the switches used in each type of network.

Structure of Circuit Switches

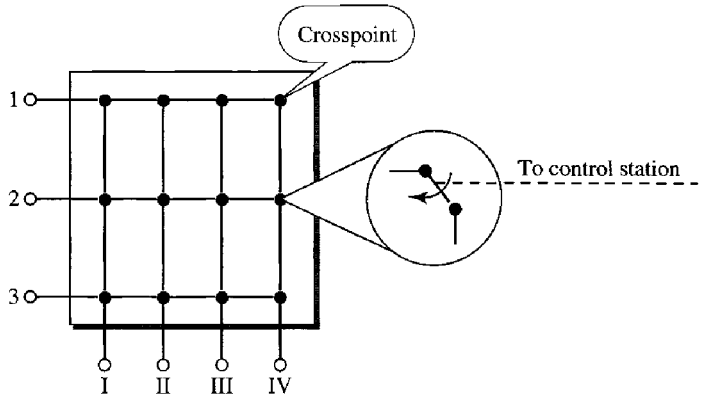
Circuit switching today can use either of two technologies: the space-division switch or the time-division switch.

Space-Division Switch

In **space-division switching,** the paths in the circuit are separated from one another spatially. This technology was originally designed for use in analog networks but is used currently in both analog and digital networks. It has evolved through a long history of many designs.

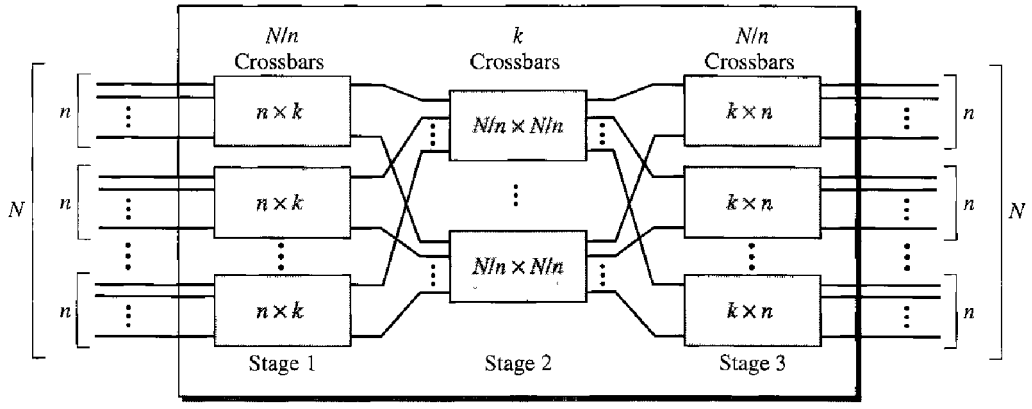
Crossbar Switch A **crossbar switch** connects n inputs to m outputs in a grid, using electronic microswitches (transistors) at each **crosspoint** (see Figure 8.17). The major limitation of this design is the number of crosspoints required. To connect n inputs to m outputs using a crossbar switch requires $n \times m$ crosspoints. For example, to connect 1000 inputs to 1000 outputs requires a switch with 1,000,000 crosspoints. A crossbar with this number of crosspoints is impractical. Such a switch is also inefficient because statistics show that, in practice, fewer than 25 percent of the crosspoints are in use at any given time. The rest are idle.

Figure 8.17 *Crossbar switch with three inputs and four outputs*



Multistage Switch The solution to the limitations of the crossbar switch is the **multistage switch,** which combines crossbar switches in several (normally three) stages, as shown in Figure 8.18. In a single crossbar switch, only one row or column (one path) is active for any connection. So we need $N \times N$ crosspoints. If we can allow multiple paths inside the switch, we can decrease the number of crosspoints. Each crosspoint in the middle stage can be accessed by multiple crosspoints in the first or third stage.

Figure 8.18 *Multistage switch*



To design a three-stage switch, we follow these steps:

1. We divide the N input lines into groups, each of n lines. For each group, we use one crossbar of size $n \times k$, where k is the number of crossbars in the middle stage. In other words, the first stage has N/n crossbars of $n \times k$ crosspoints.
2. We use k crossbars, each of size $(N/n) \times (N/n)$ in the middle stage.
3. We use N/n crossbars, each of size $k \times n$ at the third stage.

We can calculate the total number of crosspoints as follows:

$$\frac{N}{n}(n \times k) + k\left(\frac{N}{n} \times \frac{N}{n}\right) + \frac{N}{n}(k \times n) = 2kN + k\left(\frac{N}{n}\right)^2$$

In a three-stage switch, the total number of crosspoints is

$$2kN + k\left(\frac{N}{n}\right)^2$$

which is much smaller than the number of crosspoints in a single-stage switch (N^2).

Example 8.3

Design a three-stage, 200 × 200 switch ($N = 200$) with $k = 4$ and $n = 20$.

Solution

In the first stage we have N/n or 10 crossbars, each of size 20 × 4. In the second stage, we have 4 crossbars, each of size 10 × 10. In the third stage, we have 10 crossbars, each of size 4 × 20. The total number of crosspoints is $2kN + k(N/n)^2$, or 2000 crosspoints. This is 5 percent of the number of crosspoints in a single-stage switch (200 × 200 = 40,000).

The multistage switch in Example 8.3 has one drawback—**blocking** during periods of heavy traffic. The whole idea of multistage switching is to share the crosspoints in the middle-stage crossbars. Sharing can cause a lack of availability if the resources are limited and all users want a connection at the same time. Blocking refers to times when one input cannot be connected to an output because there is no path available between them—all the possible intermediate switches are occupied.

In a single-stage switch, blocking does not occur because every combination of input and output has its own crosspoint; there is always a path. (Cases in which two inputs are trying to contact the same output do not count. That path is not blocked; the output is merely busy.) In the multistage switch described in Example 8.3, however, only 4 of the first 20 inputs can use the switch at a time, only 4 of the second 20 inputs can use the switch at a time, and so on. The small number of crossbars at the middle stage creates blocking.

In large systems, such as those having 10,000 inputs and outputs, the number of stages can be increased to cut down on the number of crosspoints required. As the number of stages increases, however, possible blocking increases as well. Many people have experienced blocking on public telephone systems in the wake of a natural disaster when the calls being made to check on or reassure relatives far outnumber the regular load of the system.

Clos investigated the condition of nonblocking in multistage switches and came up with the following formula. In a nonblocking switch, the number of middle-stage switches must be at least $2n - 1$. In other words, we need to have $k \geq 2n - 1$.

Note that the number of crosspoints is still smaller than that in a single-stage switch. Now we need to minimize the number of crosspoints with a fixed N by using the Clos criteria. We can take the derivative of the equation with respect to n (the only variable) and find the value of n that makes the result zero. This n must be equal to or greater than $(N/2)^{1/2}$. In this case, the total number of crosspoints is greater than or equal to $4N\,[(2N)^{1/2} - 1]$. In other words, the minimum number of crosspoints according to the Clos criteria is proportional to $N^{3/2}$.

According to Clos criterion:

$n = (N/2)^{1/2}$

$k > 2n - 1$

Total number of crosspoints $\geq 4N\,[(2N)^{1/2} - 1]$

Example 8.4

Redesign the previous three-stage, 200 × 200 switch, using the Clos criteria with a minimum number of crosspoints.

Solution

We let $n = (200/2)^{1/2}$, or $n = 10$. We calculate $k = 2n - 1 = 19$. In the first stage, we have 200/10, or 20, crossbars, each with 10×19 crosspoints. In the second stage, we have 19 crossbars, each with 10×10 crosspoints. In the third stage, we have 20 crossbars each with 19×10 crosspoints. The total number of crosspoints is $20(10 \times 19) + 19(10 \times 10) + 20(19 \times 10) = 9500$. If we use a single-stage switch, we need $200 \times 200 = 40{,}000$ crosspoints. The number of crosspoints in this three-stage switch is 24 percent that of a single-stage switch. More points are needed than in Example 8.3 (5 percent). The extra crosspoints are needed to prevent blocking.

A multistage switch that uses the Clos criteria and a minimum number of crosspoints still requires a huge number of crosspoints. For example, to have a 100,000 input/output switch, we need something close to 200 million crosspoints (instead of 10 billion). This means that if a telephone company needs to provide a switch to connect 100,000 telephones in a city, it needs 200 million crosspoints. The number can be reduced if we accept blocking. Today, telephone companies use time-division switching or a combination of space- and time-division switches, as we will see shortly.

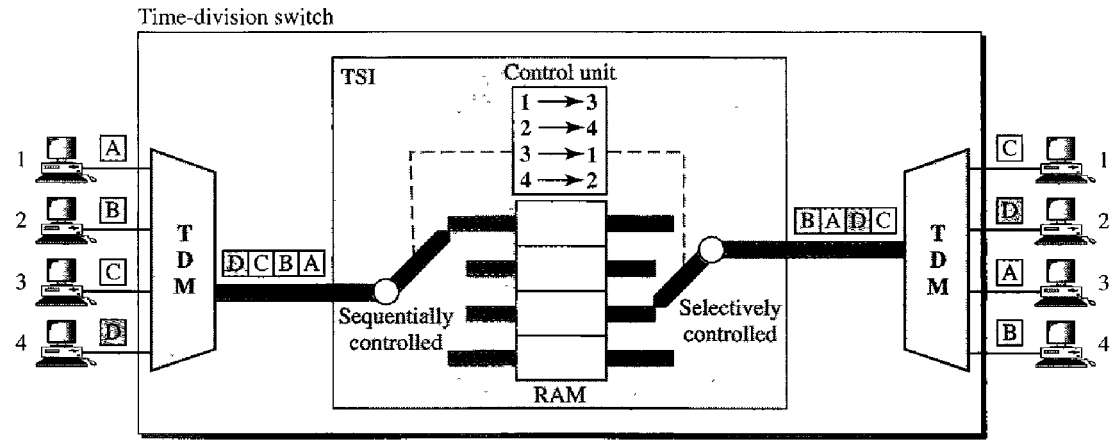
Time-Division Switch

Time-division switching uses time-division multiplexing (TDM) inside a switch. The most popular technology is called the **time-slot interchange (TSI).**

Time-Slot Interchange Figure 8.19 shows a system connecting four input lines to four output lines. Imagine that each input line wants to send data to an output line according to the following pattern:

$1 \rightarrow 3 \quad 2 \rightarrow 4 \quad 3 \rightarrow 1 \quad 4 \rightarrow 2$

Figure 8.19 *Time-slot interchange*



The figure combines a TDM multiplexer, a TDM demultiplexer, and a TSI consisting of random access memory (RAM) with several memory locations. The size of each location is the same as the size of a single time slot. The number of locations is the same as the number of inputs (in most cases, the numbers of inputs and outputs are equal). The RAM fills up with incoming data from time slots in the order received. Slots are then sent out in an order based on the decisions of a control unit.

Time- and Space-Division Switch Combinations

When we compare space-division and time-division switching, some interesting facts emerge. The advantage of space-division switching is that it is instantaneous. Its disadvantage is the number of crosspoints required to make space-division switching acceptable in terms of blocking.

