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CHAPTER 1

Introduction

Data communications and networking have changed the way we do business and the way we live. Business decisions have to be made ever more quickly, and the decision makers require immediate access to accurate information. Why wait a week for that report from Europe to arrive by mail when it could appear almost instantaneously through computer networks? Businesses today rely on computer networks and internetworks.

Data communication and networking have found their way not only through business and personal communication, they have found many applications in political and social issues. People have found how to communicate with other people in the world to express their social and political opinions and problems. Communities in the world are not isolated anymore.

But before we ask how quickly we can get hooked up, we need to know how networks operate, what types of technologies are available, and which design best fills which set of needs.

This chapter paves the way for the rest of the book. It is divided into five sections.

- The first section introduces data communications and defines their components and the types of data exchanged. It also shows how different types of data are represented and how data is flowed through the network.
- The second section introduces networks and defines their criteria and structures. It introduces four different network topologies that are encountered throughout the book.
- The third section discusses different types of networks: LANs, WANs, and internetworks (internets). It also introduces the Internet, the largest internet in the world. The concept of switching is also introduced in this section to show how small networks can be combined to create larger ones.
- The fourth section covers a brief history of the Internet. The section is divided into three eras: early history, the birth of the Internet, and the issues related to the Internet today. This section can be skipped if the reader is familiar with this history.
- The fifth section covers standards and standards organizations. The section covers Internet standards and Internet administration. We refer to these standards and organizations throughout the book.

1.1 DATA COMMUNICATIONS

When we communicate, we are sharing information. This sharing can be local or remote. Between individuals, local communication usually occurs face to face, while remote communication takes place over distance. The term ***telecommunication,*** which includes telephony, telegraphy, and television, means communication at a distance (*tele* is Greek for “far”). The word ***data*** refers to information presented in whatever form is agreed upon by the parties creating and using the data.

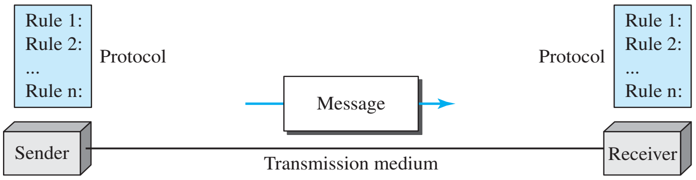
Data communications are the exchange of data between two devices via some form of transmission medium such as a wire cable. For data communications to occur, the communicating devices must be part of a communication system made up of a combination of hardware (physical equipment) and software (programs). The effectiveness of a data communications system depends on four fundamental characteristics: delivery, accuracy, timeliness, and jitter.

1. **Delivery.** The system must deliver data to the correct destination. Data must be received by the intended device or user and only by that device or user.
2. **Accuracy.** The system must deliver the data accurately. Data that have been altered in transmission and left uncorrected are unusable.
3. **Timeliness.** The system must deliver data in a timely manner. Data delivered late are useless. In the case of video and audio, timely delivery means delivering data as they are produced, in the same order that they are produced, and without significant delay. This kind of delivery is called *real-time* transmission.
4. **Jitter.** Jitter refers to the variation in the packet arrival time. It is the uneven delay in the delivery of audio or video packets. For example, let us assume that video packets are sent every 30 ms. If some of the packets arrive with 30-ms delay and others with 40-ms delay, an uneven quality in the video is the result.

1.1.1 Components

A data communications system has five components (see Figure 1.1).

Figure 1.1 *Five components of data communication*



1. **Message.** The **message** is the information (data) to be communicated. Popular forms of information include text, numbers, pictures, audio, and video.
2. **Sender.** The **sender** is the device that sends the data message. It can be a computer, workstation, telephone handset, video camera, and so on.

3. **Receiver.** The **receiver** is the device that receives the message. It can be a computer, workstation, telephone handset, television, and so on.
4. **Transmission medium.** The **transmission medium** is the physical path by which a message travels from sender to receiver. Some examples of transmission media include twisted-pair wire, coaxial cable, fiber-optic cable, and radio waves.
5. **Protocol.** A protocol is a set of rules that govern data communications. It represents an agreement between the communicating devices. Without a protocol, two devices may be connected but not communicating, just as a person speaking French cannot be understood by a person who speaks only Japanese.

1.1.2 Data Representation

Information today comes in different forms such as text, numbers, images, audio, and video.

Text

In data communications, text is represented as a bit pattern, a sequence of bits (0s or 1s). Different sets of bit patterns have been designed to represent text symbols. Each set is called a **code,** and the process of representing symbols is called coding. Today, the prevalent coding system is called **Unicode,** which uses 32 bits to represent a symbol or character used in any language in the world. The **American Standard Code for Information Interchange (ASCII),** developed some decades ago in the United States, now constitutes the first 127 characters in Unicode and is also referred to as **Basic Latin.** Appendix A includes part of the Unicode.

Numbers

Numbers are also represented by bit patterns. However, a code such as ASCII is not used to represent numbers; the number is directly converted to a binary number to simplify mathematical operations. Appendix B discusses several different numbering systems.

Images

Images are also represented by bit patterns. In its simplest form, an image is composed of a matrix of pixels (picture elements), where each pixel is a small dot. The size of the pixel depends on the *resolution.* For example, an image can be divided into 1000 pixels or 10,000 pixels. In the second case, there is a better representation of the image (better resolution), but more memory is needed to store the image.

After an image is divided into pixels, each pixel is assigned a bit pattern. The size and the value of the pattern depend on the image. For an image made of only black-and-white dots (e.g., a chessboard), a 1-bit pattern is enough to represent a pixel.

If an image is not made of pure white and pure black pixels, we can increase the size of the bit pattern to include gray scale. For example, to show four levels of gray scale, we can use 2-bit patterns. A black pixel can be represented by 00, a dark gray pixel by 01, a light gray pixel by 10, and a white pixel by 11.

There are several methods to represent color images. One method is called **RGB,** so called because each color is made of a combination of three primary colors: *r*ed, *g*reen, and *b*lue. The intensity of each color is measured, and a bit pattern is assigned to it. Another method is called **YCM,** in which a color is made of a combination of three other primary colors: *y*ellow, *c*yan, and *m*agenta.

Audio

Audio refers to the recording or broadcasting of sound or music. Audio is by nature different from text, numbers, or images. It is continuous, not discrete. Even when we use a microphone to change voice or music to an electric signal, we create a continuous signal. We will learn more about audio in Chapter 26.

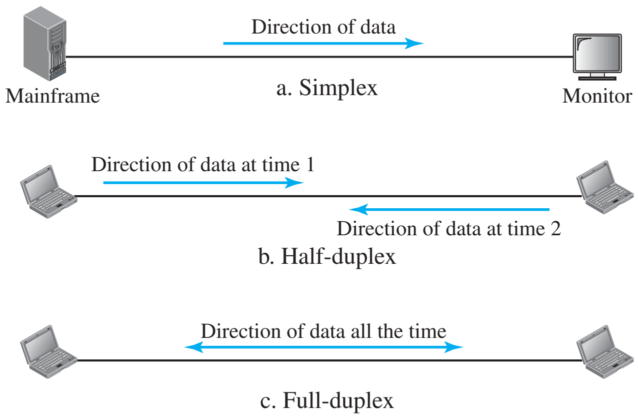
Video

Video refers to the recording or broadcasting of a picture or movie. Video can either be produced as a continuous entity (e.g., by a TV camera), or it can be a combination of images, each a discrete entity, arranged to convey the idea of motion. We will learn more about video in Chapter 26.

1.1.3 Data Flow

Communication between two devices can be simplex, half-duplex, or full-duplex as shown in Figure 1.2.

Figure 1.2 *Data flow (simplex, half-duplex, and full-duplex)*


Direction of data
Mainframe
Monitor
Direction of data at time 1
Direction of data at time 2
Direction of data all the time

a. Simplex

b. Half-duplex

c. Full-duplex

Simplex

In **simplex mode,** the communication is unidirectional, as on a one-way street. Only one of the two devices on a link can transmit; the other can only receive (see Figure 1.2a).

Keyboards and traditional monitors are examples of simplex devices. The keyboard can only introduce input; the monitor can only accept output. The simplex mode can use the entire capacity of the channel to send data in one direction.

Half-Duplex

In **half-duplex mode,** each station can both transmit and receive, but not at the same time. When one device is sending, the other can only receive, and vice versa (see Figure 1.2b).

The half-duplex mode is like a one-lane road with traffic allowed in both directions. When cars are traveling in one direction, cars going the other way must wait. In a half-duplex transmission, the entire capacity of a channel is taken over by whichever of the two devices is transmitting at the time. Walkie-talkies and CB (citizens band) radios are both half-duplex systems.

The half-duplex mode is used in cases where there is no need for communication in both directions at the same time; the entire capacity of the channel can be utilized for each direction.

Full-Duplex

In **full-duplex mode** (also called *duplex*), both stations can transmit and receive simultaneously (see Figure 1.2c).

The full-duplex mode is like a two-way street with traffic flowing in both directions at the same time. In full-duplex mode, signals going in one direction share the capacity of the link with signals going in the other direction. This sharing can occur in two ways: Either the link must contain two physically separate transmission paths, one for sending and the other for receiving; or the capacity of the channel is divided between signals traveling in both directions.

One common example of full-duplex communication is the telephone network. When two people are communicating by a telephone line, both can talk and listen at the same time.

The full-duplex mode is used when communication in both directions is required all the time. The capacity of the channel, however, must be divided between the two directions.

1.2 NETWORKS

A **network** is the interconnection of a set of devices capable of communication. In this definition, a device can be a **host** (or an *end system* as it is sometimes called) such as a large computer, desktop, laptop, workstation, cellular phone, or security system. A device in this definition can also be a **connecting device** such as a router, which connects the network to other networks, a switch, which connects devices together, a modem (modulator-demodulator), which changes the form of data, and so on. These devices in a network are connected using wired or wireless transmission media such as cable or air. When we connect two computers at home using a plug-and-play router, we have created a network, although very small.

1.2.1 Network Criteria

A network must be able to meet a certain number of criteria. The most important of these are performance, reliability, and security.

Performance

Performance can be measured in many ways, including transit time and response time. Transit time is the amount of time required for a message to travel from one device to another. Response time is the elapsed time between an inquiry and a response. The performance of a network depends on a number of factors, including the number of users, the type of transmission medium, the capabilities of the connected hardware, and the efficiency of the software.

Performance is often evaluated by two networking metrics: **throughput** and **delay.** We often need more throughput and less delay. However, these two criteria are often contradictory. If we try to send more data to the network, we may increase throughput but we increase the delay because of traffic congestion in the network.

Reliability

In addition to accuracy of delivery, network **reliability** is measured by the frequency of failure, the time it takes a link to recover from a failure, and the network's robustness in a catastrophe.

Security

Network **security** issues include protecting data from unauthorized access, protecting data from damage and development, and implementing policies and procedures for recovery from breaches and data losses.

1.2.2 Physical Structures

Before discussing networks, we need to define some network attributes.

Type of Connection

A network is two or more devices connected through links. A link is a communications pathway that transfers data from one device to another. For visualization purposes, it is simplest to imagine any link as a line drawn between two points. For communication to occur, two devices must be connected in some way to the same link at the same time. There are two possible types of connections: point-to-point and multipoint.

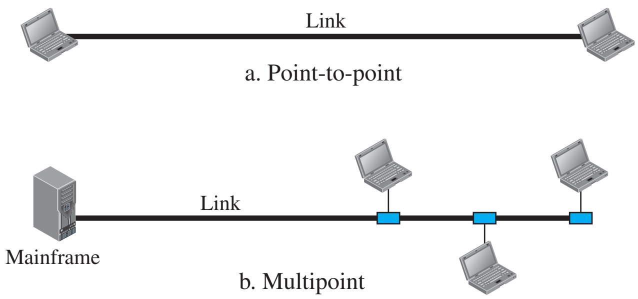
Point-to-Point

A **point-to-point connection** provides a dedicated link between two devices. The entire capacity of the link is reserved for transmission between those two devices. Most point-to-point connections use an actual length of wire or cable to connect the two ends, but other options, such as microwave or satellite links, are also possible (see Figure 1.3a). When we change television channels by infrared remote control, we are establishing a point-to-point connection between the remote control and the television's control system.

Multipoint

A **multipoint** (also called **multidrop**) **connection** is one in which more than two specific devices share a single link (see Figure 1.3b).

Figure 1.3 *Types of connections: point-to-point and multipoint*



In a multipoint environment, the capacity of the channel is shared, either spatially or temporally. If several devices can use the link simultaneously, it is a *spatially shared* connection. If users must take turns, it is a *timeshared* connection.

Physical Topology

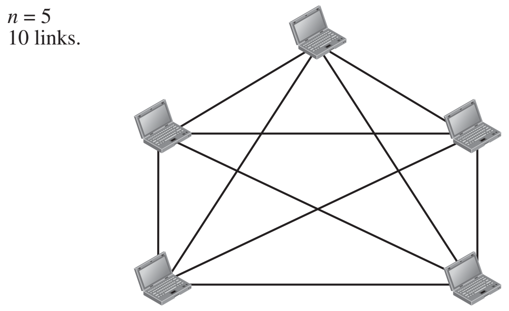
The term ***physical topology*** refers to the way in which a network is laid out physically. Two or more devices connect to a link; two or more links form a topology. The topology of a network is the geometric representation of the relationship of all the links and linking devices (usually called ***nodes***) to one another. There are four basic topologies possible: mesh, star, bus, and ring.

Mesh Topology

In a **mesh topology,** every device has a dedicated point-to-point link to every other device. The term *dedicated* means that the link carries traffic only between the two devices it connects. To find the number of physical links in a fully connected mesh network with n nodes, we first consider that each node must be connected to every other node. Node 1 must be connected to $n - 1$ nodes, node 2 must be connected to $n - 1$ nodes, and finally node n must be connected to $n - 1$ nodes. We need $n (n - 1)$ physical links. However, if each physical link allows communication in both directions (duplex mode), we can divide the number of links by 2. In other words, we can say that in a mesh topology, we need $\boldsymbol{n (n - 1) / 2}$ duplex-mode links. To accommodate that many links, every device on the network must have $n - 1$ input/output (I/O) ports (see Figure 1.4) to be connected to the other $n - 1$ stations.

A mesh offers several advantages over other network topologies. First, the use of dedicated links guarantees that each connection can carry its own data load, thus eliminating the traffic problems that can occur when links must be shared by multiple devices. Second, a mesh topology is robust. If one link becomes unusable, it does not incapacitate the entire system. Third, there is the advantage of privacy or security. When every message travels along a dedicated line, only the intended recipient sees it. Physical boundaries prevent other users from gaining access to messages. Finally, point-to-point links make fault identification and fault isolation easy. Traffic can be routed to avoid links with suspected problems. This facility enables the network manager to discover the precise location of the fault and aids in finding its cause and solution.

Figure 1.4 *A fully connected mesh topology (five devices)*



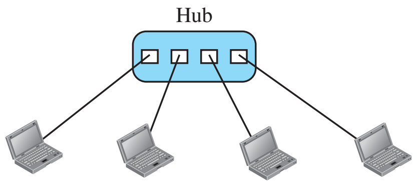
The main disadvantages of a mesh are related to the amount of cabling and the number of I/O ports required. First, because every device must be connected to every other device, installation and reconnection are difficult. Second, the sheer bulk of the wiring can be greater than the available space (in walls, ceilings, or floors) can accommodate. Finally, the hardware required to connect each link (I/O ports and cable) can be prohibitively expensive. For these reasons a mesh topology is usually implemented in a limited fashion, for example, as a backbone connecting the main computers of a hybrid network that can include several other topologies.