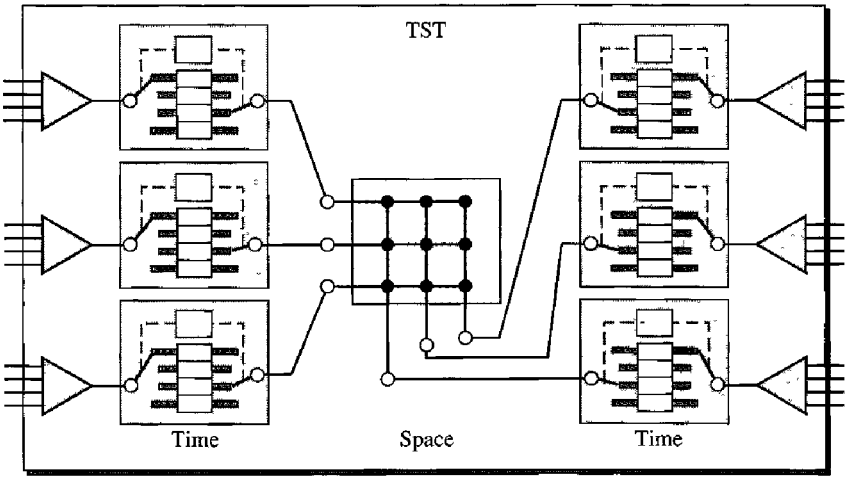
The advantage of time-division switching is that it needs no crosspoints. Its disadvantage, in the case of TSI, is that processing each connection creates delays. Each time slot must be stored by the RAM, then retrieved and passed on.

In a third option, we combine space-division and time-division technologies to take advantage of the best of both. Combining the two results in switches that are optimized both physically (the number of crosspoints) and temporally (the amount of delay). Multistage switches of this sort can be designed as **time-space-time (TST) switch.**

Figure 8.20 shows a simple TST switch that consists of two time stages and one space stage and has 12 inputs and 12 outputs. Instead of one time-division switch, it divides the inputs into three groups (of four inputs each) and directs them to three time-slot interchanges. The result is that the average delay is one-third of what would result from using one time-slot interchange to handle all 12 inputs.

The last stage is a mirror image of the first stage. The middle stage is a space-division switch (crossbar) that connects the TSI groups to allow connectivity between all possible input and output pairs (e.g., to connect input 3 of the first group to output 7 of the second group).

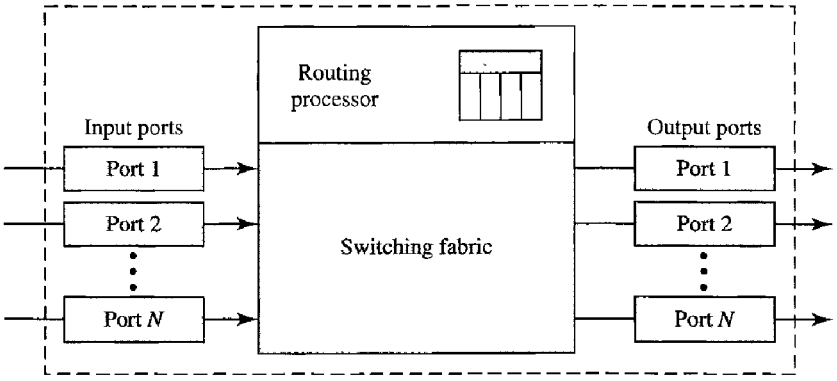
Figure 8.20 *Time-space-time switch*



Structure of Packet Switches

A switch used in a packet-switched network has a different structure from a switch used in a circuit-switched network. We can say that a packet switch has four components: **input ports, output ports,** the **routing processor,** and the **switching fabric,** as shown in Figure 8.21.

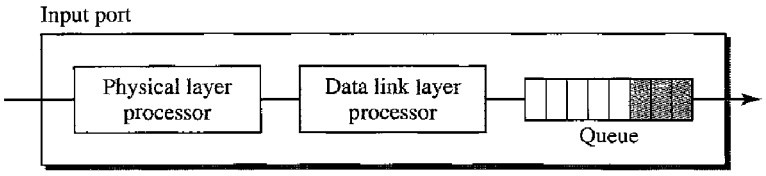
Figure 8.21 *Packet switch components*



Input Ports

An input port performs the physical and data link functions of the packet switch. The bits are constructed from the received signal. The packet is decapsulated from the frame. Errors are detected and corrected. The packet is now ready to be routed by the network layer. In addition to a physical layer processor and a data link processor, the input port has buffers (queues) to hold the packet before it is directed to the switching fabric. Figure 8.22 shows a schematic diagram of an input port.

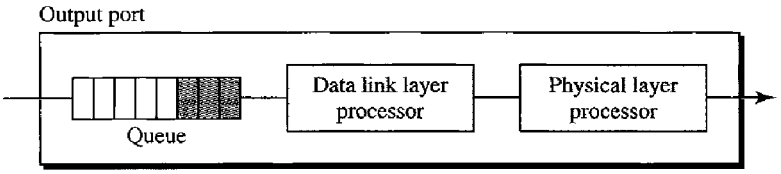
Figure 8.22 *Input port*



Output Port

The output port performs the same functions as the input port, but in the reverse order. First the outgoing packets are queued, then the packet is encapsulated in a frame, and finally the physical layer functions are applied to the frame to create the signal to be sent on the line. Figure 8.23 shows a schematic diagram of an output port.

Figure 8.23 *Output port*



Routing Processor

The routing processor performs the functions of the network layer. The destination address is used to find the address of the next hop and, at the same time, the output port number from which the packet is sent out. This activity is sometimes referred to as **table lookup** because the routing processor searches the routing table. In the newer packet switches, this function of the routing processor is being moved to the input ports to facilitate and expedite the process.

Switching Fabrics

The most difficult task in a packet switch is to move the packet from the input queue to the output queue. The speed with which this is done affects the size of the input/output queue and the overall delay in packet delivery. In the past, when a packet switch was actually a dedicated computer, the memory of the computer or a bus was used as the switching fabric. The input port stored the packet in memory; the output port retrieved the packet from memory. Today, packet switches are specialized mechanisms that use a variety of switching fabrics. We briefly discuss some of these fabrics here.

Crossbar Switch The simplest type of switching fabric is the crossbar switch, discussed in the previous section.

Banyan Switch A more realistic approach than the crossbar switch is the **banyan switch** (named after the banyan tree). A banyan switch is a multistage switch with

microswitches at each stage that route the packets based on the output port represented as a binary string. For n inputs and n outputs, we have $\log_2 n$ stages with $n/2$ microswitches at each stage. The first stage routes the packet based on the high-order bit of the binary string. The second stage routes the packet based on the second high-order bit, and so on. Figure 8.24 shows a banyan switch with eight inputs and eight outputs. The number of stages is $\log_2(8) = 3$.

Figure 8.24 *A banyan switch*

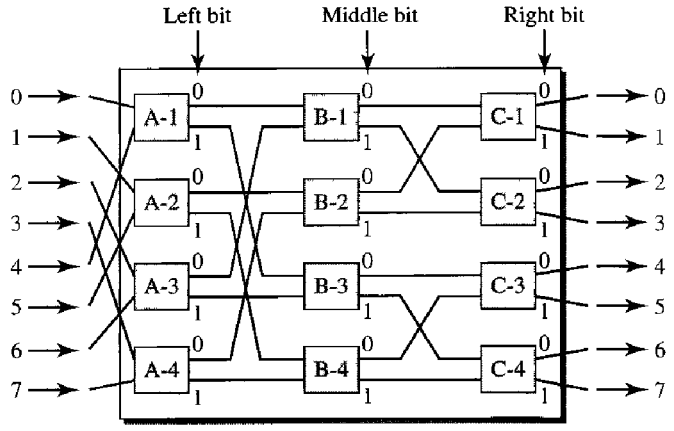
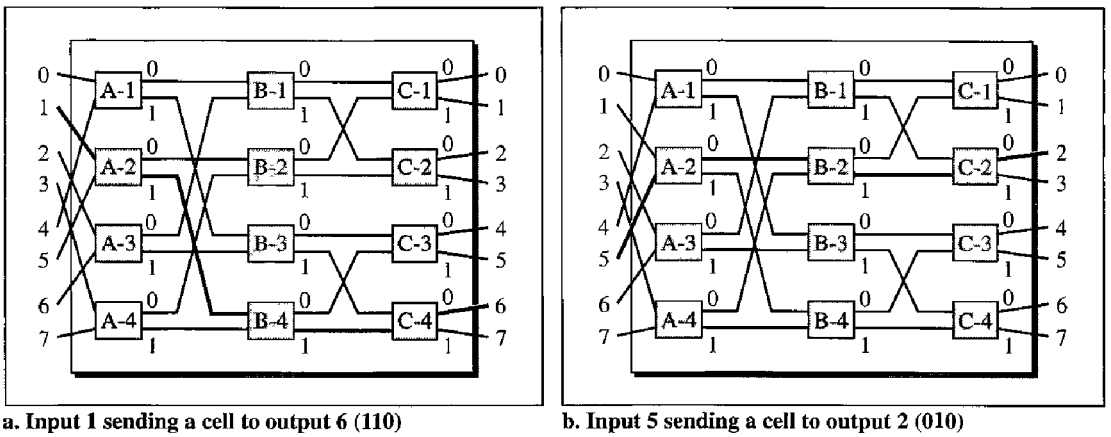


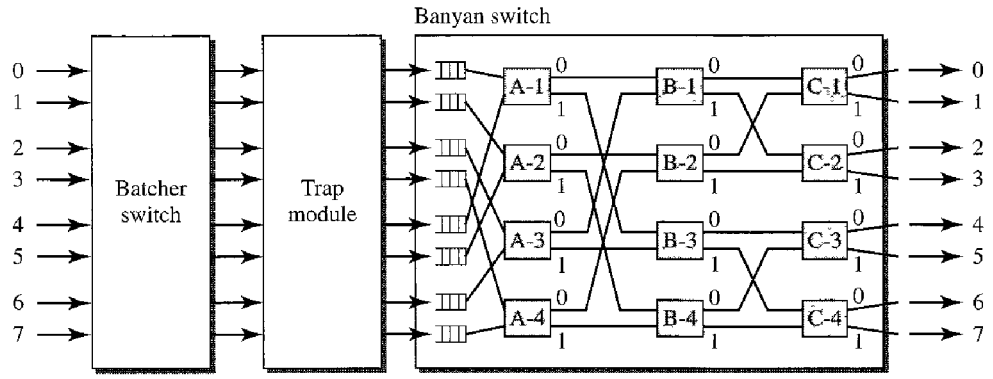
Figure 8.25 shows the operation. In part a, a packet has arrived at input port 1 and must go to output port 6 (110 in binary). The first microswitch (A-2) routes the packet based on the first bit (1), the second microswitch (B-4) routes the packet based on the second bit (1), and the third microswitch (C-4) routes the packet based on the third bit (0). In part b, a packet has arrived at input port 5 and must go to output port 2 (010 in binary). The first microswitch (A-2) routes the packet based on the first bit (0), the second microswitch (B-2) routes the packet based on the second bit (1), and the third microswitch (C-2) routes the packet based on the third bit (0).

Figure 8.25 *Examples of routing in a banyan switch*



a. Input 1 sending a cell to output 6 (110) **b. Input 5 sending a cell to output 2 (010)**

Figure 8.26 *Batcher-banyan switch*



Batcher-Banyan Switch The problem with the banyan switch is the possibility of internal collision even when two packets are not heading for the same output port. We can solve this problem by sorting the arriving packets based on their destination port.

K. E. Batcher designed a switch that comes before the banyan switch and sorts the incoming packets according to their final destinations. The combination is called the **Batcher-banyan switch.** The sorting switch uses hardware merging techniques, but we do not discuss the details here. Normally, another hardware module called a **trap** is added between the Batcher switch and the banyan switch (see Figure 8.26) The trap module prevents duplicate packets (packets with the same output destination) from passing to the banyan switch simultaneously. Only one packet for each destination is allowed at each tick; if there is more than one, they wait for the next tick.

8.5 RECOMMENDED READING

For more details about subjects discussed in this chapter, we recommend the following books. The items in brackets [. . .] refer to the reference list at the end of the text.

Books

Switching is discussed in Chapter 10 of [Sta04] and Chapters 4 and 7 of [GW04]. Circuit-switching is fully discussed in [BEL00].

8.6 KEY TERMS

banyan switch
Batcher-banyan switch
blocking
circuit switching
circuit-switched network
crossbar switch
crosspoint
data transfer phase
datagram
datagram network
end system
input port

multistage switch
output port
packet-switched network
routing processor
setup phase
space-division switching
switch
switching
switching fabric
table lookup
teardown phase
time-division switching
time-slot interchange (TSI)
time-space-time (TST) switch
trap
virtual-circuit identifier (VCI)
virtual-circuit network

8.7 SUMMARY

- A switched network consists of a series of interlinked nodes, called switches. Traditionally, three methods of switching have been important: circuit switching, packet switching, and message switching.
- We can divide today's networks into three broad categories: circuit-switched networks, packet-switched networks, and message-switched. Packet-switched networks can also be divided into two subcategories: virtual-circuit networks and datagram networks
- A circuit-switched network is made of a set of switches connected by physical links, in which each link is divided into n channels. Circuit switching takes place at the physical layer. In circuit switching, the resources need to be reserved during the setup phase; the resources remain dedicated for the entire duration of data transfer phase until the teardown phase.
- In packet switching, there is no resource allocation for a packet. This means that there is no reserved bandwidth on the links, and there is no scheduled processing time for each packet. Resources are allocated on demand.
- In a datagram network, each packet is treated independently of all others. Packets in this approach are referred to as datagrams. There are no setup or teardown phases.
- A virtual-circuit network is a cross between a circuit-switched network and a datagram network. It has some characteristics of both.
- Circuit switching uses either of two technologies: the space-division switch or the time-division switch.
- A switch in a packet-switched network has a different structure from a switch used in a circuit-switched network.We can say that a packet switch has four types of components: input ports, output ports, a routing processor, and switching fabric.

8.8 PRACTICE SET

Review Questions

1. Describe the need for switching and define a switch.
2. List the three traditional switching methods. What are the most common today?

3. What are the two approaches to packet-switching?
4. Compare and contrast a circuit-switched network and a packet-switched network.
5. What is the role of the address field in a packet traveling through a datagram network?
6. What is the role of the address field in a packet traveling through a virtual-circuit network?
7. Compare space-division and time-division switches.
8. What is TSI and its role in a time-division switching?
9. Define blocking in a switched network.
10. List four major components of a packet switch and their functions.

Exercises

11. A path in a digital circuit-switched network has a data rate of 1 Mbps. The exchange of 1000 bits is required for the setup and teardown phases. The distance between two parties is 5000 km. Answer the following questions if the propagataion speed is 2×10^8 m:
    a. What is the total delay if 1000 bits of data are exchanged during the data transfer phase?
    b. What is the total delay if 100,000 bits of data are exchanged during the data transfer phase?
    c. What is the total delay if 1,000,000 bits of data are exchanged during the data transfer phase?
    d. Find the delay per 1000 bits of data for each of the above cases and compare them. What can you infer?
12. Five equal-size datagrams belonging to the same message leave for the destination one after another. However, they travel through different paths as shown in Table 8.1.

Table 8.1 *Exercise 12*

Datagram	*Path Length*	*Visited Switches*
1	3200 Km	1, 3, 5
2	11,700 Km	1, 2, 5
3	12,200 Km	1, 2, 3, 5
4	10,200 Km	1, 4, 5
5	10,700 Km	1, 4, 3, 5

We assume that the delay for each switch (including waiting and processing) is 3, 10, 20, 7, and 20 ms respectively. Assuming that the propagation speed is 2×10^8 m, find the order the datagrams arrive at the destination and the delay for each. Ignore any other delays in transmission.

13. Transmission of information in any network involves end-to-end addressing and sometimes local addressing (such as VCI). Table 8.2 shows the types of networks and the addressing mechanism used in each of them.

Table 8.2 *Exercise 13*

Network	*Setup*	*Data Transfer*	*Teardown*
Circuit-switched	End-to-end		End-to-end
Datagram		End-to-end	
Virtual-circuit	End-to-end	Local	End-to-end

Answer the following questions:

a. Why does a circuit-switched network need end-to-end addressing during the setup and teardown phases? Why are no addresses needed during the data transfer phase for this type of network?

b. Why does a datagram network need only end-to-end addressing during the data transfer phase, but no addressing during the setup and teardown phases?

c. Why does a virtual-circuit network need addresses during all three phases?

14. We mentioned that two types of networks, datagram and virtual-circuit, need a routing or switching table to find the output port from which the information belonging to a destination should be sent out, but a circuit-switched network has no need for such a table. Give the reason for this difference.

15. An entry in the switching table of a virtual-circuit network is normally created during the setup phase and deleted during the teardown phase. In other words, the entries in this type of network reflect the current connections, the activity in the network. In contrast, the entries in a routing table of a datagram network do not depend on the current connections; they show the configuration of the network and how any packet should be routed to a final destination. The entries may remain the same even if there is no activity in the network. The routing tables, however, are updated if there are changes in the network. Can you explain the reason for these two different characteristics? Can we say that a virtual-circuit is a *connection-oriented* network and a datagram network is a *connectionless* network because of the above characteristics?

16. The minimum number of columns in a datagram network is two; the minimum number of columns in a virtual-circuit network is four. Can you explain the reason? Is the difference related to the type of addresses carried in the packets of each network?

17. Figure 8.27 shows a switch (router) in a datagram network.

    Find the output port for packets with the following destination addresses:
    Packet 1: 7176
    Packet 2: 1233
    Packet 3: 8766
    Packet 4: 9144

18. Figure 8.28 shows a switch in a virtual circuit network.

Figure 8.27 *Exercise 17*

Destination address	Output port
1233	3
1456	2
3255	1
4470	4
7176	2
8766	3
9144	2

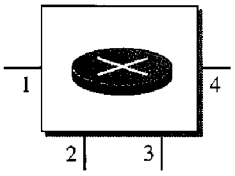
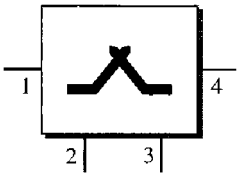


Figure 8.28 *Exercise 18*

Incoming		Outgoing	
Port	VCI	Port	VCI
1	14	3	22
2	71	4	41
2	92	1	45
3	58	2	43
3	78	2	70
4	56	3	11



Find the output port and the output VCI for packets with the following input port and input VCI addresses:
Packet 1: 3, 78
Packet 2: 2, 92
Packet 3: 4, 56
Packet 4: 2, 71

19. Answer the following questions:
    a. Can a routing table in a datagram network have two entries with the same destination address? Explain.
    b. Can a switching table in a virtual-circuit network have two entries with the same input port number? With the same output port number? With the same incoming VCIs? With the same outgoing VCIs? With the same incoming values (port, VCI)? With the same outgoing values (port, VCI)?
20. It is obvious that a router or a switch needs to do searching to find information in the corresponding table. The searching in a routing table for a datagram network is based on the destination address; the searching in a switching table in a virtual-circuit network is based on the combination of incoming port and incoming VCI. Explain the reason and define how these tables must be ordered (sorted) based on these values.
21. Consider an $n \times k$ crossbar switch with n inputs and k outputs.
    a. Can we say that switch acts as a multiplexer if $n > k$?
    b. Can we say that switch acts as a demultiplexer if $n < k$?

22. We need a three-stage space-division switch with N = 100. We use 10 crossbars at the first and third stages and 4 crossbars at the middle stage.
    a. Draw the configuration diagram.
    b. Calculate the total number of crosspoints.
    c. Find the possible number of simultaneous connections.
    d. Find the possible number of simultaneous connections if we use one single crossbar (100×100).
    e. Find the blocking factor, the ratio of the number of connections in c and in d.
23. Repeat Exercise 22 if we use 6 crossbars at the middle stage.
24. Redesign the configuration of Exercise 22 using the Clos criteria.
25. We need to have a space-division switch with 1000 inputs and outputs. What is the total number of crosspoints in each of the following cases?
    a. Using one single crossbar.
    b. Using a multi-stage switch based on the Clos criteria
26. We need a three-stage time-space-time switch with N = 100. We use 10 TSIs at the first and third stages and 4 crossbars at the middle stage.
    a. Draw the configuration diagram.
    b. Calculate the total number of crosspoints.
    c. Calculate the total number of memory locations we need for the TSIs.

CHAPTER 9

Using Telephone and Cable Networks for Data Transmission

Telephone networks were originally created to provide voice communication. The need to communicate digital data resulted in the invention of the dial-up modem. With the advent of the Internet came the need for high-speed downloading and uploading; the modem was just too slow. The telephone companies added a new technology, the *digital subscriber line* (DSL). Although dial-up modems still exist in many places all over the world, DSL provides much faster access to the Internet through the telephone network. In this chapter, we first discuss the basic structure of the telephone network. We then see how dial-up modems and DSL technology use these networks to access the Internet.

Cable networks were originally created to provide access to TV programs for those subscribers who had no reception because of natural obstructions such as mountains. Later the cable network became popular with people who just wanted a better signal. In addition, cable networks enabled access to remote broadcasting stations via microwave connections. Cable TV also found a good market in Internet access provision using some of the channels originally designed for video. After discussing the basic structure of cable networks, we discuss how cable modems can provide a high-speed connection to the Internet.

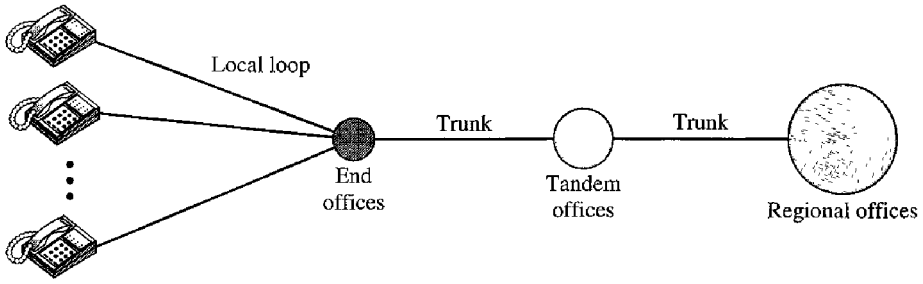
9.1 TELEPHONE NETWORK

Telephone networks use circuit switching. The telephone network had its beginnings in the late 1800s. The entire network, which is referred to as the **plain old telephone system (POTS),** was originally an analog system using analog signals to transmit voice. With the advent of the computer era, the network, in the 1980s, began to carry data in addition to voice. During the last decade, the telephone network has undergone many technical changes. The network is now digital as well as analog.

Major Components

The telephone network, as shown in Figure 9.1, is made of three major components: local loops, trunks, and switching offices. The telephone network has several levels of switching offices such as **end offices, tandem offices,** and **regional offices.**

Figure 9.1 *A telephone system*



Local Loops

One component of the telephone network is the **local loop,** a twisted-pair cable that connects the subscriber telephone to the nearest end office or local central office. The local loop, when used for voice, has a bandwidth of 4000 Hz (4 kHz). It is interesting to examine the telephone number associated with each local loop. The first three digits of a local telephone number define the office, and the next four digits define the local loop number.

Trunks

Trunks are transmission media that handle the communication between offices. A trunk normally handles hundreds or thousands of connections through multiplexing. Transmission is usually through optical fibers or satellite links.

Switching Offices

To avoid having a permanent physical link between any two subscribers, the telephone company has switches located in a **switching office.** A switch connects several local loops or trunks and allows a connection between different subscribers.

LATAs

After the divestiture of 1984 (see Appendix E), the United States was divided into more than 200 **local-access transport areas (LATAs).** The number of LATAs has increased since then. A LATA can be a small or large metropolitan area. A small state may have one single LATA; a large state may have several LATAs. A LATA boundary may overlap the boundary of a state; part of a LATA can be in one state, part in another state.