One practical example of a mesh topology is the connection of telephone regional offices in which each regional office needs to be connected to every other regional office.

Star Topology

In a **star topology,** each device has a dedicated point-to-point link only to a central controller, usually called a ***hub.*** The devices are not directly linked to one another. Unlike a mesh topology, a star topology does not allow direct traffic between devices. The controller acts as an exchange: If one device wants to send data to another, it sends the data to the controller, which then relays the data to the other connected device (see Figure 1.5) .

Figure 1.5 *A star topology connecting four stations*



A star topology is less expensive than a mesh topology. In a star, each device needs only one link and one I/O port to connect it to any number of others. This factor also makes it easy to install and reconfigure. Far less cabling needs to be housed, and additions, moves, and deletions involve only one connection: between that device and the hub.

Other advantages include robustness. If one link fails, only that link is affected. All other links remain active. This factor also lends itself to easy fault identification and fault isolation. As long as the hub is working, it can be used to monitor link problems and bypass defective links.

One big disadvantage of a star topology is the dependency of the whole topology on one single point, the hub. If the hub goes down, the whole system is dead.

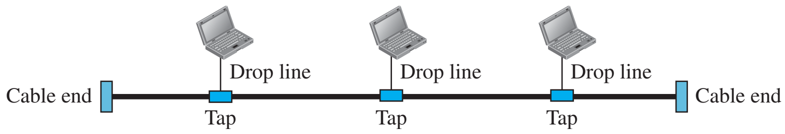
Although a star requires far less cable than a mesh, each node must be linked to a central hub. For this reason, often more cabling is required in a star than in some other topologies (such as ring or bus).

The star topology is used in local-area networks (LANs), as we will see in Chapter 13. High-speed LANs often use a star topology with a central hub.

Bus Topology

The preceding examples all describe point-to-point connections. A **bus topology,** on the other hand, is multipoint. One long cable acts as a **backbone** to link all the devices in a network (see Figure 1.6).

Figure 1.6 *A bus topology connecting three stations*



Nodes are connected to the bus cable by drop lines and taps. A drop line is a connection running between the device and the main cable. A tap is a connector that either splices into the main cable or punctures the sheathing of a cable to create a contact with the metallic core. As a signal travels along the backbone, some of its energy is transformed into heat. Therefore, it becomes weaker and weaker as it travels farther and farther. For this reason there is a limit on the number of taps a bus can support and on the distance between those taps.

Advantages of a bus topology include ease of installation. Backbone cable can be laid along the most efficient path, then connected to the nodes by drop lines of various lengths. In this way, a bus uses less cabling than mesh or star topologies. In a star, for example, four network devices in the same room require four lengths of cable reaching all the way to the hub. In a bus, this redundancy is eliminated. Only the backbone cable stretches through the entire facility. Each drop line has to reach only as far as the nearest point on the backbone.

Disadvantages include difficult reconnection and fault isolation. A bus is usually designed to be optimally efficient at installation. It can therefore be difficult to add new devices. Signal reflection at the taps can cause degradation in quality. This degradation can be controlled by limiting the number and spacing of devices connected to a given length of cable. Adding new devices may therefore require modification or replacement of the backbone.

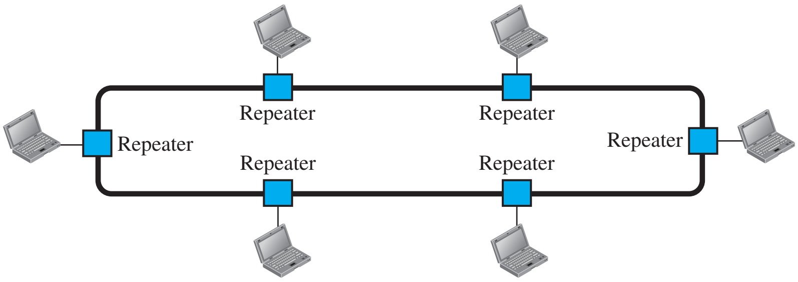
In addition, a fault or break in the bus cable stops all transmission, even between devices on the same side of the problem. The damaged area reflects signals back in the direction of origin, creating noise in both directions.

Bus topology was the one of the first topologies used in the design of early local-area networks. Traditional Ethernet LANs can use a bus topology, but they are less popular now for reasons we will discuss in Chapter 13.

Ring Topology

In a **ring topology,** each device has a dedicated point-to-point connection with only the two devices on either side of it. A signal is passed along the ring in one direction, from device to device, until it reaches its destination. Each device in the ring incorporates a repeater. When a device receives a signal intended for another device, its repeater regenerates the bits and passes them along (see Figure 1.7).

Figure 1.7 *A ring topology connecting six stations*



A ring is relatively easy to install and reconfigure. Each device is linked to only its immediate neighbors (either physically or logically). To add or delete a device requires changing only two connections. The only constraints are media and traffic considerations (maximum ring length and number of devices). In addition, fault isolation is simplified. Generally, in a ring a signal is circulating at all times. If one device does not receive a signal within a specified period, it can issue an alarm. The alarm alerts the network operator to the problem and its location.

However, unidirectional traffic can be a disadvantage. In a simple ring, a break in the ring (such as a disabled station) can disable the entire network. This weakness can be solved by using a dual ring or a switch capable of closing off the break.

Ring topology was prevalent when IBM introduced its local-area network, Token Ring. Today, the need for higher-speed LANs has made this topology less popular.

1.3 NETWORK TYPES

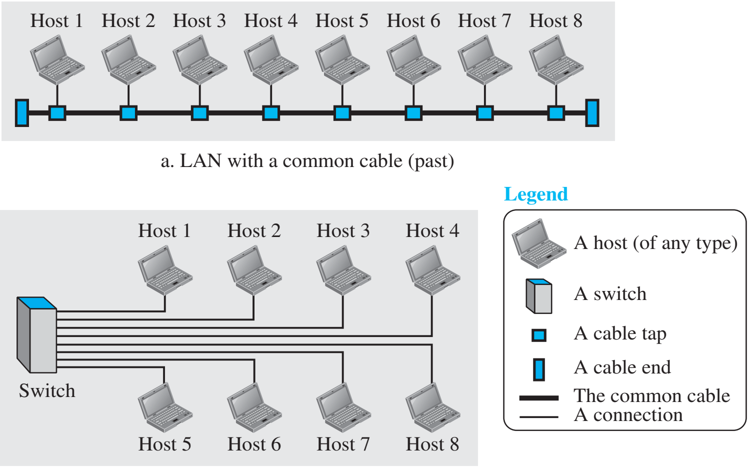
After defining networks in the previous section and discussing their physical structures, we need to discuss different types of networks we encounter in the world today. The criteria of distinguishing one type of network from another is difficult and sometimes confusing. We use a few criteria such as size, geographical coverage, and ownership to make this distinction. After discussing two types of networks, LANs and WANs, we define switching, which is used to connect networks to form an internetwork (a network of networks).

1.3.1 Local Area Network

A **local area network** (**LAN**) is usually privately owned and connects some hosts in a single office, building, or campus. Depending on the needs of an organization, a LAN can be as simple as two PCs and a printer in someone's home office, or it can extend throughout a company and include audio and video devices. Each host in a LAN has an identifier, an address, that uniquely defines the host in the LAN. A packet sent by a host to another host carries both the source host's and the destination host's addresses.

In the past, all hosts in a network were connected through a common cable, which meant that a packet sent from one host to another was received by all hosts. The intended recipient kept the packet; the others dropped the packet. Today, most LANs use a smart connecting switch, which is able to recognize the destination address of the packet and guide the packet to its destination without sending it to all other hosts. The switch alleviates the traffic in the LAN and allows more than one pair to communicate with each other at the same time if there is no common source and destination among them. Note that the above definition of a LAN does not define the minimum or maximum number of hosts in a LAN. Figure 1.8 shows a LAN using either a common cable or a switch.

Figure 1.8 *An isolated LAN in the past and today*



a. LAN with a common cable (past)

b. LAN with a switch (today)

LANs are discussed in more detail in Part III of the book.

When LANs were used in isolation (which is rare today), they were designed to allow resources to be shared between the hosts. As we will see shortly, LANs today are connected to each other and to WANs (discussed next) to create communication at a wider level.

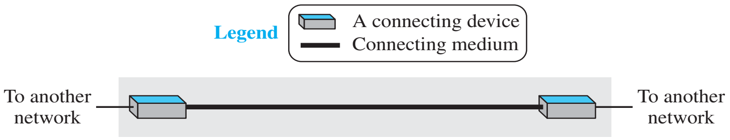
1.3.2 Wide Area Network

A **wide area network (WAN)** is also an interconnection of devices capable of communication. However, there are some differences between a LAN and a WAN. A LAN is normally limited in size, spanning an office, a building, or a campus; a WAN has a wider geographical span, spanning a town, a state, a country, or even the world. A LAN interconnects hosts; a WAN interconnects connecting devices such as switches, routers, or modems. A LAN is normally privately owned by the organization that uses it; a WAN is normally created and run by communication companies and leased by an organization that uses it. We see two distinct examples of WANs today: point-to-point WANs and switched WANs.

Point-to-Point WAN

A point-to-point WAN is a network that connects two communicating devices through a transmission media (cable or air). We will see examples of these WANs when we discuss how to connect the networks to one another. Figure 1.9 shows an example of a point-to-point WAN.

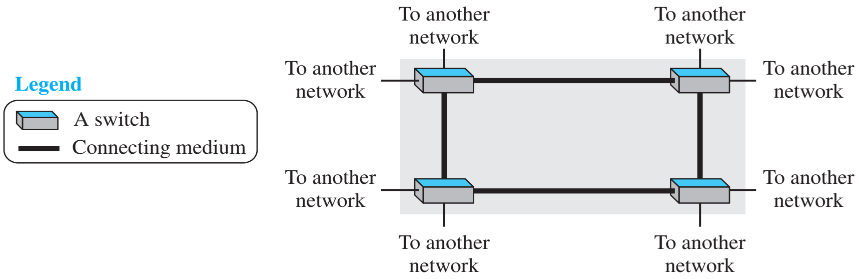
Figure 1.9 *A point-to-point WAN*



Switched WAN

A switched WAN is a network with more than two ends. A switched WAN, as we will see shortly, is used in the backbone of global communication today. We can say that a switched WAN is a combination of several point-to-point WANs that are connected by switches. Figure 1.10 shows an example of a switched WAN.

Figure 1.10 *A switched WAN*

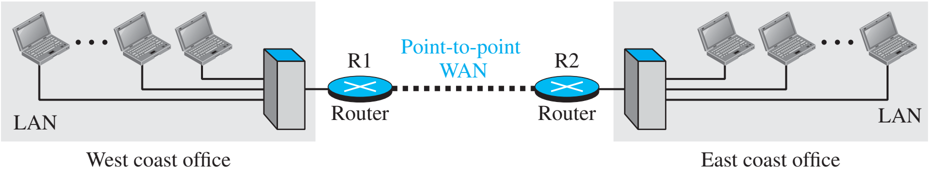


WANs are discussed in more detail in Part II of the book.

Internetwork

Today, it is very rare to see a LAN or a WAN in isolation; they are connected to one another. When two or more networks are connected, they make an **internetwork,** or **internet.** As an example, assume that an organization has two offices, one on the east coast and the other on the west coast. Each office has a LAN that allows all employees in the office to communicate with each other. To make the communication between employees at different offices possible, the management leases a point-to-point dedicated WAN from a service provider, such as a telephone company, and connects the two LANs. Now the company has an internetwork, or a private internet (with lowercase *i*). Communication between offices is now possible. Figure 1.11 shows this internet.

Figure 1.11 *An internetwork made of two LANs and one point-to-point WAN*



When a host in the west coast office sends a message to another host in the same office, the router blocks the message, but the switch directs the message to the destination. On the other hand, when a host on the west coast sends a message to a host on the east coast, router R1 routes the packet to router R2, and the packet reaches the destination.

Figure 1.12 (see next page) shows another internet with several LANs and WANs connected. One of the WANs is a switched WAN with four switches.

1.3.3 Switching

An internet is a **switched network** in which a switch connects at least two links together. A switch needs to forward data from a network to another network when required. The two most common types of switched networks are circuit-switched and packet-switched networks. We discuss both next.

Circuit-Switched Network

In a **circuit-switched network,** a dedicated connection, called a circuit, is always available between the two end systems; the switch can only make it active or inactive. Figure 1.13 shows a very simple switched network that connects four telephones to each end. We have used telephone sets instead of computers as an end system because circuit switching was very common in telephone networks in the past, although part of the telephone network today is a packet-switched network.

In Figure 1.13, the four telephones at each side are connected to a switch. The switch connects a telephone set at one side to a telephone set at the other side. The thick

Figure 1.12 *A heterogeneous network made of four WANs and three LANs*

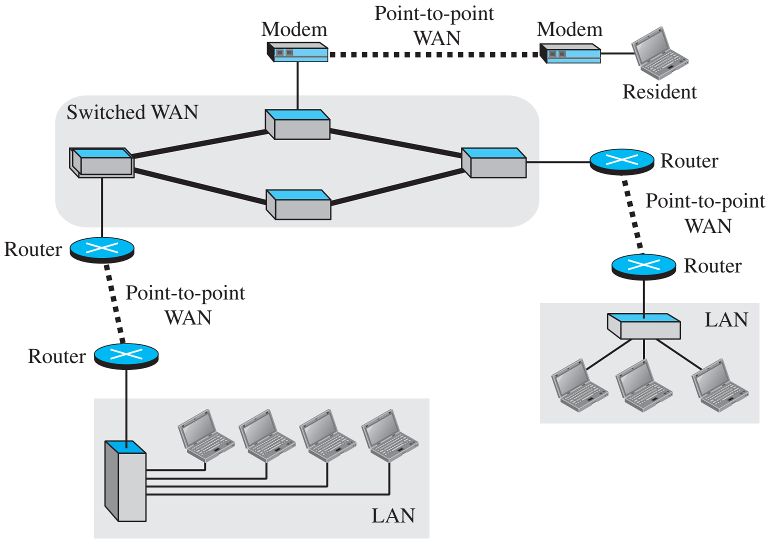
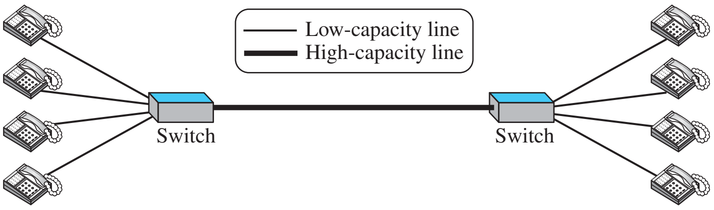


Figure 1.13 *A circuit-switched network*



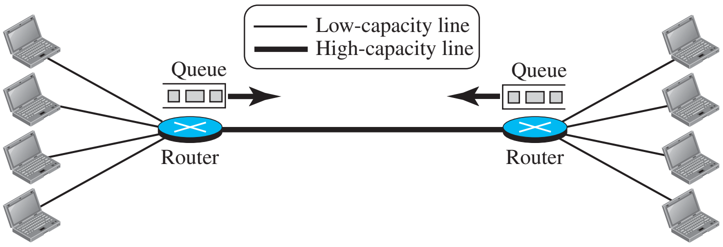
line connecting two switches is a high-capacity communication line that can handle four voice communications at the same time; the capacity can be shared between all pairs of telephone sets. The switches used in this example have forwarding tasks but no storing capability.