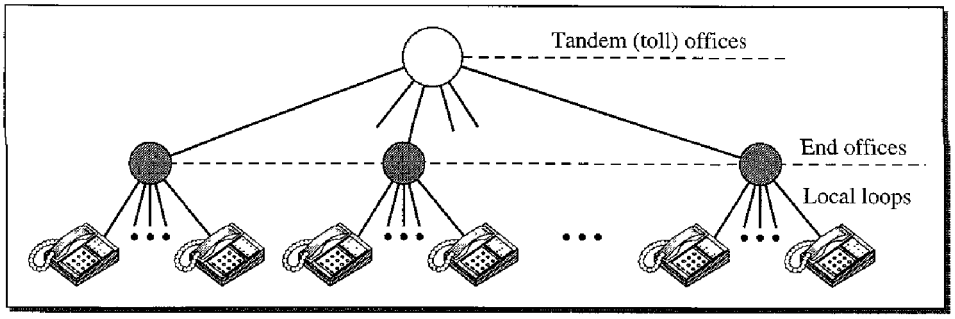
Intra-LATA Services

The services offered by the **common carriers** (telephone companies) inside a LATA are called *intra-LATA* services. The carrier that handles these services is called a **local exchange carrier (LEC).** Before the Telecommunications Act of 1996 (see Appendix E), intra-LATA services were granted to one single carrier. This was a monopoly. After 1996, more than one carrier could provide services inside a LATA. The carrier that provided services before 1996 owns the cabling system (local loops) and is called the **incumbent local exchange carrier (ILEC).** The new carriers that can provide services are called **competitive local exchange carriers (CLECs).** To avoid the costs of new cabling, it

was agreed that the ILECs would continue to provide the main services, and the CLECs would provide other services such as mobile telephone service, toll calls inside a LATA, and so on. Figure 9.2 shows a LATA and switching offices.

Intra-LATA services are provided by local exchange carriers. Since 1996, there are two types of LECs: incumbent local exchange carriers and competitive local exchange carriers.

Figure 9.2 *Switching offices in a LATA*



Communication inside a LATA is handled by end switches and tandem switches. A call that can be completed by using only end offices is considered toll-free. A call that has to go through a tandem office (intra-LATA toll office) is charged.

Inter-LATA Services

The services between LATAs are handled by **interexchange carriers (IXCs).** These carriers, sometimes called **long-distance companies,** provide communication services between two customers in different LATAs. After the act of 1996 (see Appendix E), these services can be provided by any carrier, including those involved in intra-LATA services. The field is wide open. Carriers providing inter-LATA services include AT&T, MCI, WorldCom, Sprint, and Verizon.

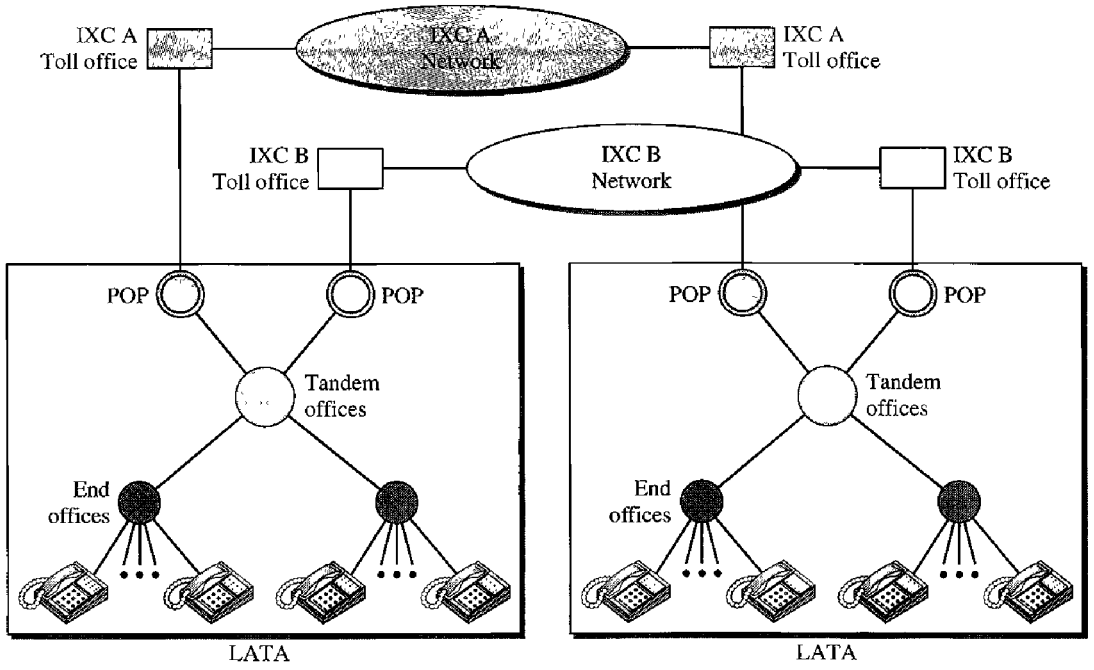
The IXCs are long-distance carriers that provide general data communications services including telephone service. A telephone call going through an IXC is normally digitized, with the carriers using several types of networks to provide service.

Points of Presence

As we discussed, intra-LATA services can be provided by several LECs (one ILEC and possibly more than one CLEC). We also said that inter-LATA services can be provided by several IXCs. How do these carriers interact with one another? The answer is, via a switching office called a **point of presence (POP).** Each IXC that wants to provide inter-LATA services in a LATA must have a POP in that LATA. The LECs that provide services inside the LATA must provide connections so that every subscriber can have access to all POPs. Figure 9.3 illustrates the concept.

A subscriber who needs to make a connection with another subscriber is connected first to an end switch and then, either directly or through a tandem switch, to a POP. The

Figure 9.3 *Point of presences (POPs)*



call now goes from the POP of an IXC (the one the subscriber has chosen) in the source LATA to the POP of the same IXC in the destination LATA. The call is passed through the toll office of the IXC and is carried through the network provided by the IXC.

Signaling

The telephone network, at its beginning, used a circuit-switched network with dedicated links (multiplexing had not yet been invented) to transfer voice communication. As we saw in Chapter 8, a circuit-switched network needs the setup and teardown phases to establish and terminate paths between the two communicating parties. In the beginning, this task was performed by human operators. The operator room was a center to which all subscribers were connected. A subscriber who wished to talk to another subscriber picked up the receiver (off-hook) and rang the operator. The operator, after listening to the caller and getting the identifier of the called party, connected the two by using a wire with two plugs inserted into the corresponding two jacks. A dedicated circuit was created in this way. One of the parties, after the conversation ended, informed the operator to disconnect the circuit. This type of signaling is called **in-band signaling** because the same circuit can be used for both signaling and voice communication.

Later, the signaling system became automatic. Rotary telephones were invented that sent a digital signal defining each digit in a multidigit telephone number. The switches in the telephone companies used the digital signals to create a connection between the caller and the called parties. Both in-band and **out-of-band signaling** were used. In in-band signaling, the 4-kHz voice channel was also used to provide signaling. In out-of-band signaling, a portion of the voice channel bandwidth was used for signaling; the voice bandwidth and the signaling bandwidth were separate.

As telephone networks evolved into a complex network, the functionality of the signaling system increased. The signaling system was required to perform other tasks such as

1. Providing dial tone, ring tone, and busy tone
2. Transferring telephone numbers between offices
3. Maintaining and monitoring the call
4. Keeping billing information
5. Maintaining and monitoring the status of the telephone network equipment
6. Providing other functions such as caller ID, voice mail, and so on

These complex tasks resulted in the provision of a separate network for signaling. This means that a telephone network today can be thought of as two networks: a signaling network and a data transfer network.

The tasks of data transfer and signaling are separated in modern telephone networks: data transfer is done by one network, signaling by another.

However, we need to emphasize a point here. Although the two networks are separate, this does not mean that there are separate physical links everywhere; the two networks may use separate channels of the same link in parts of the system.

Data Transfer Network

The data transfer network that can carry multimedia information today is, for the most part, a circuit-switched network, although it can also be a packet-switched network. This network follows the same type of protocols and model as other networks discussed in this book.

Signaling Network

The signaling network, which is our main concern in this section, is a packet-switched network involving the layers similar to those in the OSI model or Internet model, discussed in Chapter 2. The nature of signaling makes it more suited to a packet-switching network with different layers. For example, the information needed to convey a telephone address can easily be encapsulated in a packet with all the error control and addressing information. Figure 9.4 shows a simplified situation of a telephone network in which the two networks are separated.

The user telephone or computer is connected to the **signal points (SPs).** The link between the telephone set and SP is common for the two networks. The signaling network uses nodes called **signal transport ports (STPs)** that receive and forward signaling messages. The signaling network also includes a **service control point (SCP)** that controls the whole operation of the network. Other systems such as a database center may be included to provide stored information about the entire signaling network.

Signaling System Seven (SS7)

The protocol that is used in the signaling network is called **Signaling System Seven (SS7).** It is very similar to the five-layer Internet model we saw in Chapter 2, but the layers have different names, as shown in Figure 9.5.

Figure 9.4 *Data transfer and signaling networks*

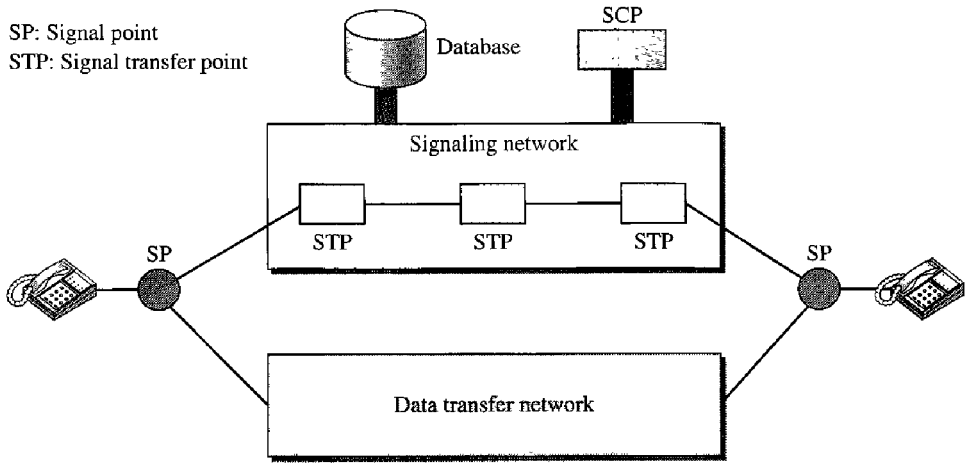
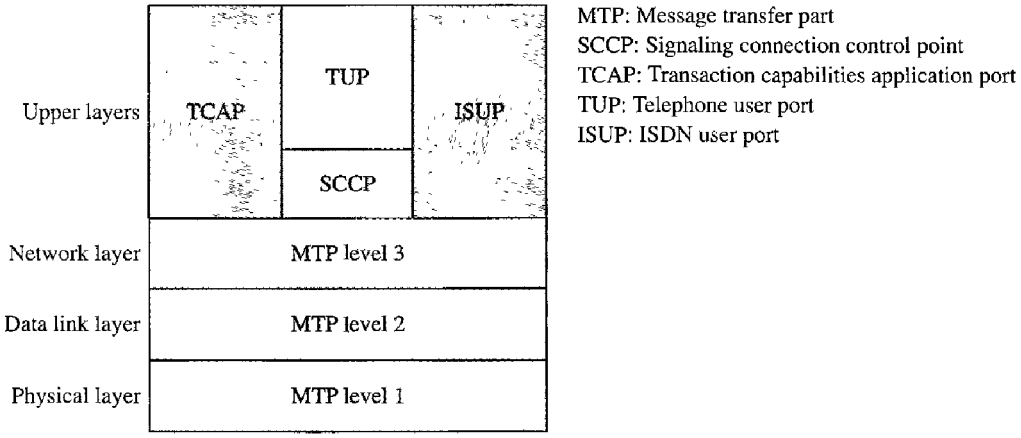


Figure 9.5 *Layers in SS7*



Physical Layer: MTP Level 1 The physical layer in SS7 called **message transport part (MTP) level** 1 uses several physical layer specifications such as T-1 (1.544 Mbps) and DC0 (64 kbps).

Data Link Layer: MTP Level 2 The MTP level 2 layer provides typical data link layer services such as packetizing, using source and destination address in the packet header, and CRC for error checking.

Network Layer: MTP Level 3 The MTP level 3 layer provides end-to-end connectivity by using the datagram approach to switching. Routers and switches route the signal packets from the source to the destination.

Transport Layer: SCCP The **signaling connection control point (SCCP)** is used for special services such as 800-call processing.

Upper Layers: TUP, TCAP, and ISUP There are three protocols at the upper layers. **Telephone user port (TUP)** is responsible for setting up voice calls. It receives the dialed

digits and routes the calls. **Transaction capabilities application port (TCAP)** provides remote calls that let an application program on a computer invoke a procedure on another computer. **ISDN user port (ISUP)** can replace TUP to provide services similar to those of an ISDN network.

Services Provided by Telephone Networks

Telephone companies provide two types of services: analog and digital.

Analog Services

In the beginning, telephone companies provided their subscribers with analog services. These services still continue today. We can categorize these services as either **analog switched services** or **analog leased services.**

Analog Switched Services This is the familiar dial-up service most often encountered when a home telephone is used. The signal on a local loop is analog, and the bandwidth is usually between 0 and 4000 Hz. A local call service is normally provided for a flat monthly rate, although in some LATAs, the carrier charges for each call or a set of calls. The rationale for a non flat-rate charge is to provide cheaper service for those customers who do not make many calls. A toll call can be intra-LATA or inter-LATA. If the LATA is geographically large, a call may go through a tandem office (toll office) and the subscriber will pay a fee for the call. The inter-LATA calls are long-distance calls and are charged as such.

Another service is called 800 service. If a subscriber (normally an organization) needs to provide free connections for other subscribers (normally customers), it can request the **800 service.** In this case, the call is free for the caller, but it is paid by the callee. An organization uses this service to encourage customers to call. The rate is less expensive than that for a normal long-distance call.

The **wide-area telephone service (WATS)** is the opposite of the 800 service. The latter are inbound calls paid by the organization; the former are outbound calls paid by the organization. This service is a less expensive alternative to regular toll calls; charges are based on the number of calls. The service can be specified as outbound calls to the same state, to several states, or to the whole country, with rates charged accordingly.

The **900 services** are like the 800 service, in that they are inbound calls to a subscriber. However, unlike the 800 service, the call is paid by the caller and is normally much more expensive than a normal long-distance call. The reason is that the carrier charges *two* fees: the first is the long-distance toll, and the second is the fee paid to the callee for each call.

Analog Leased Service An **analog leased service** offers customers the opportunity to lease a line, sometimes called a *dedicated line,* that is permanently connected to another customer. Although the connection still passes through the switches in the telephone network, subscribers experience it as a single line because the switch is always closed; no dialing is needed.

Digital Services

Recently telephone companies began offering **digital services** to their subscribers. Digital services are less sensitive than analog services to noise and other forms of interference.

The two most common digital services are switched/56 service and **digital data service (DDS).** We already discussed high-speed digital services—the T lines—in Chapter 6. We discuss the other services in this chapter.

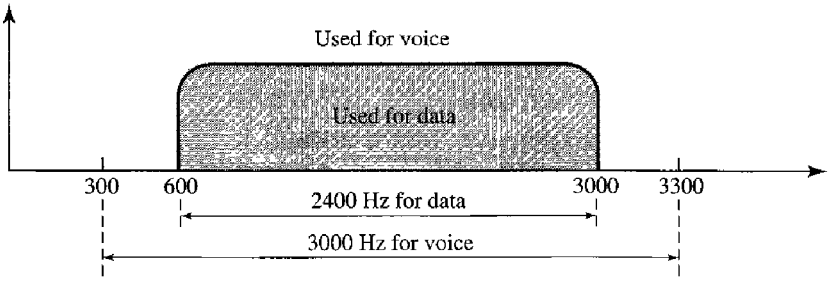
Switched/56 Service **Switched/56 service** is the digital version of an analog switched line. It is a switched digital service that allows data rates of up to 56 kbps. To communicate through this service, both parties must subscribe. A caller with normal telephone service cannot connect to a telephone or computer with switched/56 service even if the caller is using a modem. On the whole, digital and analog services represent two completely different domains for the telephone companies. Because the line in a switched/56 service is already digital, subscribers do not need modems to transmit digital data. However, they do need another device called a **digital service unit (DSU).**

Digital Data Service **Digital data service (DDS)** is the digital version of an analog leased line; it is a digital leased line with a maximum data rate of 64 kbps.

9.2 DIAL-UP MODEMS

Traditional telephone lines can carry frequencies between 300 and 3300 Hz, giving them a bandwidth of 3000 Hz. All this range is used for transmitting voice, where a great deal of interference and distortion can be accepted without loss of intelligibility. As we have seen, however, data signals require a higher degree of accuracy to ensure integrity. For safety's sake, therefore, the edges of this range are not used for data communications. In general, we can say that the signal bandwidth must be smaller than the cable bandwidth. The effective bandwidth of a telephone line being used for data transmission is 2400 Hz, covering the range from 600 to 3000 Hz. Note that today some telephone lines are capable of handling greater bandwidth than traditional lines. However, modem design is still based on traditional capability (see Figure 9.6).

Figure 9.6 *Telephone line bandwidth*

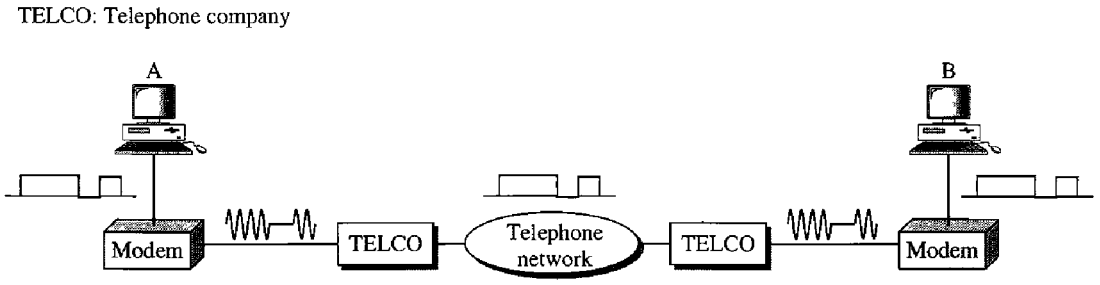


The term **modem** is a composite word that refers to the two functional entities that make up the device: a signal *mo*dulator and a signal *demo*dulator. A **modulator** creates a bandpass analog signal from binary data. A **demodulator** recovers the binary data from the modulated signal.

***Modem* stands for modulator/demodulator.**

Figure 9.7 shows the relationship of modems to a communications link. The computer on the left sends a digital signal to the modulator portion of the modem; the data are sent as an analog signal on the telephone lines. The modem on the right receives the analog signal, demodulates it through its demodulator, and delivers data to the computer on the right. The communication can be bidirectional, which means the computer on the right can simultaneously send data to the computer on the left, using the same modulation/demodulation processes.

Figure 9.7 *Modulation/demodulation*



Modem Standards

Today, many of the most popular modems available are based on the **V-series** standards published by the ITU-T. We discuss just the most recent series.

V.32 and V.32bis

The **V.32** modem uses a combined modulation and encoding technique called **trellis-coded modulation.** Trellis is essentially QAM plus a redundant bit. The data stream is divided into 4-bit sections. Instead of a quadbit (4-bit pattern), however, a *pentabit* (5-bit pattern) is transmitted. The value of the extra bit is calculated from the values of the data bits. The extra bit is used for error detection.

The V.32 calls for 32-QAM with a baud rate of 2400. Because only 4 bits of each pentabit represent data, the resulting data rate is $4 \times 2400 = 9600$ bps. The constellation diagram and bandwidth are shown in Figure 9.8.

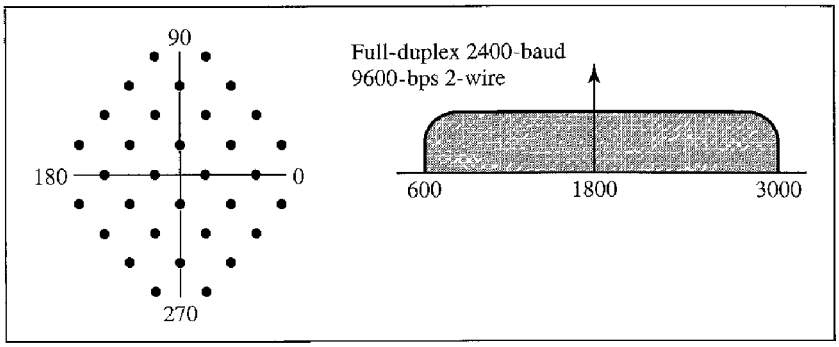
The **V.32bis** modem was the first of the ITU-T standards to support 14,400-bps transmission. The V.32bis uses 128-QAM transmission (7 bits/baud with 1 bit for error control) at a rate of 2400 baud ($2400 \times 6 = 14{,}400$ bps).

An additional enhancement provided by V.32bis is the inclusion of an automatic fall-back and fall-forward feature that enables the modem to adjust its speed upward or downward depending on the quality of the line or signal. The constellation diagram and bandwidth are also shown in Figure 9.8.

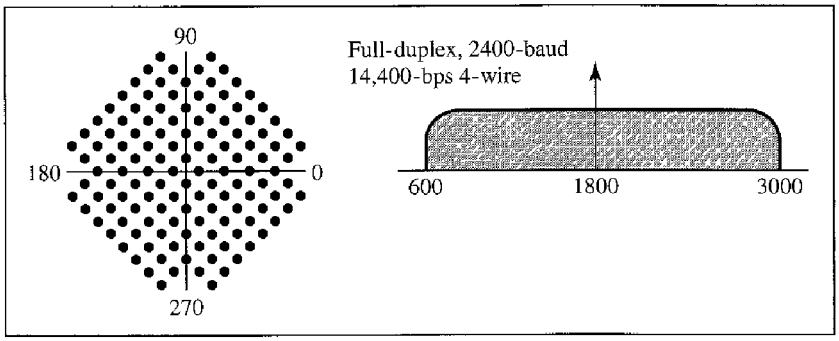
V.34bis

The **V.34bis** modem provides a bit rate of 28,800 with a 960-point constellation and a bit rate of 33,600 bps with a 1664-point constellation.