Let us look at two cases. In the first case, all telephone sets are busy; four people at one site are talking with four people at the other site; the capacity of the thick line is fully used. In the second case, only one telephone set at one side is connected to a telephone set at the other side; only one-fourth of the capacity of the thick line is used. This means that a circuit-switched network is efficient only when it is working at its full capacity; most of the time, it is inefficient because it is working at partial capacity. The reason that we need to make the capacity of the thick line four times the capacity of each voice line is that we do not want communication to fail when all telephone sets at one side want to be connected with all telephone sets at the other side.

Packet-Switched Network

In a computer network, the communication between the two ends is done in blocks of data called **packets.** In other words, instead of the continuous communication we see between two telephone sets when they are being used, we see the exchange of individual data packets between the two computers. This allows us to make the switches function for both storing and forwarding because a packet is an independent entity that can be stored and sent later. Figure 1.14 shows a small packet-switched network that connects four computers at one site to four computers at the other site.

Figure 1.14 *A packet-switched network*



A router in a packet-switched network has a queue that can store and forward the packet. Now assume that the capacity of the thick line is only twice the capacity of the data line connecting the computers to the routers. If only two computers (one at each site) need to communicate with each other, there is no waiting for the packets. However, if packets arrive at one router when the thick line is already working at its full capacity, the packets should be stored and forwarded in the order they arrived. The two simple examples show that a packet-switched network is more efficient than a circuit-switched network, but the packets may encounter some delays.

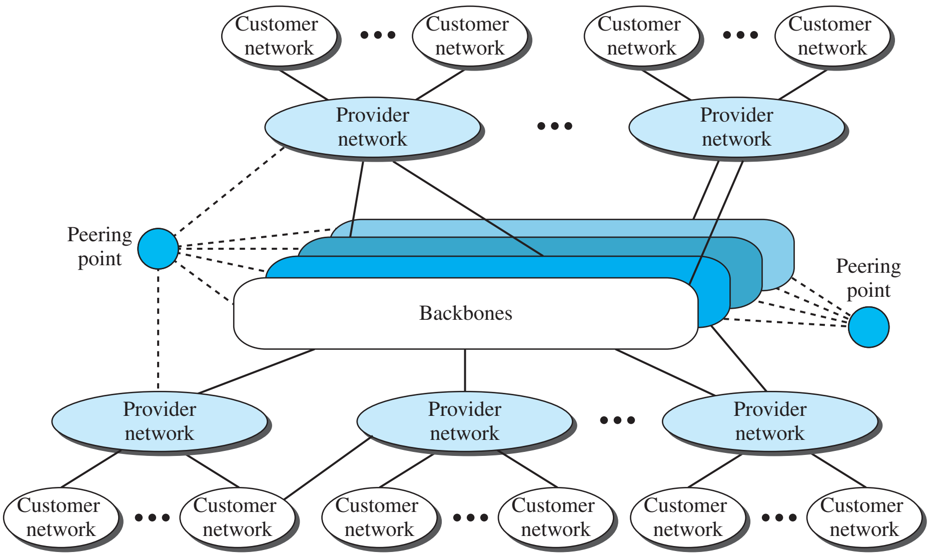
In this book, we mostly discuss packet-switched networks. In Chapter 18, we discuss packet-switched networks in more detail and discuss the performance of these networks.

1.3.4 The Internet

As we discussed before, an internet (note the lowercase *i*) is two or more networks that can communicate with each other. The most notable internet is called the **Internet** (uppercase *I*), and is composed of thousands of interconnected networks. Figure 1.15 shows a conceptual (not geographical) view of the Internet.

The figure shows the Internet as several backbones, provider networks, and customer networks. At the top level, the *backbones* are large networks owned by some communication companies such as Sprint, Verizon (MCI), AT&T, and NTT. The backbone networks are connected through some complex switching systems, called *peering points*. At the second level, there are smaller networks, called *provider networks*, that use the services of the backbones for a fee. The provider networks are connected to backbones and sometimes to other provider networks. The *customer networks* are

Figure 1.15 *The Internet today*



networks at the edge of the Internet that actually use the services provided by the Internet. They pay fees to provider networks for receiving services.

Backbones and provider networks are also called **Internet Service Providers (ISPs).** The backbones are often referred to as *international ISPs;* the provider networks are often referred to as *national* or *regional ISPs*.

1.3.5 Accessing the Internet

The Internet today is an internetwork that allows any user to become part of it. The user, however, needs to be physically connected to an ISP. The physical connection is normally done through a point-to-point WAN. In this section, we briefly describe how this can happen, but we postpone the technical details of the connection until Chapters 14 and 16.

Using Telephone Networks

Today most residences and small businesses have telephone service, which means they are connected to a telephone network. Since most telephone networks have already connected themselves to the Internet, one option for residences and small businesses to connect to the Internet is to change the voice line between the residence or business and the telephone center to a point-to-point WAN. This can be done in two ways.

- ***Dial-up service.*** The first solution is to add to the telephone line a modem that converts data to voice. The software installed on the computer dials the ISP and imitates making a telephone connection. Unfortunately, the dial-up service is very slow, and when the line is used for Internet connection, it cannot be used for telephone (voice) connection. It is only useful for small residences. We discuss dial-up service in Chapter 14.

- ***DSL Service.*** Since the advent of the Internet, some telephone companies have upgraded their telephone lines to provide higher speed Internet services to residences or small businesses. The DSL service also allows the line to be used simultaneously for voice and data communication. We discuss DSL in Chapter 14.

Using Cable Networks

More and more residents over the last two decades have begun using cable TV services instead of antennas to receive TV broadcasting. The cable companies have been upgrading their cable networks and connecting to the Internet. A residence or a small business can be connected to the Internet by using this service. It provides a higher speed connection, but the speed varies depending on the number of neighbors that use the same cable. We discuss the cable networks in Chapter 14.

Using Wireless Networks

Wireless connectivity has recently become increasingly popular. A household or a small business can use a combination of wireless and wired connections to access the Internet. With the growing wireless WAN access, a household or a small business can be connected to the Internet through a wireless WAN. We discuss wireless access in Chapter 16.

Direct Connection to the Internet

A large organization or a large corporation can itself become a local ISP and be connected to the Internet. This can be done if the organization or the corporation leases a high-speed WAN from a carrier provider and connects itself to a regional ISP. For example, a large university with several campuses can create an internetwork and then connect the internetwork to the Internet.

1.4 INTERNET HISTORY

Now that we have given an overview of the Internet, let us give a brief history of the Internet. This brief history makes it clear how the Internet has evolved from a private network to a global one in less than 40 years.

1.4.1 Early History

There were some communication networks, such as telegraph and telephone networks, before 1960. These networks were suitable for constant-rate communication at that time, which means that after a connection was made between two users, the encoded message (telegraphy) or voice (telephony) could be exchanged. A computer network, on the other hand, should be able to handle *bursty* data, which means data received at variable rates at different times. The world needed to wait for the packet-switched network to be invented.

Birth of Packet-Switched Networks

The theory of packet switching for bursty traffic was first presented by Leonard Kleinrock in 1961 at MIT. At the same time, two other researchers, Paul Baran at Rand Institute and Donald Davies at National Physical Laboratory in England, published some papers about packet-switched networks.

ARPANET

In the mid-1960s, mainframe computers in research organizations were stand-alone devices. Computers from different manufacturers were unable to communicate with one another. The **Advanced Research Projects Agency (ARPA)** in the Department of Defense (DOD) was interested in finding a way to connect computers so that the researchers they funded could share their findings, thereby reducing costs and eliminating duplication of effort.

In 1967, at an Association for Computing Machinery (ACM) meeting, ARPA presented its ideas for the **Advanced Research Projects Agency Network (ARPANET),** a small network of connected computers. The idea was that each host computer (not necessarily from the same manufacturer) would be attached to a specialized computer, called an *interface message processor* (IMP). The IMPs, in turn, would be connected to each other. Each IMP had to be able to communicate with other IMPs as well as with its own attached host.

By 1969, ARPANET was a reality. Four nodes, at the University of California at Los Angeles (UCLA), the University of California at Santa Barbara (UCSB), Stanford Research Institute (SRI), and the University of Utah, were connected via the IMPs to form a network. Software called the *Network Control Protocol* (NCP) provided communication between the hosts.

1.4.2 Birth of the Internet

In 1972, Vint Cerf and Bob Kahn, both of whom were part of the core ARPANET group, collaborated on what they called the *Internetting Project*. They wanted to link dissimilar networks so that a host on one network could communicate with a host on another. There were many problems to overcome: diverse packet sizes, diverse interfaces, and diverse transmission rates, as well as differing reliability requirements. Cerf and Kahn devised the idea of a device called a *gateway* to serve as the intermediary hardware to transfer data from one network to another.

TCP/IP

Cerf and Kahn's landmark 1973 paper outlined the protocols to achieve end-to-end delivery of data. This was a new version of NCP. This paper on transmission control protocol (TCP) included concepts such as encapsulation, the datagram, and the functions of a gateway. A radical idea was the transfer of responsibility for error correction from the IMP to the host machine. This ARPA Internet now became the focus of the communication effort. Around this time, responsibility for the ARPANET was handed over to the Defense Communication Agency (DCA).

In October 1977, an internet consisting of three different networks (ARPANET, packet radio, and packet satellite) was successfully demonstrated. Communication between networks was now possible.

Shortly thereafter, authorities made a decision to split TCP into two protocols: **Transmission Control Protocol (TCP)** and **Internet Protocol (IP).** IP would handle datagram routing while TCP would be responsible for higher level functions such as segmentation, reassembly, and error detection. The new combination became known as TCP/IP.

In 1981, under a Defence Department contract, UC Berkeley modified the UNIX operating system to include TCP/IP. This inclusion of network software along with a popular operating system did much for the popularity of internetworking. The open (non-manufacturer-specific) implementation of the Berkeley UNIX gave every manufacturer a working code base on which they could build their products.

In 1983, authorities abolished the original ARPANET protocols, and TCP/IP became the official protocol for the ARPANET. Those who wanted to use the Internet to access a computer on a different network had to be running TCP/IP.

MILNET

In 1983, ARPANET split into two networks: **Military Network (MILNET)** for military users and ARPANET for nonmilitary users.

CSNET

Another milestone in Internet history was the creation of CSNET in 1981. **Computer Science Network (CSNET)** was a network sponsored by the National Science Foundation (NSF). The network was conceived by universities that were ineligible to join ARPANET due to an absence of ties to the Department of Defense. CSNET was a less expensive network; there were no redundant links and the transmission rate was slower.

By the mid-1980s, most U.S. universities with computer science departments were part of CSNET. Other institutions and companies were also forming their own networks and using TCP/IP to interconnect. The term *Internet,* originally associated with government-funded connected networks, now referred to the connected networks using TCP/IP protocols.

NSFNET

With the success of CSNET, the NSF in 1986 sponsored the **National Science Foundation Network (NSFNET),** a backbone that connected five supercomputer centers located throughout the United States. Community networks were allowed access to this backbone, a T-1 line (see Chapter 6) with a 1.544-Mbps data rate, thus providing connectivity throughout the United States. In 1990, ARPANET was officially retired and replaced by NSFNET. In 1995, NSFNET reverted back to its original concept of a research network.

ANSNET

In 1991, the U.S. government decided that NSFNET was not capable of supporting the rapidly increasing Internet traffic. Three companies, IBM, Merit, and Verizon, filled the void by forming a nonprofit organization called Advanced Network & Services (ANS) to build a new, high-speed Internet backbone called **Advanced Network Services Network (ANSNET).**

1.4.3 Internet Today

Today, we witness a rapid growth both in the infrastructure and new applications. The Internet today is a set of pier networks that provide services to the whole world. What has made the Internet so popular is the invention of new applications.

World Wide Web

The 1990s saw the explosion of Internet applications due to the emergence of the World Wide Web (WWW). The Web was invented at CERN by Tim Berners-Lee. This invention has added the commercial applications to the Internet.

Multimedia

Recent developments in the multimedia applications such as voice over IP (telephony), video over IP (Skype), view sharing (YouTube), and television over IP (PPLive) has increased the number of users and the amount of time each user spends on the network. We discuss multimedia in Chapter 28.

Peer-to-Peer Applications

Peer-to-peer networking is also a new area of communication with a lot of potential. We introduce some peer-to-peer applications in Chapter 29.

1.5 STANDARDS AND ADMINISTRATION

In the discussion of the Internet and its protocol, we often see a reference to a standard or an administration entity. In this section, we introduce these standards and administration entities for those readers that are not familiar with them; the section can be skipped if the reader is familiar with them.

1.5.1 Internet Standards

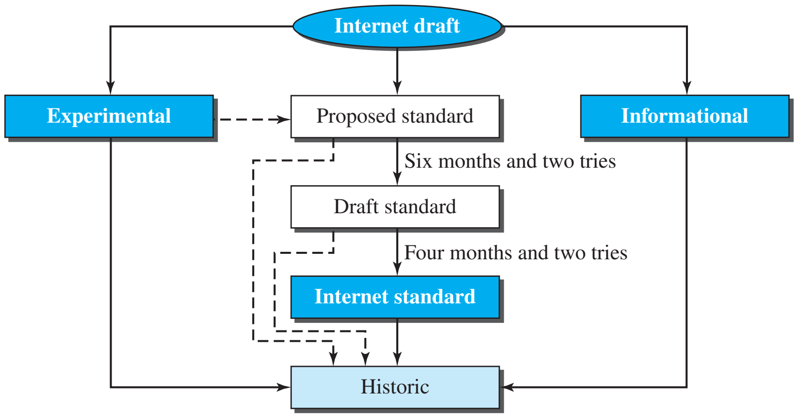
An **Internet standard** is a thoroughly tested specification that is useful to and adhered to by those who work with the Internet. It is a formalized regulation that must be followed. There is a strict procedure by which a specification attains Internet standard status. A specification begins as an Internet draft. An **Internet draft** is a working document (a work in progress) with no official status and a six-month lifetime. Upon recommendation from the Internet authorities, a draft may be published as a **Request for Comment (RFC).** Each RFC is edited, assigned a number, and made available to all interested parties. RFCs go through maturity levels and are categorized according to their requirement level.

Maturity Levels

An RFC, during its lifetime, falls into one of six *maturity levels:* proposed standard, draft standard, Internet standard, historic, experimental, and informational (see Figure 1.16).

- ❑ ***Proposed Standard.*** A proposed standard is a specification that is stable, well understood, and of sufficient interest to the Internet community. At this level, the specification is usually tested and implemented by several different groups.

Figure 1.16 *Maturity levels of an RFC*



- ❑ ***Draft Standard.*** A proposed standard is elevated to draft standard status after at least two successful independent and interoperable implementations. Barring difficulties, a draft standard, with modifications if specific problems are encountered, normally becomes an Internet standard.
- ❑ ***Internet Standard.*** A draft standard reaches Internet standard status after demonstrations of successful implementation.
- ❑ ***Historic.*** The historic RFCs are significant from a historical perspective. They either have been superseded by later specifications or have never passed the necessary maturity levels to become an Internet standard.
- ❑ ***Experimental.*** An RFC classified as experimental describes work related to an experimental situation that does not affect the operation of the Internet. Such an RFC should not be implemented in any functional Internet service.
- ❑ ***Informational.*** An RFC classified as informational contains general, historical, or tutorial information related to the Internet. It is usually written by someone in a non-Internet organization, such as a vendor.

Requirement Levels

RFCs are classified into five *requirement levels:* required, recommended, elective, limited use, and not recommended.

- ❑ ***Required.*** An RFC is labeled *required* if it must be implemented by all Internet systems to achieve minimum conformance. For example, IP and ICMP (Chapter 19) are required protocols.
- ❑ ***Recommended.*** An RFC labeled recommended is not required for minimum conformance; it is recommended because of its usefulness. For example, FTP (Chapter 26) and TELNET (Chapter 26) are recommended protocols.
- ❑ ***Elective.*** An RFC labeled elective is not required and not recommended. However, a system can use it for its own benefit.

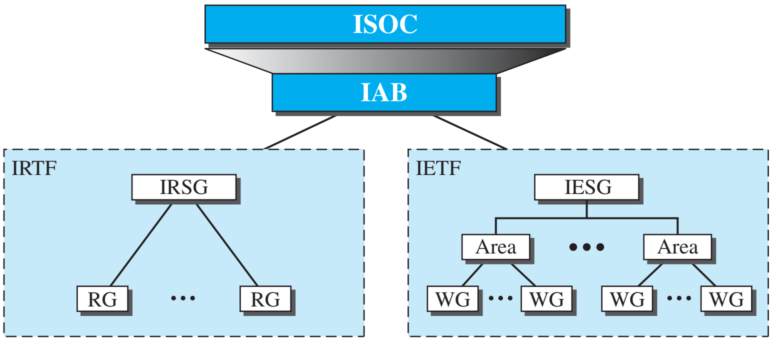
- ❑ ***Limited Use.*** An RFC labeled limited use should be used only in limited situations. Most of the experimental RFCs fall under this category.
- ❑ ***Not Recommended.*** An RFC labeled not recommended is inappropriate for general use. Normally a historic (deprecated) RFC may fall under this category.

RFCs can be found at http://www.rfc-editor.org.

1.5.2 Internet Administration

The Internet, with its roots primarily in the research domain, has evolved and gained a broader user base with significant commercial activity. Various groups that coordinate Internet issues have guided this growth and development. Appendix G gives the addresses, e-mail addresses, and telephone numbers for some of these groups. Figure 1.17 shows the general organization of Internet administration.

Figure 1.17 *Internet administration*



ISOC

The **Internet Society (ISOC)** is an international, nonprofit organization formed in 1992 to provide support for the Internet standards process. ISOC accomplishes this through maintaining and supporting other Internet administrative bodies such as IAB, IETF, IRTF, and IANA (see the following sections). ISOC also promotes research and other scholarly activities relating to the Internet.

IAB

The **Internet Architecture Board (IAB)** is the technical advisor to the ISOC. The main purposes of the IAB are to oversee the continuing development of the TCP/IP Protocol Suite and to serve in a technical advisory capacity to research members of the Internet community. IAB accomplishes this through its two primary components, the Internet Engineering Task Force (IETF) and the Internet Research Task Force (IRTF). Another responsibility of the IAB is the editorial management of the RFCs, described earlier. IAB is also the external liaison between the Internet and other standards organizations and forums.

IETF

The **Internet Engineering Task Force (IETF)** is a forum of working groups managed by the Internet Engineering Steering Group (IESG). IETF is responsible for identifying operational problems and proposing solutions to these problems. IETF also develops and reviews specifications intended as Internet standards. The working groups are collected into areas, and each area concentrates on a specific topic. Currently nine areas have been defined. The areas include applications, protocols, routing, network management next generation (IPng), and security.

IRTF

The **Internet Research Task Force (IRTF)** is a forum of working groups managed by the Internet Research Steering Group (IRSG). IRTF focuses on long-term research topics related to Internet protocols, applications, architecture, and technology.

1.6 END-CHAPTER MATERIALS

1.6.1 Recommended Reading

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

Books

The introductory materials covered in this chapter can be found in [Sta04] and [PD03]. [Tan03] also discusses standardization.

1.6.2 Key Terms

Advanced Network Services Network (ANSNET)
Advanced Research Projects Agency (ARPA)
Advanced Research Projects Agency Network (ARPANET)
American Standard Code for Information Interchange (ASCII)
audio
backbone
Basic Latin
bus topology
circuit-switched network
code
Computer Science Network (CSNET)
data
data communications
delay
full-duplex mode
half-duplex mode
hub
image
internet
Internet
Internet Architecture Board (IAB)
Internet draft
Internet Engineering Task Force (IETF)
Internet Research Task Force (IRTF)
Internet Service Provider (ISP)
Internet Society (ISOC)
Internet standard
internetwork
local area network (LAN)
mesh topology
message

Military Network (MILNET)
multipoint or multidrop connection
National Science Foundation Network (NSFNET)
network
node
packet
packet-switched network
performance
physical topology
point-to-point connection
protocol
Request for Comment (RFC)
RGB
ring topology
simplex mode
star topology
switched network
TCP/IP protocol suite
telecommunication
throughput
Transmission Control Protocol/ Internet Protocol (TCP/IP)
transmission medium
Unicode
video
wide area network (WAN)
YCM

1.6.3 Summary

Data communications are the transfer of data from one device to another via some form of transmission medium. A data communications system must transmit data to the correct destination in an accurate and timely manner. The five components that make up a data communications system are the message, sender, receiver, medium, and protocol. Text, numbers, images, audio, and video are different forms of information. Data flow between two devices can occur in one of three ways: simplex, half-duplex, or full-duplex.

A network is a set of communication devices connected by media links. In a point-to-point connection, two and only two devices are connected by a dedicated link. In a multipoint connection, three or more devices share a link. Topology refers to the physical or logical arrangement of a network. Devices may be arranged in a mesh, star, bus, or ring topology.

A network can be categorized as a local area network or a wide area network. A LAN is a data communication system within a building, plant, or campus, or between nearby buildings. A WAN is a data communication system spanning states, countries, or the whole world. An internet is a network of networks. The Internet is a collection of many separate networks.

The Internet history started with the theory of packet switching for bursty traffic. The history continued when The ARPA was interested in finding a way to connect computers so that the researchers they funded could share their findings, resulting in the creation of ARPANET. The Internet was born when Cerf and Kahn devised the idea of a device called a *gateway* to serve as the intermediary hardware to transfer data from one network to another. The TCP/IP protocol suite paved the way for creation of today's Internet. The invention of WWW, the use of multimedia, and peer-to-peer communication helps the growth of the Internet.

An Internet standard is a thoroughly tested specification. An Internet draft is a working document with no official status and a six-month lifetime. A draft may be published as a Request for Comment (RFC). RFCs go through maturity levels and are categorized according to their requirement level. The Internet administration has evolved with the Internet. ISOC promotes research and activities. IAB is the technical advisor to the ISOC. IETF is a forum of working groups responsible for operational problems. IRTF is a forum of working groups focusing on long-term research topics.

1.7 PRACTICE SET

1.7.1 Quizzes

A set of interactive quizzes for this chapter can be found on the book website. It is strongly recommended that the student take the quizzes to check his/her understanding of the materials before continuing with the practice set.

1.7.2 Questions

Q1-1. Identify the five components of a data communications system.

Q1-2. What are the three criteria necessary for an effective and efficient network?

Q1-3. What are the advantages of a multipoint connection over a point-to-point one?

Q1-4. What are the two types of line configuration?

Q1-5. Categorize the four basic topologies in terms of line configuration.

Q1-6. What is the difference between half-duplex and full-duplex transmission modes?

Q1-7. Name the four basic network topologies, and cite an advantage of each type.

Q1-8. For *n* devices in a network, what is the number of cable links required for a mesh, ring, bus, and star topology?

Q1-9. What are some of the factors that determine whether a communication system is a LAN or WAN?

Q1-10. What is an internet? What is the Internet?

Q1-11. Why are protocols needed?

Q1-12. In a LAN with a link-layer switch (Figure 1.8b), Host 1 wants to send a message to Host 3. Since communication is through the link-layer switch, does the switch need to have an address? Explain.

Q1-13. How many point-to-point WANs are needed to connect *n* LANs if each LAN should be able to directly communicate with any other LAN?

Q1-14. When we use local telephones to talk to a friend, are we using a circuit-switched network or a packet-switched network?

Q1-15. When a resident uses a dial-up or DLS service to connect to the Internet, what is the role of the telephone company?

Q1-16. What is the first principle we discussed in this chapter for protocol layering that needs to be followed to make the communication bidirectional?

Q1-17. Explain the difference between an Internet draft and a proposed standard.

Q1-18. Explain the difference between a required RFC and a recommended RFC.

Q1-19. Explain the difference between the duties of the IETF and IRTF.

1.7.3 Problems

P1-1. What is the maximum number of characters or symbols that can be represented by Unicode?

P1-2. A color image uses 16 bits to represent a pixel. What is the maximum number of different colors that can be represented?

P1-3. Assume six devices are arranged in a mesh topology. How many cables are needed? How many ports are needed for each device?

P1-4. For each of the following four networks, discuss the consequences if a connection fails.

a. Five devices arranged in a mesh topology

b. Five devices arranged in a star topology (not counting the hub)

c. Five devices arranged in a bus topology

d. Five devices arranged in a ring topology

P1-5. We have two computers connected by an Ethernet hub at home. Is this a LAN or a WAN? Explain the reason.

P1-6. In the ring topology in Figure 1.7, what happens if one of the stations is unplugged?

P1-7. In the bus topology in Figure 1.6, what happens if one of the stations is unplugged?

P1-8. Performance is inversely related to delay. When we use the Internet, which of the following applications are more sensitive to delay?

a. Sending an e-mail

b. Copying a file

c. Surfing the Internet

P1-9. When a party makes a local telephone call to another party, is this a point-to-point or multipoint connection? Explain the answer.

P1-10. Compare the telephone network and the Internet. What are the similarities? What are the differences?

1.8 SIMULATION EXPERIMENTS

1.8.1 Applets

One of the ways to show the network protocols in action or visually see the solution to some examples is through the use of interactive animation. We have created some Java applets to show some of the main concepts discussed in this chapter. It is strongly recommended that the students activate these applets on the book website and carefully examine the protocols in action. However, note that applets have been created only for some chapters, not all (see the book website).

1.8.2 Lab Assignments

Experiments with networks and network equipment can be done using at least two methods. In the first method, we can create an isolated networking laboratory and use networking hardware and software to simulate the topics discussed in each chapter. We can create an internet and send and receive packets from any host to another. The flow of packets can be observed and the performance can be measured. Although the first method is more effective and more instructional, it is expensive to implement and not all institutions are ready to invest in such an exclusive laboratory.

In the second method, we can use the Internet, the largest network in the world, as our virtual laboratory. We can send and receive packets using the Internet. The existence of some free-downloadable software allows us to capture and examine the packets exchanged. We can analyze the packets to see how theoretical aspects of networking are put into action. Although the second method may not be as effective as the first method, in that we cannot control and change the packet routes to see how the Internet behaves, the method is much cheaper to implement. It does not need a physical networking lab; it can be implemented using our desktop or laptop. The required software is also free to download.

There are many programs and utilities available for Windows and UNIX operating systems that allow us to sniff, capture, trace, and analyze packets that are exchanged between our computer and the Internet. Some of these, such as *Wireshark* and *Ping-Plotter,* have graphical user interface (GUI); others, such as *traceroute, nslookup*, *dig*, *ipconfig,* and *ifconfig,* are network administration command-line utilities. Any of these programs and utilities can be a valuable debugging tool for network administrators and educational tool for computer network students.

In this book, we mostly use Wireshark for lab assignments, although we occasionally use other tools. It captures live packet data from a network interface and displays them with detailed protocol information. Wireshark, however, is a passive analyzer. It only "measures" things from the network without manipulating them; it doesn't send packets on the network or perform other active operations. Wireshark is not an intrusion detection tool either. It does not give warning about any network intrusion. It, nevertheless, can help network administrators or network security engineers to figure out what is going on inside a network and to troubleshoot network problems. In addition to being an indispensable tool for network administrators and security engineers, Wireshark is a valuable tool for protocol developers, who may use it to debug protocol implementations, and a great educational tool for computer networking students who can use it to see details of protocol operations in real time. However, note that we can use lab assignments only with a few chapters.

Lab1-1. In this lab assignment we learn how to download and install Wireshark. The instructions for downloading and installing the software are posted on the book website in the lab section for Chapter 1. In this document, we also discuss the general idea behind the software, the format of its window, and how to use it. The full study of this lab prepares the student to use Wireshark in the lab assignments for other chapters.

CHAPTER 2

Network Models

The second chapter is a preparation for the rest of the book. The next five parts of the book is devoted to one of the layers in the TCP/IP protocol suite. In this chapter, we first discuss the idea of network models in general and the TCP/IP protocol suite in particular.

Two models have been devised to define computer network operations: the TCP/IP protocol suite and the OSI model. In this chapter, we first discuss a general subject, protocol layering, which is used in both models. We then concentrate on the TCP/IP protocol suite, on which the book is based. The OSI model is briefly discuss for comparison with the TCP/IP protocol suite.

❑ The first section introduces the concept of protocol layering using two scenarios. The section also discusses the two principles upon which the protocol layering is based. The first principle dictates that each layer needs to have two opposite tasks. The second principle dictates that the corresponding layers should be identical. The section ends with a brief discussion of logical connection between two identical layers in protocol layering. Throughout the book, we need to distinguish between logical and physical connections.

❑ The second section discusses the five layers of the TCP/IP protocol suite. We show how packets in each of the five layers (physical, data-link, network, transport, and application) are named. We also mention the addressing mechanism used in each layer. Each layer of the TCP/IP protocol suite is a subject of a part of the book. In other words, each layer is discussed in several chapters; this section is just an introduction and preparation.

❑ The third section gives a brief discussion of the OSI model. This model was never implemented in practice, but a brief discussion of the model and its comparison with the TCP/IP protocol suite may be useful to better understand the TCP/IP protocol suite. In this section we also give a brief reason for the OSI model's lack of success.

2.1 PROTOCOL LAYERING

We defined the term *protocol* in Chapter 1. In data communication and networking, a protocol defines the rules that both the sender and receiver and all intermediate devices need to follow to be able to communicate effectively. When communication is simple, we may need only one simple protocol; when the communication is complex, we may need to divide the task between different layers, in which case we need a protocol at each layer, or **protocol layering.**

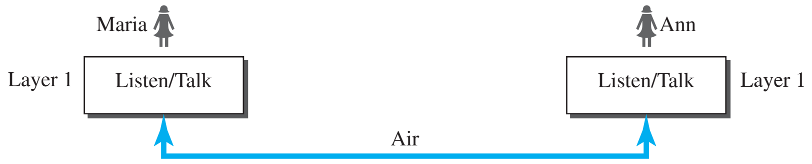
2.1.1 Scenarios

Let us develop two simple scenarios to better understand the need for protocol layering.

First Scenario

In the first scenario, communication is so simple that it can occur in only one layer. Assume Maria and Ann are neighbors with a lot of common ideas. Communication between Maria and Ann takes place in one layer, face to face, in the same language, as shown in Figure 2.1.

Figure 2.1 *A single-layer protocol*



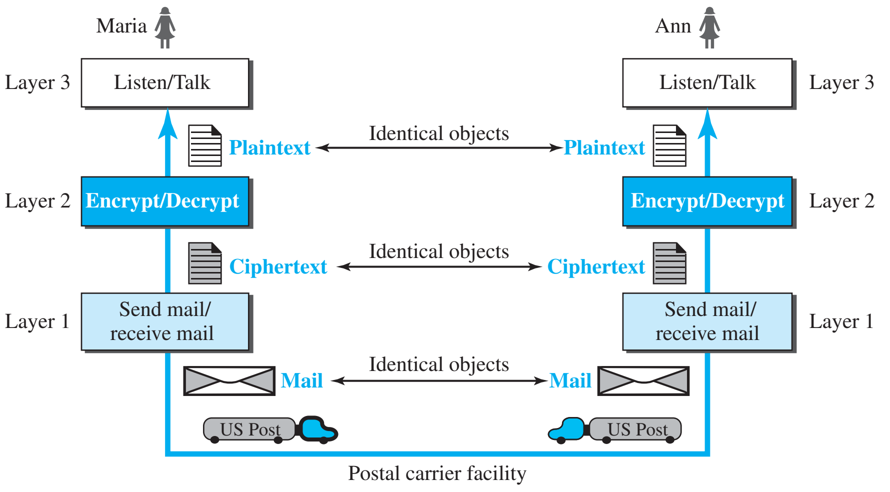
Even in this simple scenario, we can see that a set of rules needs to be followed. First, Maria and Ann know that they should greet each other when they meet. Second, they know that they should confine their vocabulary to the level of their friendship. Third, each party knows that she should refrain from speaking when the other party is speaking. Fourth, each party knows that the conversation should be a dialog, not a monolog: both should have the opportunity to talk about the issue. Fifth, they should exchange some nice words when they leave.