Figure 9.8 *The V.32 and V.32bis constellation and bandwidth*



a. Constellation and bandwidth for V.32



b. Constellation and bandwidth for V.32bis

V.90

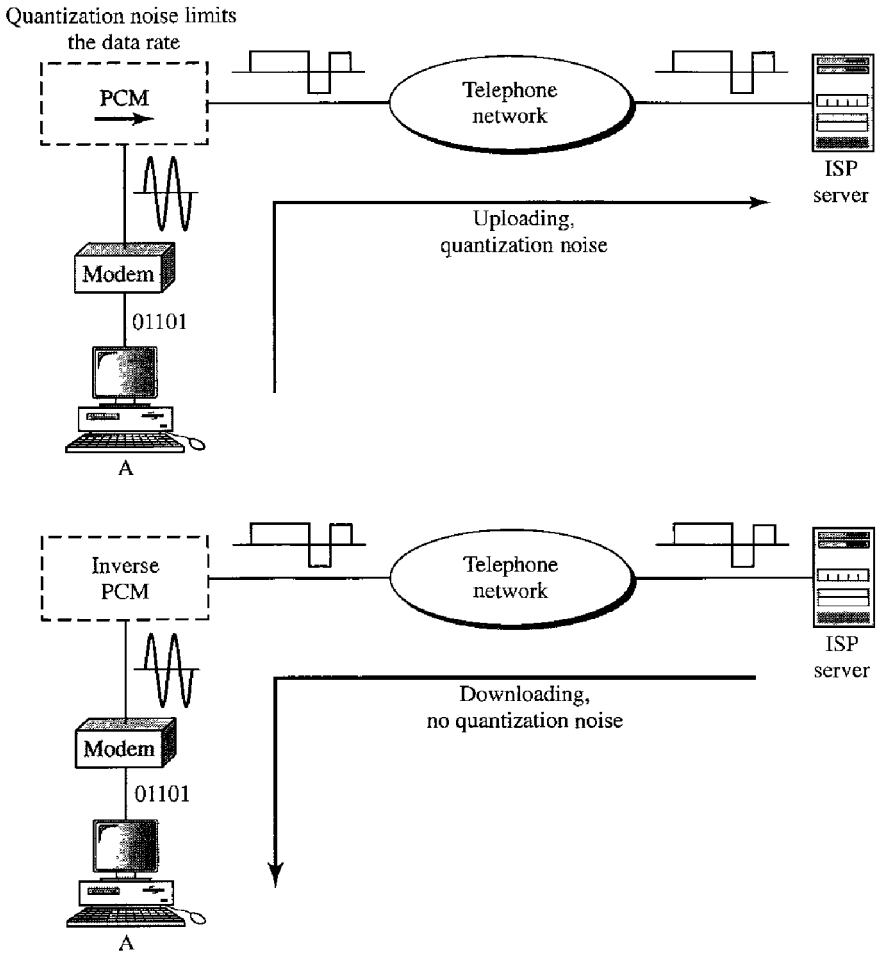
Traditional modems have a data rate limitation of 33.6 kbps, as determined by the Shannon capacity (see Chapter 3). However, **V.90** modems with a bit rate of 56,000 bps are available; these are called **56K modems.** These modems may be used only if one party is using digital signaling (such as through an Internet provider). They are asymmetric in that the downloading rate (flow of data from the Internet service provider to the PC) is a maximum of 56 kbps, while the uploading rate (flow of data from the PC to the Internet provider) can be a maximum of 33.6 kbps. Do these modems violate the Shannon capacity principle? No, in the downstream direction, the SNR ratio is higher because there is no quantization error (see Figure 9.9).

In **uploading,** the analog signal must still be sampled at the switching station. In this direction, quantization noise (as we saw in Chapter 4) is introduced into the signal, which reduces the SNR ratio and limits the rate to 33.6 kbps.

However, there is no sampling in the **downloading.** The signal is not affected by quantization noise and not subject to the Shannon capacity limitation. The maximum data rate in the uploading direction is still 33.6 kbps, but the data rate in the downloading direction is now 56 kbps.

One may wonder how we arrive at the 56-kbps figure. The telephone companies sample 8000 times per second with 8 bits per sample. One of the bits in each sample is used for control purposes, which means each sample is 7 bits. The rate is therefore 8000×7, or 56,000 bps or 56 kbps.

Figure 9.9 *Uploading and downloading in 56K modems*



V.92

The standard above V.90 is called **V.92.** These modems can adjust their speed, and if the noise allows, they can upload data at the rate of 48 kbps. The downloading rate is still 56 kbps. The modem has additional features. For example, the modem can interrupt the Internet connection when there is an incoming call if the line has call-waiting service.

9.3 DIGITAL SUBSCRIBER LINE

After traditional modems reached their peak data rate, telephone companies developed another technology, DSL, to provide higher-speed access to the Internet. **Digital subscriber line (DSL)** technology is one of the most promising for supporting high-speed digital communication over the existing local loops. DSL technology is a set of technologies, each differing in the first letter (ADSL, VDSL, HDSL, and SDSL). The set is often referred to as *x*DSL, where *x* can be replaced by A, V, H, or S.

ADSL

The first technology in the set is **asymmetric DSL (ADSL).** ADSL, like a 56K modem, provides higher speed (bit rate) in the downstream direction (from the Internet to the resident) than in the upstream direction (from the resident to the Internet). That is the reason it is called asymmetric. Unlike the asymmetry in 56K modems, the designers of ADSL specifically divided the available bandwidth of the local loop unevenly for the residential customer. The service is not suitable for business customers who need a large bandwidth in both directions.

ADSL is an asymmetric communication technology designed for residential users; it is not suitable for businesses.

Using Existing Local Loops

One interesting point is that ADSL uses the existing local loops. But how does ADSL reach a data rate that was never achieved with traditional modems? The answer is that the twisted-pair local loop is actually capable of handling bandwidths up to 1.1 MHz, but the filter installed at the end office of the telephone company where each local loop terminates limits the bandwidth to 4 kHz (sufficient for voice communication). If the filter is removed, however, the entire 1.1 MHz is available for data and voice communications.

The existing local loops can handle bandwidths up to 1.1 MHz.

Adaptive Technology

Unfortunately, 1.1 MHz is just the theoretical bandwidth of the local loop. Factors such as the distance between the residence and the switching office, the size of the cable, the signaling used, and so on affect the bandwidth. The designers of ADSL technology were aware of this problem and used an adaptive technology that tests the condition and bandwidth availability of the line before settling on a data rate. The data rate of ADSL is not fixed; it changes based on the condition and type of the local loop cable.

ADSL is an adaptive technology. The system uses a data rate based on the condition of the local loop line.

Discrete Multitone Technique

The modulation technique that has become standard for ADSL is called the **discrete multitone technique (DMT)** which combines QAM and FDM. There is no set way that the bandwidth of a system is divided. Each system can decide on its bandwidth division. Typically, an available bandwidth of 1.104 MHz is divided into 256 channels. Each channel uses a bandwidth of 4.312 kHz, as shown in Figure 9.10. Figure 9.11 shows how the bandwidth can be divided into the following:

- **Voice.** Channel 0 is reserved for voice communication.
- **Idle.** Channels 1 to 5 are not used and provide a gap between voice and data communication.

Figure 9.10 *Discrete multitone technique*

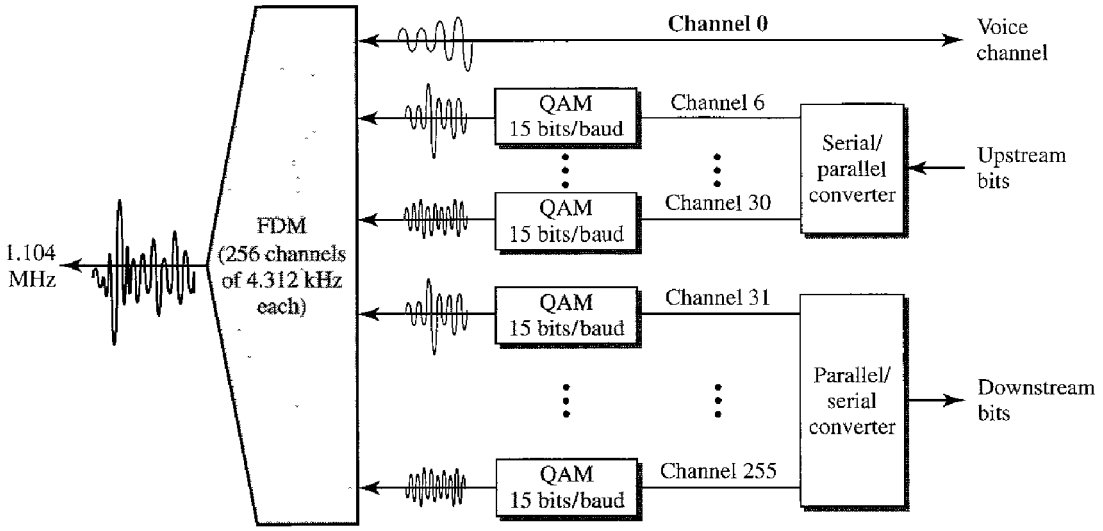
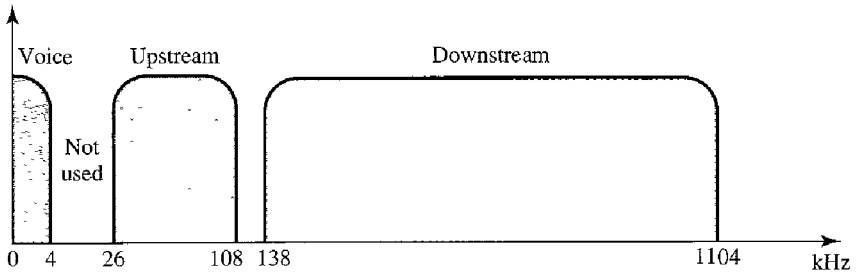


Figure 9.11 *Bandwidth division in ADSL*



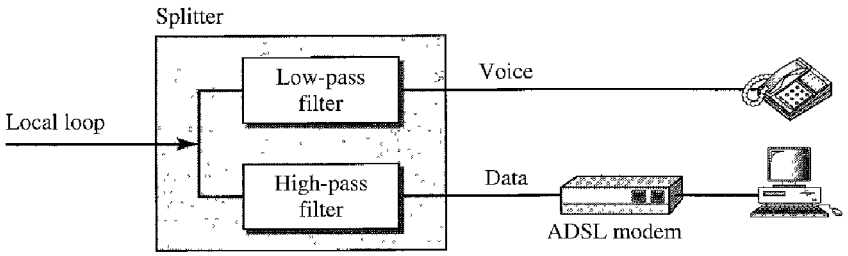
- **Upstream data and control.** Channels 6 to 30 (25 channels) are used for upstream data transfer and control. One channel is for control, and 24 channels are for data transfer. If there are 24 channels, each using 4 kHz (out of 4.312 kHz available) with QAM modulation, we have $24 \times 4000 \times 15$, or a 1.44-Mbps bandwidth, in the upstream direction. However, the data rate is normally below 500 kbps because some of the carriers are deleted at frequencies where the noise level is large. In other words, some of channels may be unused.
- **Downstream data and control.** Channels 31 to 255 (225 channels) are used for downstream data transfer and control. One channel is for control, and 224 channels are for data. If there are 224 channels, we can achieve up to $224 \times 4000 \times 15$, or 13.4 Mbps. However, the data rate is normally below 8 Mbps because some of the carriers are deleted at frequencies where the noise level is large. In other words, some of channels may be unused.

Customer Site: ADSL Modem

Figure 9.12 shows an **ADSL modem** installed at a customer's site. The local loop connects to a splitter which separates voice and data communications. The ADSL

modem modulates and demodulates the data, using DMT, and creates downstream and upstream channels.

Figure 9.12 *ADSL modem*

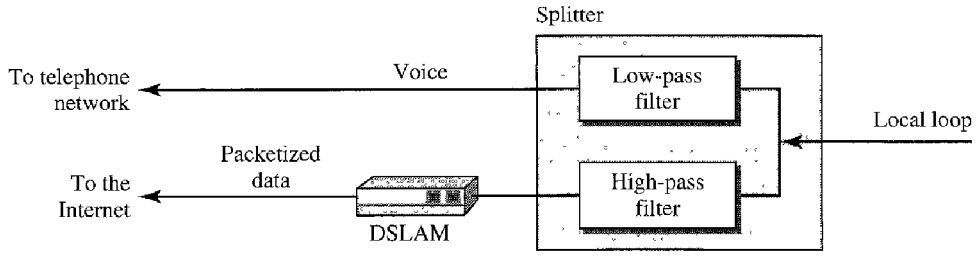


Note that the splitter needs to be installed at the customer's premises, normally by a technician from the telephone company. The voice line can use the existing telephone wiring in the house, but the data line needs to be installed by a professional. All this makes the ADSL line expensive. We will see that there is an alternative technology, Universal ADSL (or ADSL Lite).