We can see that the protocol used by Maria and Ann is different from the communication between a professor and the students in a lecture hall. The communication in the second case is mostly monolog; the professor talks most of the time unless a student has a question, a situation in which the protocol dictates that she should raise her hand and wait for permission to speak. In this case, the communication is normally very formal and limited to the subject being taught.

Second Scenario

In the second scenario, we assume that Ann is offered a higher-level position in her company, but needs to move to another branch located in a city very far from Maria. The two friends still want to continue their communication and exchange ideas because they have come up with an innovative project to start a new business when they both retire. They decide to continue their conversation using regular mail through the post office. However, they do not want their ideas to be revealed by other people if the letters are intercepted. They agree on an encryption/decryption technique. The sender of the letter encrypts it to make it unreadable by an intruder; the receiver of the letter decrypts it to get the original letter. We discuss the encryption/decryption methods in Chapter 31, but for the moment we assume that Maria and Ann use one technique that makes it hard to decrypt the letter if one does not have the key for doing so. Now we can say that the communication between Maria and Ann takes place in three layers, as shown in Figure 2.2. We assume that Ann and Maria each have three machines (or robots) that can perform the task at each layer.

Figure 2.2 *A three-layer protocol*



Let us assume that Maria sends the first letter to Ann. Maria talks to the machine at the third layer as though the machine is Ann and is listening to her. The third layer machine listens to what Maria says and creates the plaintext (a letter in English), which is passed to the second layer machine. The second layer machine takes the plaintext, encrypts it, and creates the ciphertext, which is passed to the first layer machine. The first layer machine, presumably a robot, takes the ciphertext, puts it in an envelope, adds the sender and receiver addresses, and mails it.

At Ann's side, the first layer machine picks up the letter from Ann's mail box, recognizing the letter from Maria by the sender address. The machine takes out the ciphertext from the envelope and delivers it to the second layer machine. The second layer machine decrypts the message, creates the plaintext, and passes the plaintext to the third-layer machine. The third layer machine takes the plaintext and reads it as though Maria is speaking.

Protocol layering enables us to divide a complex task into several smaller and simpler tasks. For example, in Figure 2.2, we could have used only one machine to do the job of all three machines. However, if Maria and Ann decide that the encryption/decryption done by the machine is not enough to protect their secrecy, they would have to change the whole machine. In the present situation, they need to change only the second layer machine; the other two can remain the same. This is referred to as *modularity*. Modularity in this case means independent layers. A layer (module) can be defined as a black box with inputs and outputs, without concern about how inputs are changed to outputs. If two machines provide the same outputs when given the same inputs, they can replace each other. For example, Ann and Maria can buy the second layer machine from two different manufacturers. As long as the two machines create the same ciphertext from the same plaintext and vice versa, they do the job.

One of the advantages of protocol layering is that it allows us to separate the services from the implementation. A layer needs to be able to receive a set of services from the lower layer and to give the services to the upper layer; we don't care about how the layer is implemented. For example, Maria may decide not to buy the machine (robot) for the first layer; she can do the job herself. As long as Maria can do the tasks provided by the first layer, in both directions, the communication system works.

Another advantage of protocol layering, which cannot be seen in our simple examples but reveals itself when we discuss protocol layering in the Internet, is that communication does not always use only two end systems; there are intermediate systems that need only some layers, but not all layers. If we did not use protocol layering, we would have to make each intermediate system as complex as the end systems, which makes the whole system more expensive.

Is there any disadvantage to protocol layering? One can argue that having a single layer makes the job easier. There is no need for each layer to provide a service to the upper layer and give service to the lower layer. For example, Ann and Maria could find or build one machine that could do all three tasks. However, as mentioned above, if one day they found that their code was broken, each would have to replace the whole machine with a new one instead of just changing the machine in the second layer.

2.1.2 Principles of Protocol Layering

Let us discuss two principles of protocol layering.

First Principle

The first principle dictates that if we want bidirectional communication, we need to make each layer so that it is able to perform two opposite tasks, one in each direction. For example, the third layer task is to listen (in one direction) and *talk* (in the other direction). The second layer needs to be able to encrypt and decrypt. The first layer needs to send and receive mail.

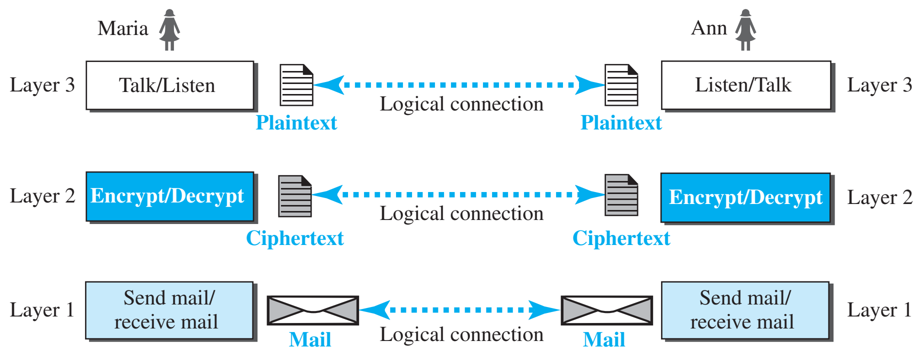
Second Principle

The second principle that we need to follow in protocol layering is that the two objects under each layer at both sites should be identical. For example, the object under layer 3 at both sites should be a plaintext letter. The object under layer 2 at both sites should be a ciphertext letter. The object under layer 1 at both sites should be a piece of mail.

2.1.3 Logical Connections

After following the above two principles, we can think about logical connection between each layer as shown in Figure 2.3. This means that we have layer-to-layer communication. Maria and Ann can think that there is a logical (imaginary) connection at each layer through which they can send the object created from that layer. We will see that the concept of logical connection will help us better understand the task of layering we encounter in data communication and networking.

Figure 2.3 *Logical connection between peer layers*



2.2 TCP/IP PROTOCOL SUITE

Now that we know about the concept of protocol layering and the logical communication between layers in our second scenario, we can introduce the TCP/IP (Transmission Control Protocol/Internet Protocol). TCP/IP is a protocol suite (a set of protocols organized in different layers) used in the Internet today. It is a hierarchical protocol made up of interactive modules, each of which provides a specific functionality. The term *hierarchical* means that each upper level protocol is supported by the services provided by one or more lower level protocols. The original TCP/IP protocol suite was defined as four software layers built upon the hardware. Today, however, TCP/IP is thought of as a five-layer model. Figure 2.4 shows both configurations.

2.2.1 Layered Architecture

To show how the layers in the TCP/IP protocol suite are involved in communication between two hosts, we assume that we want to use the suite in a small internet made up of three LANs (links), each with a link-layer switch. We also assume that the links are connected by one router, as shown in Figure 2.5.

Figure 2.4 *Layers in the TCP/IP protocol suite*

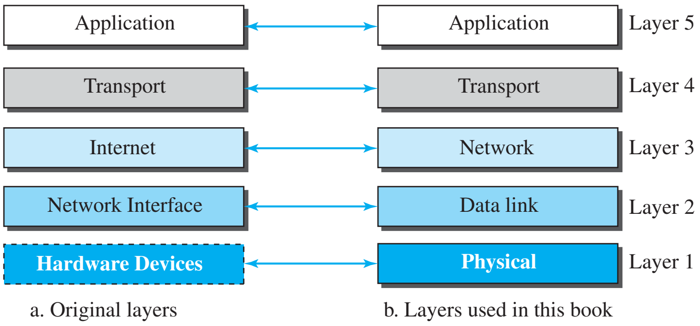
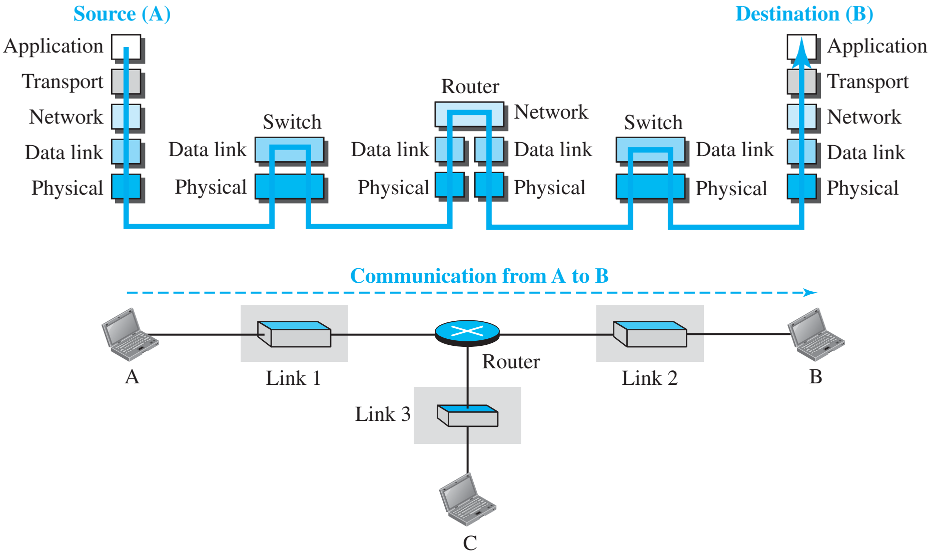


Figure 2.5 *Communication through an internet*



Let us assume that computer A communicates with computer B. As the figure shows, we have five communicating devices in this communication: source host (computer A), the link-layer switch in link 1, the router, the link-layer switch in link 2, and the destination host (computer B). Each device is involved with a set of layers depending on the role of the device in the internet. The two hosts are involved in all five layers; the source host needs to create a message in the application layer and send it down the layers so that it is physically sent to the destination host. The destination host needs to receive the communication at the physical layer and then deliver it through the other layers to the application layer.

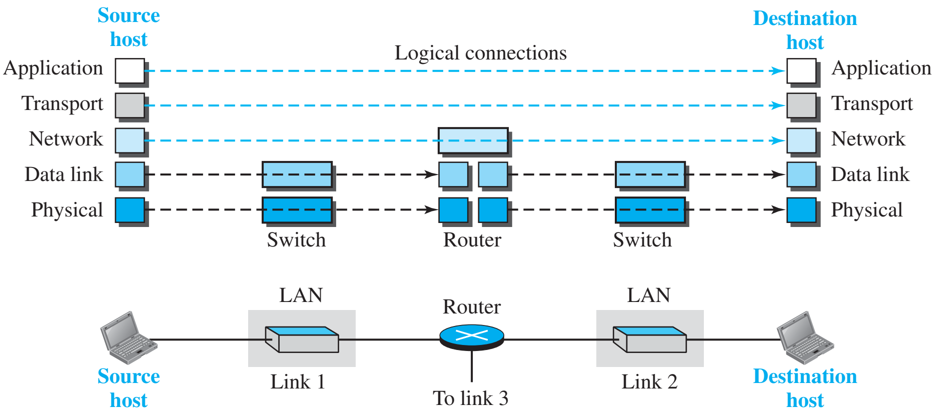
The router is involved in only three layers; there is no transport or application layer in a router as long as the router is used only for routing. Although a router is always involved in one network layer, it is involved in *n* combinations of link and physical layers in which *n* is the number of links the router is connected to. The reason is that each link may use its own data-link or physical protocol. For example, in the above figure, the router is involved in three links, but the message sent from source A to destination B is involved in two links. Each link may be using different link-layer and physical-layer protocols; the router needs to receive a packet from link 1 based on one pair of protocols and deliver it to link 2 based on another pair of protocols.

A link-layer switch in a link, however, is involved only in two layers, data-link and physical. Although each switch in the above figure has two different connections, the connections are in the same link, which uses only one set of protocols. This means that, unlike a router, a link-layer switch is involved only in one data-link and one physical layer.

2.2.2 Layers in the TCP/IP Protocol Suite

After the above introduction, we briefly discuss the functions and duties of layers in the TCP/IP protocol suite. Each layer is discussed in detail in the next five parts of the book. To better understand the duties of each layer, we need to think about the logical connections between layers. Figure 2.6 shows logical connections in our simple internet.

Figure 2.6 *Logical connections between layers of the TCP/IP protocol suite*

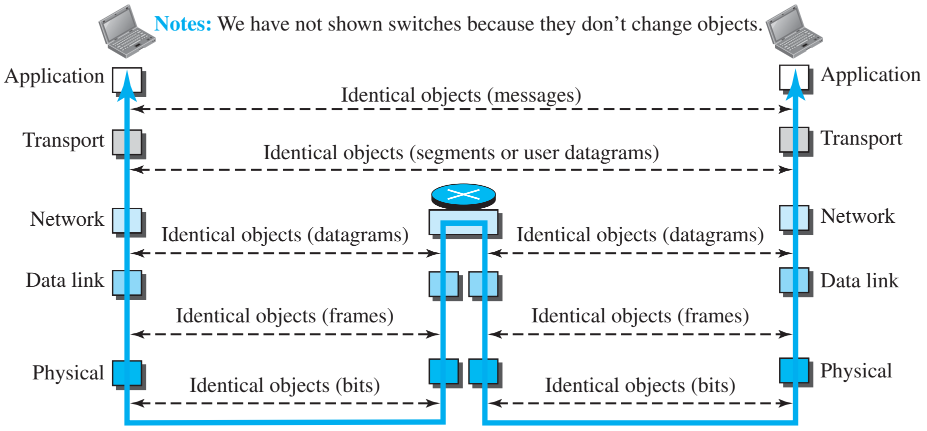


Using logical connections makes it easier for us to think about the duty of each layer. As the figure shows, the duty of the application, transport, and network layers is end-to-end. However, the duty of the data-link and physical layers is hop-to-hop, in which a hop is a host or router. In other words, the domain of duty of the top three layers is the internet, and the domain of duty of the two lower layers is the link.

Another way of thinking of the logical connections is to think about the data unit created from each layer. In the top three layers, the data unit (packets) should not be changed by any router or link-layer switch. In the bottom two layers, the packet created by the host is changed only by the routers, not by the link-layer switches.

Figure 2.7 shows the second principle discussed previously for protocol layering. We show the identical objects below each layer related to each device.

Figure 2.7 *Identical objects in the TCP/IP protocol suite*


Notes: We have not shown switches because they don't change objects.
Application
Application
Identical objects (messages)
Transport
Transport
Identical objects (segments or user datagrams)
Network
Network
Identical objects (datagrams)
Identical objects (datagrams)
Data link
Data link
Identical objects (frames)
Identical objects (frames)
Physical
Physical
Identical objects (bits)
Identical objects (bits)

Note that, although the logical connection at the network layer is between the two hosts, we can only say that identical objects exist between two hops in this case because a router may fragment the packet at the network layer and send more packets than received (see fragmentation in Chapter 19). Note that the link between two hops does not change the object.

2.2.3 Description of Each Layer

After understanding the concept of logical communication, we are ready to briefly discuss the duty of each layer. Our discussion in this chapter will be very brief, but we come back to the duty of each layer in next five parts of the book.

Physical Layer

We can say that the physical layer is responsible for carrying individual bits in a frame across the link. Although the physical layer is the lowest level in the TCP/IP protocol suite, the communication between two devices at the physical layer is still a logical communication because there is another, hidden layer, the transmission media, under the physical layer. Two devices are connected by a transmission medium (cable or air). We need to know that the transmission medium does not carry bits; it carries electrical or optical signals. So the bits received in a frame from the data-link layer are transformed and sent through the transmission media, but we can think that the logical unit between two physical layers in two devices is a *bit*. There are several protocols that transform a bit to a signal. We discuss them in Part II when we discuss the physical layer and the transmission media.

Data-link Layer

We have seen that an internet is made up of several links (LANs and WANs) connected by routers. There may be several overlapping sets of links that a datagram can travel from the host to the destination. The routers are responsible for choosing the *best* links. However, when the next link to travel is determined by the router, the data-link layer is responsible for taking the datagram and moving it across the link. The link can be a wired LAN with a link-layer switch, a wireless LAN, a wired WAN, or a wireless WAN. We can also have different protocols used with any link type. In each case, the data-link layer is responsible for moving the packet through the link.

TCP/IP does not define any specific protocol for the data-link layer. It supports all the standard and proprietary protocols. Any protocol that can take the datagram and carry it through the link suffices for the network layer. The data-link layer takes a datagram and encapsulates it in a packet called a *frame*.

Each link-layer protocol may provide a different service. Some link-layer protocols provide complete error detection and correction, some provide only error correction. We discuss wired links in Chapters 13 and 14 and wireless links in Chapters 15 and 16.

Network Layer

The network layer is responsible for creating a connection between the source computer and the destination computer. The communication at the network layer is host-to-host. However, since there can be several routers from the source to the destination, the routers in the path are responsible for choosing the best route for each packet. We can say that the network layer is responsible for host-to-host communication and routing the packet through possible routes. Again, we may ask ourselves why we need the network layer. We could have added the routing duty to the transport layer and dropped this layer. One reason, as we said before, is the separation of different tasks between different layers. The second reason is that the routers do not need the application and transport layers. Separating the tasks allows us to use fewer protocols on the routers.

The network layer in the Internet includes the main protocol, Internet Protocol (IP), that defines the format of the packet, called a datagram at the network layer. IP also defines the format and the structure of addresses used in this layer. IP is also responsible for routing a packet from its source to its destination, which is achieved by each router forwarding the datagram to the next router in its path.

IP is a connectionless protocol that provides no flow control, no error control, and no congestion control services. This means that if any of theses services is required for an application, the application should rely only on the transport-layer protocol. The network layer also includes unicast (one-to-one) and multicast (one-to-many) routing protocols. A routing protocol does not take part in routing (it is the responsibility of IP), but it creates forwarding tables for routers to help them in the routing process.

The network layer also has some auxiliary protocols that help IP in its delivery and routing tasks. The Internet Control Message Protocol (ICMP) helps IP to report some problems when routing a packet. The Internet Group Management Protocol (IGMP) is another protocol that helps IP in multitasking. The Dynamic Host Configuration Protocol (DHCP) helps IP to get the network-layer address for a host. The Address Resolution Protocol (ARP) is a protocol that helps IP to find the link-layer address of a host or a router when its network-layer address is given. ARP is discussed in Chapter 9, ICMP in Chapter 19, and IGMP in Chapter 21.

Transport Layer

The logical connection at the transport layer is also end-to-end. The transport layer at the source host gets the message from the application layer, encapsulates it in a transport-layer packet (called a *segment* or a *user datagram* in different protocols) and sends it, through the logical (imaginary) connection, to the transport layer at the destination host. In other words, the transport layer is responsible for giving services to the application layer: to get a message from an application program running on the source host and deliver it to the corresponding application program on the destination host. We may ask why we need an end-to-end transport layer when we already have an end-to-end application layer. The reason is the separation of tasks and duties, which we discussed earlier. The transport layer should be independent of the application layer. In addition, we will see that we have more than one protocol in the transport layer, which means that each application program can use the protocol that best matches its requirement.

As we said, there are a few transport-layer protocols in the Internet, each designed for some specific task. The main protocol, Transmission Control Protocol (TCP), is a connection-oriented protocol that first establishes a logical connection between transport layers at two hosts before transferring data. It creates a logical pipe between two TCPs for transferring a stream of bytes. TCP provides flow control (matching the sending data rate of the source host with the receiving data rate of the destination host to prevent overwhelming the destination), error control (to guarantee that the segments arrive at the destination without error and resending the corrupted ones), and congestion control to reduce the loss of segments due to congestion in the network. The other common protocol, User Datagram Protocol (UDP), is a connectionless protocol that transmits user datagrams without first creating a logical connection. In UDP, each user datagram is an independent entity without being related to the previous or the next one (the meaning of the term *connectionless*). UDP is a simple protocol that does not provide flow, error, or congestion control. Its simplicity, which means small overhead, is attractive to an application program that needs to send short messages and cannot afford the retransmission of the packets involved in TCP, when a packet is corrupted or lost. A new protocol, Stream Control Transmission Protocol (SCTP) is designed to respond to new applications that are emerging in the multimedia. We will discuss UDP, TCP, and SCTP in Chapter 24.

Application Layer

As Figure 2.6 shows, the logical connection between the two application layers is end-to-end. The two application layers exchange *messages* between each other as though there were a bridge between the two layers. However, we should know that the communication is done through all the layers.

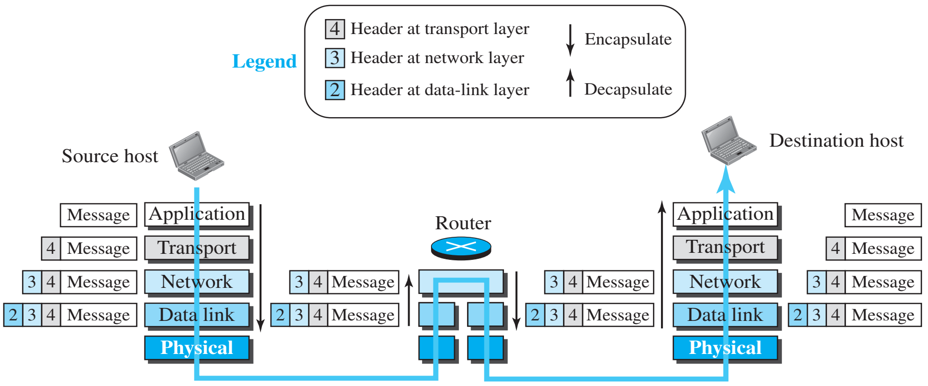
Communication at the application layer is between two *processes* (two programs running at this layer). To communicate, a process sends a request to the other process and receives a response. Process-to-process communication is the duty of the application layer. The application layer in the Internet includes many predefined protocols, but a user can also create a pair of processes to be run at the two hosts. In Chapter 25, we explore this situation.

The Hypertext Transfer Protocol (HTTP) is a vehicle for accessing the World Wide Web (WWW). The Simple Mail Transfer Protocol (SMTP) is the main protocol used in electronic mail (e-mail) service. The File Transfer Protocol (FTP) is used for transferring files from one host to another. The Terminal Network (TELNET) and Secure Shell (SSH) are used for accessing a site remotely. The Simple Network Management Protocol (SNMP) is used by an administrator to manage the Internet at global and local levels. The Domain Name System (DNS) is used by other protocols to find the network-layer address of a computer. The Internet Group Management Protocol (IGMP) is used to collect membership in a group. We discuss most of these protocols in Chapter 26 and some in other chapters.

2.2.4 Encapsulation and Decapsulation

One of the important concepts in protocol layering in the Internet is encapsulation/decapsulation. Figure 2.8 shows this concept for the small internet in Figure 2.5.

Figure 2.8 *Encapsulation/Decapsulation*



We have not shown the layers for the link-layer switches because no encapsulation/decapsulation occurs in this device. In Figure 2.8, we show the encapsulation in the source host, decapsulation in the destination host, and encapsulation and decapsulation in the router.

Encapsulation at the Source Host

At the source, we have only encapsulation.

1. At the application layer, the data to be exchanged is referred to as a *message*. A message normally does not contain any header or trailer, but if it does, we refer to the whole as the message. The message is passed to the transport layer.
2. The transport layer takes the message as the payload, the load that the transport layer should take care of. It adds the transport layer header to the payload, which contains the identifiers of the source and destination application programs that

want to communicate plus some more information that is needed for the end-to-end delivery of the message, such as information needed for flow, error control, or congestion control. The result is the transport-layer packet, which is called the *segment* (in TCP) and the *user datagram* (in UDP). The transport layer then passes the packet to the network layer.

3. The network layer takes the transport-layer packet as data or payload and adds its own header to the payload. The header contains the addresses of the source and destination hosts and some more information used for error checking of the header, fragmentation information, and so on. The result is the network-layer packet, called a *datagram*. The network layer then passes the packet to the data-link layer.
4. The data-link layer takes the network-layer packet as data or payload and adds its own header, which contains the link-layer addresses of the host or the next hop (the router). The result is the link-layer packet, which is called a *frame*. The frame is passed to the physical layer for transmission.

Decapsulation and Encapsulation at the Router

At the router, we have both decapsulation and encapsulation because the router is connected to two or more links.

1. After the set of bits are delivered to the data-link layer, this layer decapsulates the datagram from the frame and passes it to the network layer.
2. The network layer only inspects the source and destination addresses in the datagram header and consults its forwarding table to find the next hop to which the datagram is to be delivered. The contents of the datagram should not be changed by the network layer in the router unless there is a need to fragment the datagram if it is too big to be passed through the next link. The datagram is then passed to the data-link layer of the next link.
3. The data-link layer of the next link encapsulates the datagram in a frame and passes it to the physical layer for transmission.

Decapsulation at the Destination Host

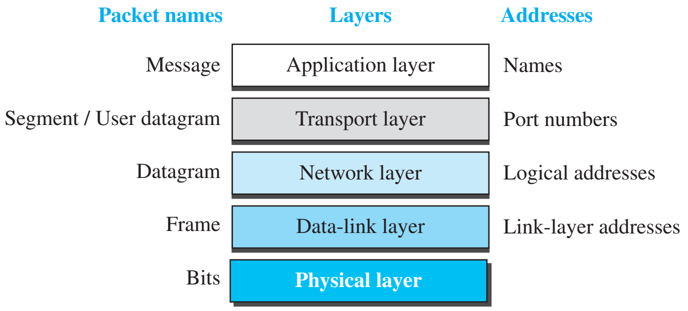
At the destination host, each layer only decapsulates the packet received, removes the payload, and delivers the payload to the next-higher layer protocol until the message reaches the application layer. It is necessary to say that decapsulation in the host involves error checking.

2.2.5 Addressing

It is worth mentioning another concept related to protocol layering in the Internet, *addressing*. As we discussed before, we have logical communication between pairs of layers in this model. Any communication that involves two parties needs two addresses: source address and destination address. Although it looks as if we need five pairs of addresses, one pair per layer, we normally have only four because the physical layer does not need addresses; the unit of data exchange at the physical layer is a bit, which definitely cannot have an address. Figure 2.9 shows the addressing at each layer.

As the figure shows, there is a relationship between the layer, the address used in that layer, and the packet name at that layer. At the application layer, we normally use names to define the site that provides services, such as *someorg.com*, or the e-mail

Figure 2.9 *Addressing in the TCP/IP protocol suite*



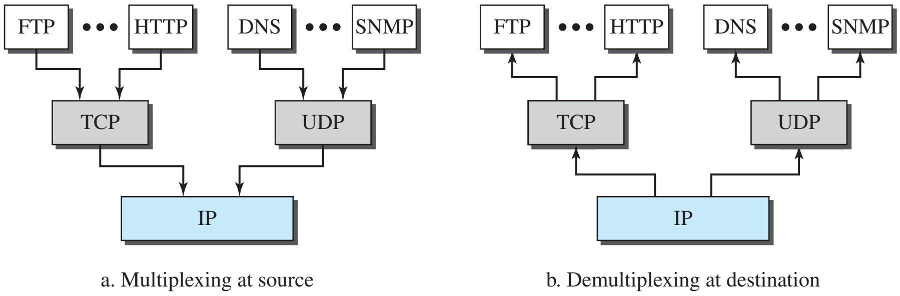
Packet names	Layers	Addresses
Message	Application layer	Names
Segment / User datagram	Transport layer	Port numbers
Datagram	Network layer	Logical addresses
Frame	Data-link layer	Link-layer addresses
Bits	Physical layer	

address, such as *somebody@coldmail.com.* At the transport layer, addresses are called port numbers, and these define the application-layer programs at the source and destination. Port numbers are local addresses that distinguish between several programs running at the same time. At the network-layer, the addresses are global, with the whole Internet as the scope. A network-layer address uniquely defines the connection of a device to the Internet. The link-layer addresses, sometimes called MAC addresses, are locally defined addresses, each of which defines a specific host or router in a network (LAN or WAN). We will come back to these addresses in future chapters.

2.2.6 Multiplexing and Demultiplexing

Since the TCP/IP protocol suite uses several protocols at some layers, we can say that we have multiplexing at the source and demultiplexing at the destination. Multiplexing in this case means that a protocol at a layer can encapsulate a packet from several next-higher layer protocols (one at a time); demultiplexing means that a protocol can decapsulate and deliver a packet to several next-higher layer protocols (one at a time). Figure 2.10 shows the concept of multiplexing and demultiplexing at the three upper layers.

Figure 2.10 *Multiplexing and demultiplexing*



To be able to multiplex and demultiplex, a protocol needs to have a field in its header to identify to which protocol the encapsulated packets belong. At the transport layer, either UDP or TCP can accept a message from several application-layer protocols. At the network layer, IP can accept a segment from TCP or a user datagram from UDP. IP can also accept a packet from other protocols such as ICMP, IGMP, and so on. At the data-link layer, a frame may carry the payload coming from IP or other protocols such as ARP (see Chapter 9).

2.3 THE OSI MODEL

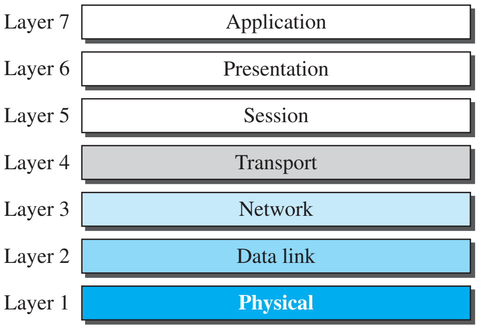
Although, when speaking of the Internet, everyone talks about the TCP/IP protocol suite, this suite is not the only suite of protocols defined. Established in 1947, the **International Organization for Standardization (ISO)** is a multinational body dedicated to worldwide agreement on international standards. Almost three-fourths of the countries in the world are represented in the ISO. An ISO standard that covers all aspects of network communications is the **Open Systems Interconnection (OSI) model.** It was first introduced in the late 1970s.

ISO is the organization; OSI is the model.

An *open system* is a set of protocols that allows any two different systems to communicate regardless of their underlying architecture. The purpose of the OSI model is to show how to facilitate communication between different systems without requiring changes to the logic of the underlying hardware and software. The OSI model is not a protocol; it is a model for understanding and designing a network architecture that is flexible, robust, and interoperable. The OSI model was intended to be the basis for the creation of the protocols in the OSI stack.