Telephone Company Site: DSLAM

At the telephone company site, the situation is different. Instead of an ADSL modem, a device called a **digital subscriber line access multiplexer (DSLAM)** is installed that functions similarly. In addition, it packetizes the data to be sent to the Internet (ISP server). Figure 9.13 shows the configuration.

Figure 9.13 *DSLAM*



ADSL Lite

The installation of splitters at the border of the premises and the new wiring for the data line can be expensive and impractical enough to dissuade most subscribers. A new version of ADSL technology called **ADSL Lite** (or Universal ADSL or splitterless ADSL) is available for these subscribers. This technology allows an ASDL Lite modem to be plugged directly into a telephone jack and connected to the computer. The splitting is done at the telephone company. ADSL Lite uses 256 DMT carriers with 8-bit modulation

(instead of 15-bit). However, some of the carriers may not be available because errors created by the voice signal might mingle with them. It can provide a maximum downstream data rate of 1.5 Mbps and an upstream data rate of 512 kbps.

HDSL

The **high-bit-rate digital subscriber line (HDSL)** was designed as an alternative to the T-1 line (1.544 Mbps). The T-1 line uses alternate mark inversion (AMI) encoding, which is very susceptible to attenuation at high frequencies. This limits the length of a T-1 line to 3200 ft (1 km). For longer distances, a repeater is necessary, which means increased costs.

HDSL uses 2B1Q encoding (see Chapter 4), which is less susceptible to attenuation. A data rate of 1.544 Mbps (sometimes up to 2 Mbps) can be achieved without repeaters up to a distance of 12,000 ft (3.86 km). HDSL uses two twisted pairs (one pair for each direction) to achieve full-duplex transmission.

SDSL

The **symmetric digital subscriber line (SDSL)** is a one twisted-pair version of HDSL. It provides full-duplex symmetric communication supporting up to 768 kbps in each direction. SDSL, which provides symmetric communication, can be considered an alternative to ADSL. ADSL provides asymmetric communication, with a downstream bit rate that is much higher than the upstream bit rate. Although this feature meets the needs of most residential subscribers, it is not suitable for businesses that send and receive data in large volumes in both directions.

VDSL

The **very high-bit-rate digital subscriber line (VDSL),** an alternative approach that is similar to ADSL, uses coaxial, fiber-optic, or twisted-pair cable for short distances. The modulating technique is DMT. It provides a range of bit rates (25 to 55 Mbps) for upstream communication at distances of 3000 to 10,000 ft. The downstream rate is normally 3.2 Mbps.

Summary

Table 9.1 shows a summary of DSL technologies. Note that the data rate and distances are approximations and can vary from one implementation to another.

Table 9.1 *Summary of DSL technologies*

Technology	*Downstream Rate*	*Upstream Rate*	*Distance (ft)*	*Twisted Pairs*	*Line Code*
ADSL	1.5–6.1 Mbps	16–640 kbps	12,000	1	DMT
ADSL Lite	1.5 Mbps	500 kbps	18,000	1	DMT
HDSL	1.5–2.0 Mbps	1.5–2.0 Mbps	12,000	2	2B1Q
SDSL	768 kbps	768 kbps	12,000	1	2B1Q
VDSL	25–55 Mbps	3.2 Mbps	3000–10,000	1	DMT

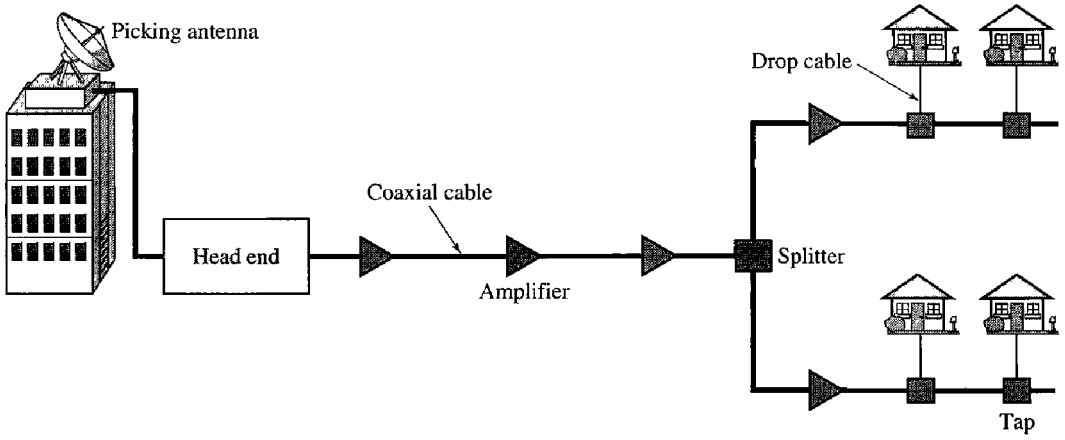
9.4 CABLE TV NETWORKS

The **cable TV network** started as a video service provider, but it has moved to the business of Internet access. In this section, we discuss cable TV networks per se; in Section 9.5 we discuss how this network can be used to provide high-speed access to the Internet.

Traditional Cable Networks

Cable TV started to distribute broadcast video signals to locations with poor or no reception in the late 1940s. It was called **community antenna TV (CATV)** because an antenna at the top of a tall hill or building received the signals from the TV stations and distributed them, via coaxial cables, to the community. Figure 9.14 shows a schematic diagram of a traditional cable TV network.

Figure 9.14 *Traditional cable TV network*



The cable TV office, called the **head end,** receives video signals from broadcasting stations and feeds the signals into coaxial cables. The signals became weaker and weaker with distance, so amplifiers were installed through the network to renew the signals. There could be up to 35 amplifiers between the head end and the subscriber premises. At the other end, splitters split the cable, and taps and drop cables make the connections to the subscriber premises.

The traditional cable TV system used coaxial cable end to end. Due to attenuation of the signals and the use of a large number of amplifiers, communication in the traditional network was unidirectional (one-way). Video signals were transmitted downstream, from the head end to the subscriber premises.

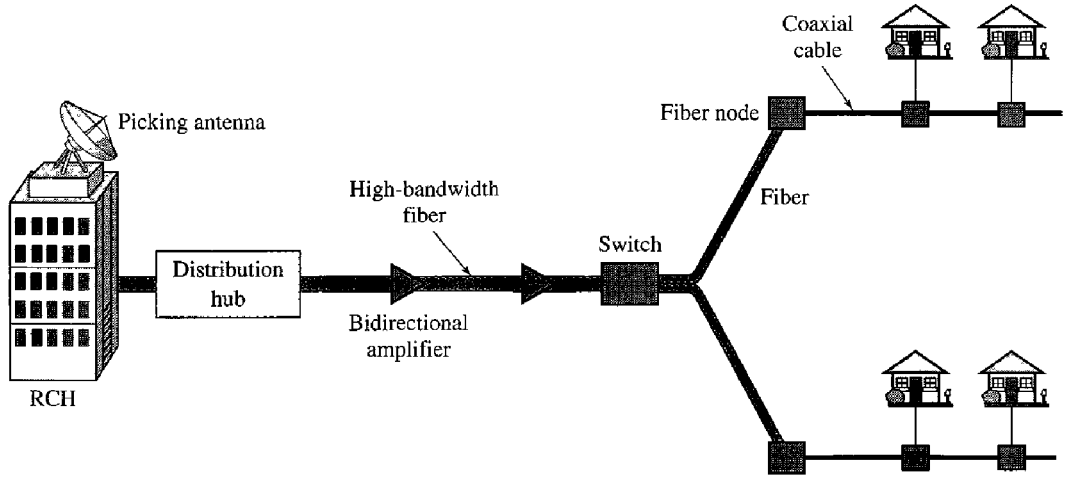
Communication in the traditional cable TV network is unidirectional.

Hybrid Fiber-Coaxial (HFC) Network

The second generation of cable networks is called a **hybrid fiber-coaxial (HFC) network.** The network uses a combination of fiber-optic and coaxial cable. The transmission

medium from the cable TV office to a box, called the **fiber node,** is optical fiber; from the fiber node through the neighborhood and into the house is still coaxial cable. Figure 9.15 shows a schematic diagram of an HFC network.

Figure 9.15 *Hybrid fiber-coaxial (HFC) network*



The **regional cable head (RCH)** normally serves up to 400,000 subscribers. The RCHs feed the **distribution hubs,** each of which serves up to 40,000 subscribers. The distribution hub plays an important role in the new infrastructure. Modulation and distribution of signals are done here; the signals are then fed to the fiber nodes through fiber-optic cables. The fiber node splits the analog signals so that the same signal is sent to each coaxial cable. Each coaxial cable serves up to 1000 subscribers. The use of fiber-optic cable reduces the need for amplifiers down to eight or less.

One reason for moving from traditional to hybrid infrastructure is to make the cable network bidirectional (two-way).

Communication in an HFC cable TV network can be bidirectional.

9.5 CABLE TV FOR DATA TRANSFER

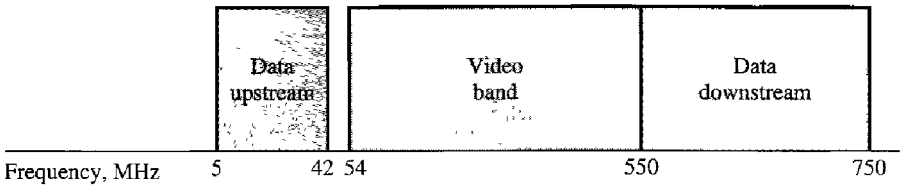
Cable companies are now competing with telephone companies for the residential customer who wants high-speed data transfer. DSL technology provides high-data-rate connections for residential subscribers over the local loop. However, DSL uses the existing unshielded twisted-pair cable, which is very susceptible to interference. This imposes an upper limit on the data rate. Another solution is the use of the cable TV network. In this section, we briefly discuss this technology.

Bandwidth

Even in an HFC system, the last part of the network, from the fiber node to the subscriber premises, is still a coaxial cable. This coaxial cable has a bandwidth that ranges

from 5 to 750 MHz (approximately). To provide Internet access, the cable company has divided this bandwidth into three bands: video, downstream data, and upstream data, as shown in Figure 9.16.

Figure 9.16 *Division of coaxial cable band by CATV*



Downstream Video Band

The **downstream video band** occupies frequencies from 54 to 550 MHz. Since each TV channel occupies 6 MHz, this can accommodate more than 80 channels.

Downstream Data Band

The downstream data (from the Internet to the subscriber premises) occupies the upper band, from 550 to 750 MHz. This band is also divided into 6-MHz channels.

Modulation **Downstream data band** uses the 64-QAM (or possibly 256-QAM) modulation technique.

Downstream data are modulated using the 64-QAM modulation technique.

Data Rate There is 6 bits/baud in 64-QAM. One bit is used for forward error correction; this leaves 5 bits of data per baud. The standard specifies 1 Hz for each baud; this means that, theoretically, downstream data can be received at 30 Mbps (5 bits/Hz × 6 MHz). The standard specifies only 27 Mbps. However, since the cable modem is normally connected to the computer through a 10Base-T cable (see Chapter 13), this limits the data rate to 10 Mbps.

The theoretical downstream data rate is 30 Mbps.

Upstream Data Band

The upstream data (from the subscriber premises to the Internet) occupies the lower band, from 5 to 42 MHz. This band is also divided into 6-MHz channels.