The OSI model is a layered framework for the design of network systems that allows communication between all types of computer systems. It consists of seven separate but related layers, each of which defines a part of the process of moving information across a network (see Figure 2.11).

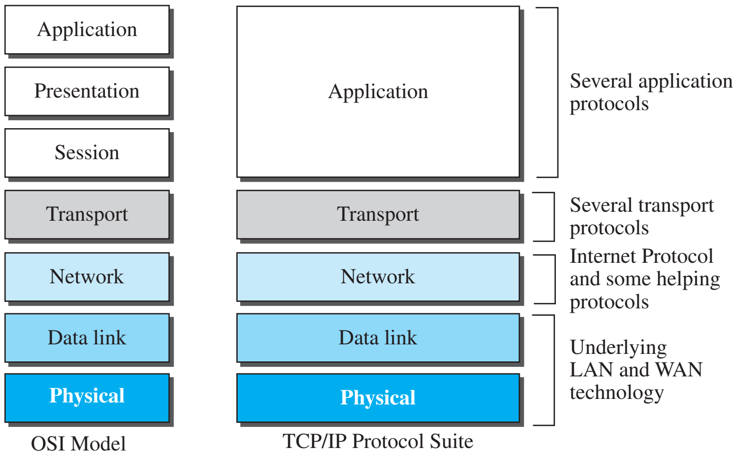
Figure 2.11 *The OSI model*



2.3.1 OSI versus TCP/IP

When we compare the two models, we find that two layers, session and presentation, are missing from the TCP/IP protocol suite. These two layers were not added to the TCP/IP protocol suite after the publication of the OSI model. The application layer in the suite is usually considered to be the combination of three layers in the OSI model, as shown in Figure 2.12.

Figure 2.12 *TCP/IP and OSI model*



Two reasons were mentioned for this decision. First, TCP/IP has more than one transport-layer protocol. Some of the functionalities of the session layer are available in some of the transport-layer protocols. Second, the application layer is not only one piece of software. Many applications can be developed at this layer. If some of the functionalities mentioned in the session and presentation layers are needed for a particular application, they can be included in the development of that piece of software.

2.3.2 Lack of OSI Model's Success

The OSI model appeared after the TCP/IP protocol suite. Most experts were at first excited and thought that the TCP/IP protocol would be fully replaced by the OSI model. This did not happen for several reasons, but we describe only three, which are agreed upon by all experts in the field. First, OSI was completed when TCP/IP was fully in place and a lot of time and money had been spent on the suite; changing it would cost a lot. Second, some layers in the OSI model were never fully defined. For example, although the services provided by the presentation and the session layers were listed in the document, actual protocols for these two layers were not fully defined, nor were they fully described, and the corresponding software was not fully developed. Third, when OSI was implemented by an organization in a different application, it did not show a high enough level of performance to entice the Internet authority to switch from the TCP/IP protocol suite to the OSI model.

2.4 END-CHAPTER MATERIALS

2.4.1 Recommended Reading

For more details about subjects discussed in this chapter, we recommend the following books, and RFCs. The items enclosed in brackets refer to the reference list at the end of the book.

Books and Papers

Several books and papers give a thorough coverage about the materials discussed in this chapter: [Seg 98], [Lei et al. 98], [Kle 04], [Cer 89], and [Jen et al. 86].

RFCs

Two RFCs in particular discuss the TCP/IP suite: RFC 791 (IP) and RFC 817 (TCP). In future chapters we list different RFCs related to each protocol in each layer.

2.4.2 Key Terms

International Organization for Standardization (ISO)
Open Systems Interconnection (OSI) model
protocol layering

2.4.3 Summary

A protocol is a set of rules that governs communication. In protocol layering, we need to follow two principles to provide bidirectional communication. First, each layer needs to perform two opposite tasks. Second, two objects under each layer at both sides should be identical. In a protocol layering, we need to distinguish between a logical connection and a physical connection. Two protocols at the same layer can have a logical connection; a physical connection is only possible through the physical layers.

TCP/IP is a hierarchical protocol suite made of five layers: physical, data link, network, transport, and application. The physical layer coordinates the functions required to transmit a bit stream over a physical medium. The data-link layer is responsible for delivering data units from one station to the next without errors. The network layer is responsible for the source-to-destination delivery of a packet across multiple network links. The transport layer is responsible for the process-to-process delivery of the entire message. The application layer enables the users to access the network.

Four levels of addresses are used in an internet following the TCP/IP protocols: physical (link) addresses, logical (IP) addresses, port addresses, and specific addresses. The physical address, also known as the link address, is the address of a node as defined by its LAN or WAN. The IP address uniquely defines a host on the Internet. The port address identifies a process on a host. A specific address is a user-friendly address.

Another model that defines protocol layering is the Open Systems Interconnection (OSI) model. Two layers in the OSI model, session and presentation, are missing from the TCP/IP protocol suite. These two layers were not added to the TCP/IP protocol suite after the publication of the OSI model. The application layer in the suite is usually considered to be the combination of three layers in the OSI model. The OSI model did not replace the TCP/IP protocol suite because it was completed when TCP/IP was fully in place and because some layers in the OSI model were never fully defined.

2.5 PRACTICE SET

2.5.1 Quizzes

A set of interactive quizzes for this chapter can be found on the book website. It is strongly recommended that the student take the quizzes to check his/her understanding of the materials before continuing with the practice set.

2.5.2 Questions

Q2-1. What is the first principle we discussed in this chapter for protocol layering that needs to be followed to make the communication bidirectional?

Q2-2. Which layers of the TCP/IP protocol suite are involved in a link-layer switch?

Q2-3. A router connects three links (networks). How many of each of the following layers can the router be involved with?

a. physical layer **b.** data-link layer **c.** network layer

Q2-4. In the TCP/IP protocol suite, what are the identical objects at the sender and the receiver sites when we think about the logical connection at the application layer?

Q2-5. A host communicates with another host using the TCP/IP protocol suite. What is the unit of data sent or received at each of the following layers?

a. application layer **b.** network layer **c.** data-link layer

Q2-6. Which of the following data units is encapsulated in a frame?

a. a user datagram **b.** a datagram **c.** a segment

Q2-7. Which of the following data units is decapsulated from a user datagram?

a. a datagram **b.** a segment **c.** a message

Q2-8. Which of the following data units has an application-layer message plus the header from layer 4?

a. a frame **b.** a user datagram **c.** a bit

Q2-9. List some application-layer protocols mentioned in this chapter.

Q2-10. If a port number is 16 bits (2 bytes), what is the minimum header size at the transport layer of the TCP/IP protocol suite?

Q2-11. What are the types of addresses (identifiers) used in each of the following layers?

a. application layer **b.** network layer **c.** data-link layer

Q2-12. When we say that the transport layer multiplexes and demultiplexes application-layer messages, do we mean that a transport-layer protocol can combine several messages from the application layer in one packet? Explain.

Q2-13. Can you explain why we did not mention multiplexing/demultiplexing services for the application layer?

Q2-14. Assume we want to connect two isolated hosts together to let each host communicate with the other. Do we need a link-layer switch between the two? Explain.

Q2-15. If there is a single path between the source host and the destination host, do we need a router between the two hosts?

2.5.3 Problems

P2-1. Answer the following questions about Figure 2.2 when the communication is from Maria to Ann:

a. What is the service provided by layer 1 to layer 2 at Maria's site?

b. What is the service provided by layer 1 to layer 2 at Ann's site?

P2-2. Answer the following questions about Figure 2.2 when the communication is from Maria to Ann:

a. What is the service provided by layer 2 to layer 3 at Maria's site?

b. What is the service provided by layer 2 to layer 3 at Ann's site?

P2-3. Assume that the number of hosts connected to the Internet at year 2010 is five hundred million. If the number of hosts increases only 20 percent per year, what is the number of hosts in year 2020?

P2-4. Assume a system uses five protocol layers. If the application program creates a message of 100 bytes and each layer (including the fifth and the first) adds a header of 10 bytes to the data unit, what is the efficiency (the ratio of application-layer bytes to the number of bytes transmitted) of the system?

P2-5. Assume we have created a packet-switched internet. Using the TCP/IP protocol suite, we need to transfer a huge file. What are the advantage and disadvantage of sending large packets?

P2-6. Match the following to one or more layers of the TCP/IP protocol suite:

a. route determination

b. connection to transmission media

c. providing services for the end user

P2-7. Match the following to one or more layers of the TCP/IP protocol suite:

a. creating user datagrams

b. responsibility for handling frames between adjacent nodes

c. transforming bits to electromagnetic signals

P2-8. In Figure 2.10, when the IP protocol decapsulates the transport-layer packet, how does it know to which upper-layer protocol (UDP or TCP) the packet should be delivered?

P2-9. Assume a private internet uses three different protocols at the data-link layer (L1, L2, and L3). Redraw Figure 2.10 with this assumption. Can we say that, in the data-link layer, we have demultiplexing at the source node and multiplexing at the destination node?

P2-10. Assume that a private internet requires that the messages at the application layer be encrypted and decrypted for security purposes. If we need to add some information about the encryption/decryption process (such as the algorithms used in the process), does it mean that we are adding one layer to the TCP/IP protocol suite? Redraw the TCP/IP layers (Figure 2.4 part b) if you think so.

P2-11. Protocol layering can be found in many aspects of our lives such as air travelling. Imagine you make a round-trip to spend some time on vacation at a resort. You need to go through some processes at your city airport before flying. You also need to go through some processes when you arrive at the resort airport. Show the protocol layering for the round trip using some layers such as baggage checking/claiming, boarding/unboarding, takeoff/landing.

P2-12. The presentation of data is becoming more and more important in today's Internet. Some people argue that the TCP/IP protocol suite needs to add a new layer to take care of the presentation of data. If this new layer is added in the future, where should its position be in the suite? Redraw Figure 2.4 to include this layer.

P2-13. In an internet, we change the LAN technology to a new one. Which layers in the TCP/IP protocol suite need to be changed?

P2-14. Assume that an application-layer protocol is written to use the services of UDP. Can the application-layer protocol uses the services of TCP without change?

P2-15. Using the internet in Figure 1.11 (Chapter 1) in the text, show the layers of the TCP/IP protocol suite and the flow of data when two hosts, one on the west coast and the other on the east coast, exchange messages.

PART II

Physical Layer

In the second part of the book, we discuss the physical layer, including the transmission media that is connected to the physical layer. The part is made of six chapters. The first introduces the entities involved in the physical layer. The next two chapters cover transmission. The following chapter discusses how to use the available bandwidth. The transmission media alone occupy all of the next chapter. Finally, the last chapter discusses switching, which can occur in any layer, but we introduce the topic in this part of the book.

https://hemanthrajhemu.github.io

CHAPTER 3

Introduction to Physical Layer

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. For transmission, data needs to be changed to **signals.**

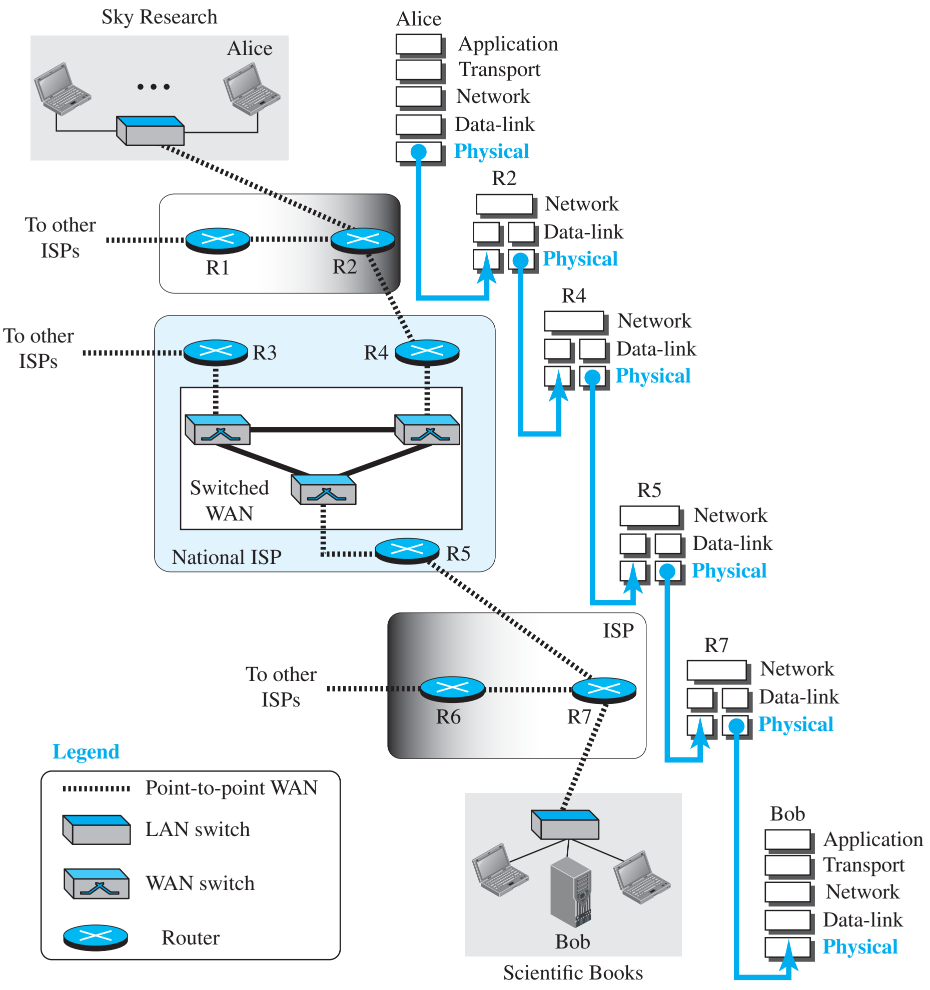
This chapter is divided into six sections:

- The first section shows how data and signals can be either analog or digital. Analog refers to an entity that is continuous; digital refers to an entity that is discrete.
- The second section shows that only periodic analog signals can be used in data communication. The section discusses simple and composite signals. The attributes of analog signals such as period, frequency, and phase are also explained.
- The third section shows that only nonperiodic digital signals can be used in data communication. The attributes of a digital signal such as bit rate and bit length are discussed. We also show how digital data can be sent using analog signals. Baseband and broadband transmission are also discussed in this section.
- The fourth section is devoted to transmission impairment. The section shows how attenuation, distortion, and noise can impair a signal.
- The fifth section discusses the data rate limit: how many bits per second we can send with the available channel. The data rates of noiseless and noisy channels are examined and compared.
- The sixth section discusses the performance of data transmission. Several channel measurements are examined including bandwidth, throughput, latency, and jitter. Performance is an issue that is revisited in several future chapters.

3.1 DATA AND SIGNALS

Figure 3.1 shows a scenario in which a scientist working in a research company, Sky Research, needs to order a book related to her research from an online bookseller, Scientific Books.

Figure 3.1 *Communication at the physical layer*



We can think of five different levels of communication between Alice, the computer on which our scientist is working, and Bob, the computer that provides online service. Communication at application, transport, network, or data-link is *logical*; communication at the physical layer is *physical*. For simplicity, we have shown only host-to-router, router-to-router, and router-to-host, but the switches are also involved in the physical communication.

Although Alice and Bob need to exchange *data*, communication at the physical layer means exchanging *signals*. Data need to be transmitted and received, but the media have to change data to signals. Both data and the signals that represent them can be either **analog** or **digital** in form.

3.1.1 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.