Modulation The **upstream data band** uses lower frequencies that are more susceptible to noise and interference. For this reason, the QAM technique is not suitable for this band. A better solution is QPSK.

Upstream data are modulated using the QPSK modulation technique.

Data Rate There are 2 bits/baud in QPSK. The standard specifies 1 Hz for each baud; this means that, theoretically, upstream data can be sent at 12 Mbps (2 bits/Hz × 6 MHz). However, the data rate is usually less than 12 Mbps.

The theoretical upstream data rate is 12 Mbps.

Sharing

Both upstream and downstream bands are shared by the subscribers.

Upstream Sharing

The upstream data bandwidth is 37 MHz. This means that there are only six 6-MHz channels available in the upstream direction. A subscriber needs to use one channel to send data in the upstream direction. The question is, "How can six channels be shared in an area with 1000, 2000, or even 100,000 subscribers?" The solution is timesharing. The band is divided into channels using FDM; these channels must be shared between subscribers in the same neighborhood. The cable provider allocates one channel, statically or dynamically, for a group of subscribers. If one subscriber wants to send data, she or he contends for the channel with others who want access; the subscriber must wait until the channel is available.

Downstream Sharing

We have a similar situation in the downstream direction. The downstream band has 33 channels of 6 MHz. A cable provider probably has more than 33 subscribers; therefore, each channel must be shared between a group of subscribers. However, the situation is different for the downstream direction; here we have a multicasting situation. If there are data for any of the subscribers in the group, the data are sent to that channel. Each subscriber is sent the data. But since each subscriber also has an address registered with the provider; the cable modem for the group matches the address carried with the data to the address assigned by the provider. If the address matches, the data are kept; otherwise, they are discarded.

CM and CMTS

To use a cable network for data transmission, we need two key devices: a **cable modem (CM)** and a **cable modem transmission system (CMTS).**

CM

The **cable modem (CM)** is installed on the subscriber premises. It is similar to an ADSL modem. Figure 9.17 shows its location.

CMTS

The **cable modem transmission system (CMTS)** is installed inside the distribution hub by the cable company. It receives data from the Internet and passes them to the combiner, which sends them to the subscriber. The CMTS also receives data from the subscriber and passes them to the Internet. Figure 9.18 shows the location of the CMTS.

Figure 9.17 *Cable modem (CM)*

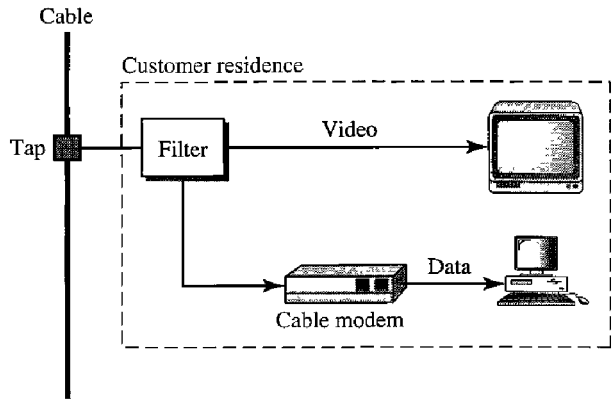
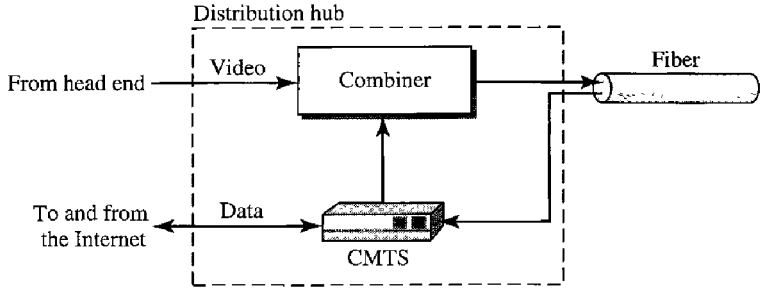


Figure 9.18 *Cable modem transmission system (CMTS)*



Data Transmission Schemes: DOCSIS

During the last few decades, several schemes have been designed to create a standard for data transmission over an HFC network. Prevalent is the one devised by Multimedia Cable Network Systems (MCNS), called **Data Over Cable System Interface Specification (DOCSIS).** DOCSIS defines all the protocols necessary to transport data from a CMTS to a CM.

Upstream Communication

The following is a very simplified version of the protocol defined by DOCSIS for upstream communication. It describes the steps that must be followed by a CM:

1. The CM checks the downstream channels for a specific packet periodically sent by the CMTS. The packet asks any new CM to announce itself on a specific upstream channel.
2. The CMTS sends a packet to the CM, defining its allocated downstream and upstream channels.
3. The CM then starts a process, called **ranging,** which determines the distance between the CM and CMTS. This process is required for synchronization between all

CMs and CMTSs for the minislots used for timesharing of the upstream channels. We will learn about this timesharing when we discuss contention protocols in Chapter 12.

4. The CM sends a packet to the ISP, asking for the Internet address.
5. The CM and CMTS then exchange some packets to establish security parameters, which are needed for a public network such as cable TV.
6. The CM sends its unique identifier to the CMTS.
7. Upstream communication can start in the allocated upstream channel; the CM can contend for the minislots to send data.

Downstream Communication

In the downstream direction, the communication is much simpler. There is no contention because there is only one sender. The CMTS sends the packet with the address of the receiving CM, using the allocated downstream channel.

9.6 RECOMMENDED READING

For more details about subjects discussed in this chapter, we recommend the following books. The items in brackets [. . .] refer to the reference list at the end of the text.

Books

[Cou01] gives an interesting discussion about telephone systems, DSL technology, and CATV in Chapter 8. [Tan03] discusses telephone systems and DSL technology in Section 2.5 and CATV in Section 2.7. [GW04] discusses telephone systems in Section 1.1.1 and standard modems in Section 3.7.3. A complete coverage of residential broadband (DSL and CATV) can be found in [Max99].

9.7 KEY TERMS

56K modem
800 service
900 service
ADSL Lite
ADSL modem
analog leased service
analog switched service
asymmetric DSL (ADSL)
cable modem (CM)
cable modem transmission system (CMTS)
cable TV network
common carrier
community antenna TV (CATV)
competitive local exchange carrier (CLEC)
Data Over Cable System Interface Specification (DOCSIS)
demodulator
digital data service (DDS)
digital service
digital subscriber line (DSL)
digital subscriber line access multiplexer (DSLAM)
discrete multitone technique (DMT)
distribution hub

downloading
downstream data band
end office
fiber node
head end
high-bit-rate DSL (HDSL)
hybrid fiber-coaxial (HFC) network
in-band signaling
incumbent local exchange carrier (ILEC)
interexchange carrier (IXC)
ISDN user port (ISUP)
local access transport area (LATA)
local exchange carrier (LEC)
local loop
long distance company
message transport port (MTP) level
modem
modulator
out-of-band signaling
plain old telephone system (POTS)
point of presence (POP)
ranging
regional cable head (RCH)
regional office
server control point (SCP)
signal point (SP)
signal transport port (STP)
signaling connection control point (SCCP)
Signaling System Seven (SS7)
switched/56 service
switching office
symmetric DSL (SDSL)
tandem office
telephone user port (TUP)
transaction capabilities application port (TCAP)
trunk
uploading
upstream data band
V.32
V.32bis
V.34bis
V.90
V.92
very-high-bit-rate DSL (VDSL)
video band
V-series
wide-area telephone service (WATS)

9.8 SUMMARY

- The telephone, which is referred to as the plain old telephone system (POTS), was originally an analog system. During the last decade, the telephone network has undergone many technical changes. The network is now digital as well as analog.
- The telephone network is made of three major components: local loops, trunks, and switching offices. It has several levels of switching offices such as end offices, tandem offices, and regional offices.
- The United States is divided into many local access transport areas (LATAs). The services offered inside a LATA are called intra-LATA services. The carrier that handles these services is called a local exchange carrier (LEC). The services between LATAs are handled by interexchange carriers (IXCs).
- In in-band signaling, the same circuit is used for both signaling and data. In out-of-band signaling, a portion of the bandwidth is used for signaling and another portion

for data. The protocol that is used for signaling in the telephone network is called Signaling System Seven (SS7).

- Telephone companies provide two types of services: analog and digital. We can categorize analog services as either analog switched services or analog leased services. The two most common digital services are switched/56 service and digital data service (DDS).
- Data transfer using the telephone local loop was traditionally done using a dial-up modem. The term *modem* is a composite word that refers to the two functional entities that make up the device: a signal modulator and a signal demodulator.
- Most popular modems available are based on the V-series standards. The V.32 modem has a data rate of 9600 bps. The V.32bis modem supports 14,400-bps transmission. V.90 modems, called 56K modems, with a downloading rate of 56 kbps and uploading rate of 33.6 kbps are very common. The standard above V.90 is called V.92. These modems can adjust their speed, and if the noise allows, they can upload data at the rate of 48 kbps.
- Telephone companies developed another technology, digital subscriber line (DSL), to provide higher-speed access to the Internet. DSL technology is a set of technologies, each differing in the first letter (ADSL, VDSL, HDSL, and SDSL. ADSL provides higher speed in the downstream direction than in the upstream direction. The high-bit-rate digital subscriber line (HDSL) was designed as an alternative to the T-1 line (1.544 Mbps). The symmetric digital subscriber line (SDSL) is a one twisted-pair version of HDSL. The very high-bit-rate digital subscriber line (VDSL) is an alternative approach that is similar to ADSL.
- Community antenna TV (CATV) was originally designed to provide video services for the community. The traditional cable TV system used coaxial cable end to end. The second generation of cable networks is called a hybrid fiber-coaxial (HFC) network. The network uses a combination of fiber-optic and coaxial cable.
- Cable companies are now competing with telephone companies for the residential customer who wants high-speed access to the Internet. To use a cable network for data transmission, we need two key devices: a cable modem (CM) and a cable modem transmission system (CMTS).

9.9 PRACTICE SET

Review Questions

1. What are the three major components of a telephone network?
2. Give some hierarchical switching levels of a telephone network.
3. What is LATA? What are intra-LATA and inter-LATA services?
4. Describe the SS7 service and its relation to the telephone network.
5. What are the two major services provided by telephone companies in the United States?
6. What is dial-up modem technology? List some of the common modem standards discussed in this chapter and give their data rates.

7. What is DSL technology? What are the services provided by the telephone companies using DSL? Distinguish between a DSL modem and a DSLAM.
8. Compare and contrast a traditional cable network with a hybrid fiber-coaxial network.
9. How is data transfer achieved using CATV channels?
10. Distinguish between CM and CMTS.

Exercises

11. Using the discussion of circuit-switching in Chapter 8, explain why this type of switching was chosen for telephone networks.
12. In Chapter 8, we discussed the three communication phases involved in a circuit-switched network. Match these phases with the phases in a telephone call between two parties.
13. In Chapter 8, we learned that a circuit-switched network needs end-to-end addressing during the setup and teardown phases. Define end-to-end addressing in a telephone network when two parties communicate.
14. When we have an overseas telephone conversation, we sometimes experience a delay. Can you explain the reason?
15. Draw a barchart to compare the different downloading data rates of common modems.
16. Draw a barchart to compare the different downloading data rates of common DSL technology implementations (use minimum data rates).
17. Calculate the minimum time required to download one million bytes of information using each of the following technologies:
    a. V.32 modem
    b. V.32bis modem
    c. V.90 modem
18. Repeat Exercise 17 using different DSL implementations (consider the minimum rates).
19. Repeat Exercise 17 using a cable modem (consider the minimum rates).
20. What type of topology is used when customers in an area use DSL modems for data transfer purposes? Explain.
21. What type of topology is used when customers in an area use cable modems for data transfer purposes? Explain.