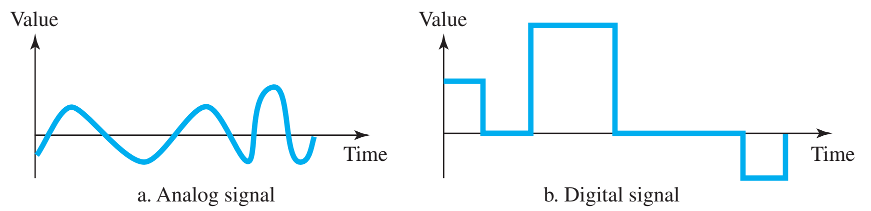
Digital data take on discrete values. For example, data are stored in computer memory in the form of 0s and 1s. They can be converted to a digital signal or modulated into an analog signal for transmission across a medium.

3.1.2 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 3.2 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.

Figure 3.2 *Comparison of analog and digital signals*



a. Analog signal

b. Digital signal

3.1.3 Periodic and Nonperiodic

Both analog and digital signals can take one of two forms: *periodic* or *nonperiodic* (sometimes referred to as *aperiodic;* 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 and nonperiodic digital signals, as we will see in future chapters.

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

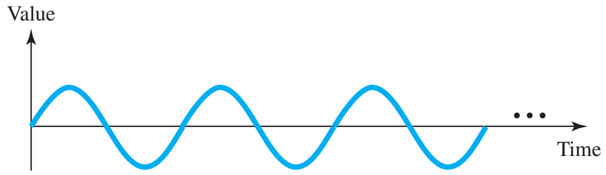
3.2 PERIODIC ANALOG SIGNALS

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.

3.2.1 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 3.3 shows a sine wave. Each cycle consists of a single arc above the time axis followed by a single arc below it.

Figure 3.3 *A sine wave*



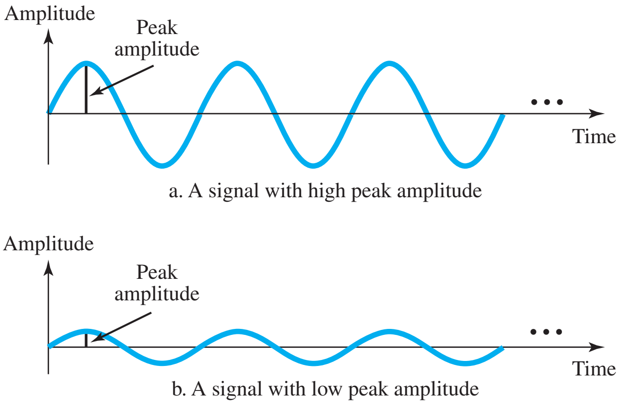
We discuss a mathematical approach to sine waves in Appendix E.

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 3.4 shows two signals and their peak amplitudes.

Figure 3.4 *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

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. 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 $2^{1/2}$ × rms value.

Example 3.2

The voltage of a 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.

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

Frequency and period are the inverse of each other.

Figure 3.5 shows two signals and their frequencies. 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.

Figure 3.5 *Two signals with the same amplitude and phase, but different frequencies*

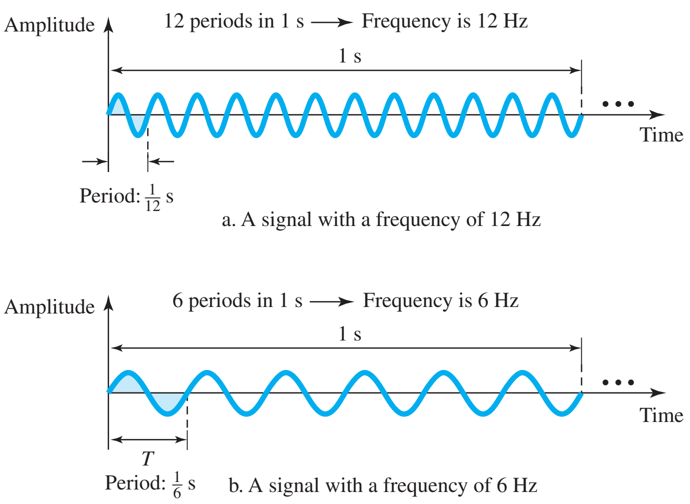

Amplitude
12 periods in 1 s → Frequency is 12 Hz
1 s
Time
Period: $\frac{1}{12}$ s
a. A signal with a frequency of 12 Hz
Amplitude
6 periods in 1 s → Frequency is 6 Hz
1 s
Time
T
Period: $\frac{1}{6}$ s
b. A signal with a frequency of 6 Hz

Table 3.1 *Units of period and frequency*

Period		*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:

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

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.

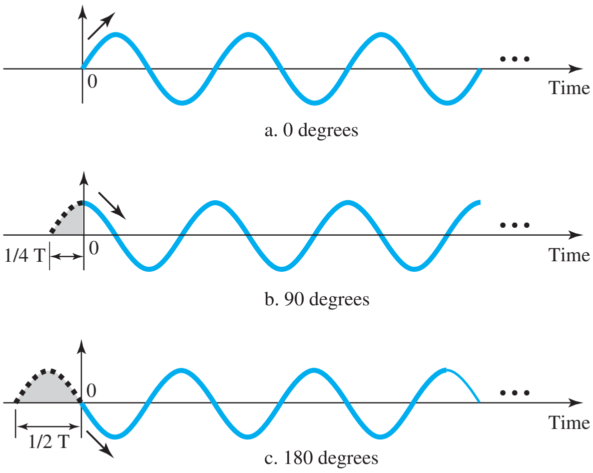
3.2.2 Phase

The term **phase,** or phase shift, 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 [360º is 2π rad; 1° is 2π/360 rad, and 1 rad is 360/(2π)]. A phase shift of 360º 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 90º corresponds to a shift of one-quarter of a period (see Figure 3.6).

Figure 3.6 *Three sine waves with the same amplitude and frequency, but different phases*



Looking at Figure 3.6, we can say that

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

b. A sine wave with a phase of 90° starts at time 0 with a peak amplitude. The amplitude is decreasing.

c. 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

a. A sine wave with a phase of 0° is not shifted.

b. 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.

c. 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

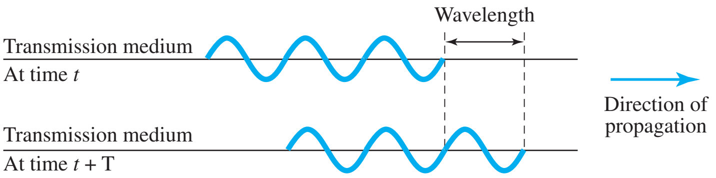
We know that 1 complete cycle is 360°. Therefore, $\frac{1}{6}$ cycle is

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

3.2.3 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 3.7).

Figure 3.7 *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

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

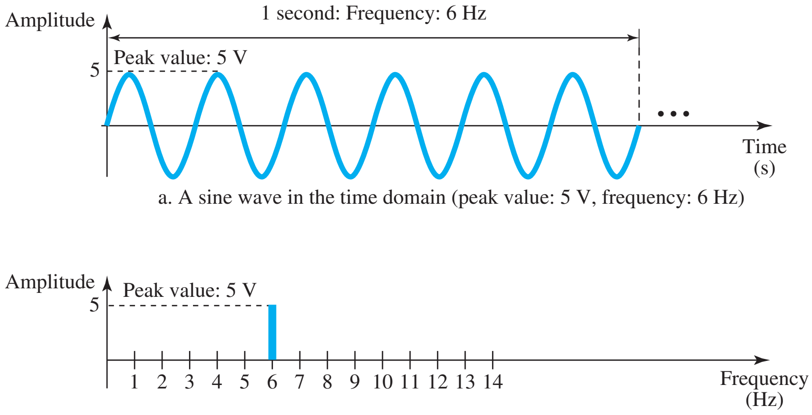
In a coaxial or fiber-optic cable, however, the wavelength is shorter (0.5 μm) because the propagation speed in the cable is decreased.

3.2.4 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 3.8 shows a signal in both the time and frequency domains.

Figure 3.8 *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)

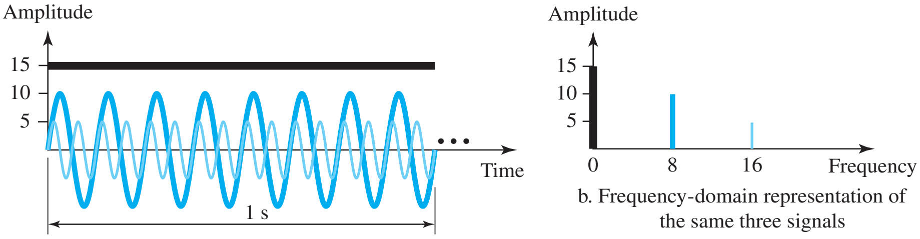
It is obvious that the frequency domain is easy to plot and conveys the information that one can find in a time domain plot. The advantage of the frequency domain is that we can immediately see the values of the frequency and peak amplitude. A complete sine wave is represented by one spike. The position 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 3.9 shows three sine waves, each with different amplitude and frequency. All can be represented by three spikes in the frequency domain.

Figure 3.9 *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

3.2.5 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 E; 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 E.

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 3.10 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.

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 3.11 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 3*f* is one-third of that of

Figure 3.10 *A composite periodic signal*

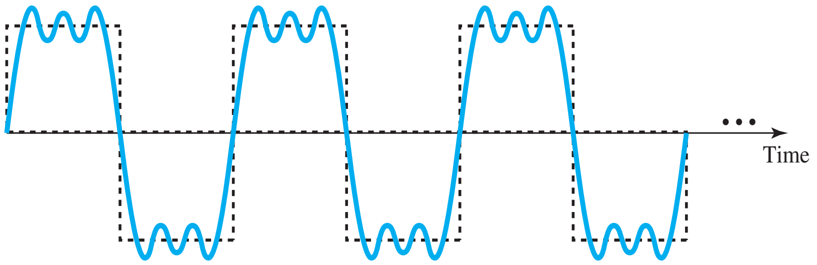
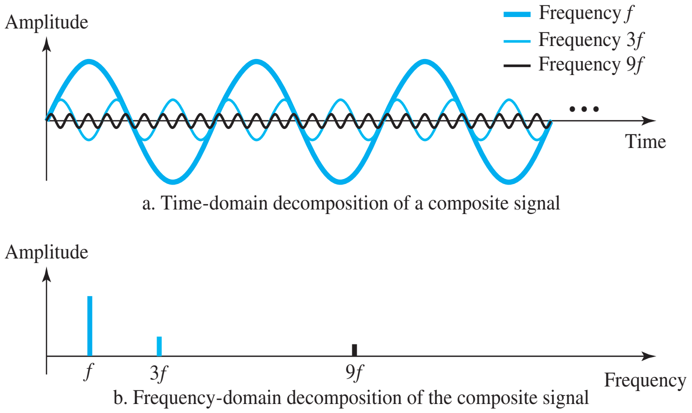


Figure 3.11 *Decomposition of a composite periodic signal in the time and frequency domains*



a. Time-domain decomposition of a composite signal

b. Frequency-domain decomposition of the composite signal

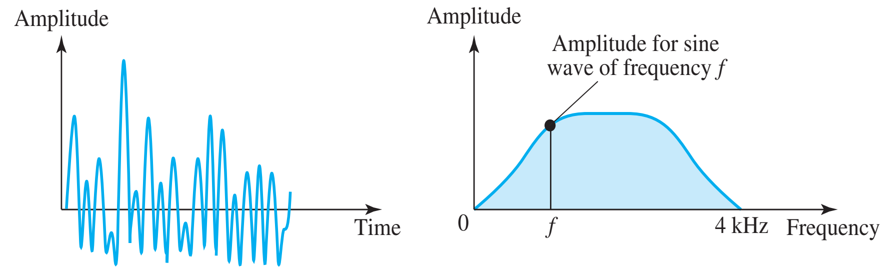
the first, and the amplitude of the sine wave with frequency 9*f* is one-ninth of the first. The frequency of the sine wave with frequency *f* is the same as the frequency of the composite signal; it is called the **fundamental frequency,** or first **harmonic.** The sine wave with frequency 3*f* has a frequency of 3 times the fundamental frequency; it is called the third harmonic. The third sine wave with frequency 9*f* 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*, 3*f*, and 9*f*. Because *f* is an integral number, 3*f* and 9*f* are also integral numbers. There are no frequencies such as 1.2*f* or 2.6*f*. The frequency domain of a periodic composite signal is always made of discrete spikes.

Example 3.9

Figure 3.12 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 3.12 *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.

3.2.6 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 3.13 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.

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.

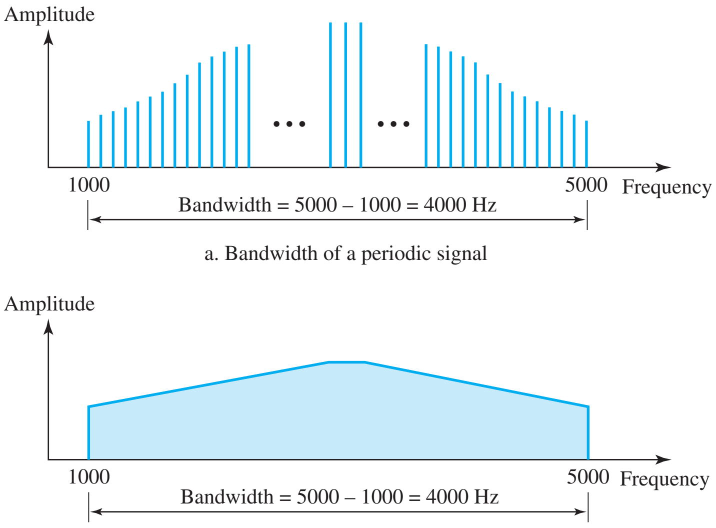
Solution

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

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

The spectrum has only five spikes, at 100, 300, 500, 700, and 900 Hz (see Figure 3.14).

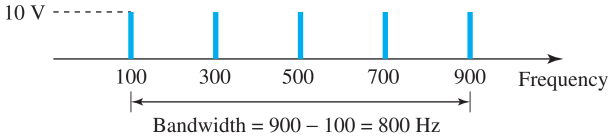
Figure 3.13 *The bandwidth of periodic and nonperiodic composite signals*



a. Bandwidth of a periodic signal

b. Bandwidth of a nonperiodic signal

Figure 3.14 *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

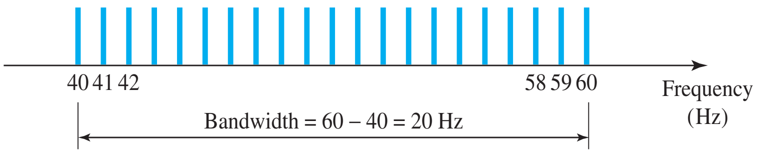
$$\mathbf{B = f_h - f_l \quad \longrightarrow \quad 20 = 60 - f_l \quad \longrightarrow \quad f_l = 60 - 20 = 40\ Hz}$$

The spectrum contains all integer frequencies. We show this by a series of spikes (see Figure 3.15).

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.

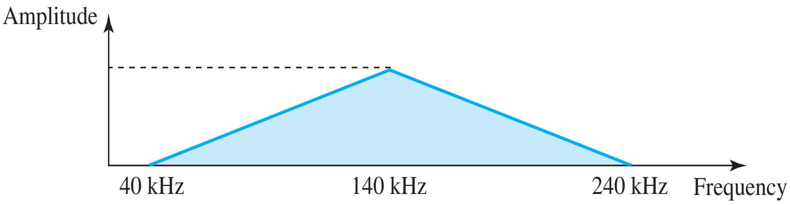
Figure 3.15 *The bandwidth for Example 3.11*



Solution

The lowest frequency must be at 40 kHz and the highest at 240 kHz. Figure 3.16 shows the frequency domain and the bandwidth.

Figure 3.16 *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

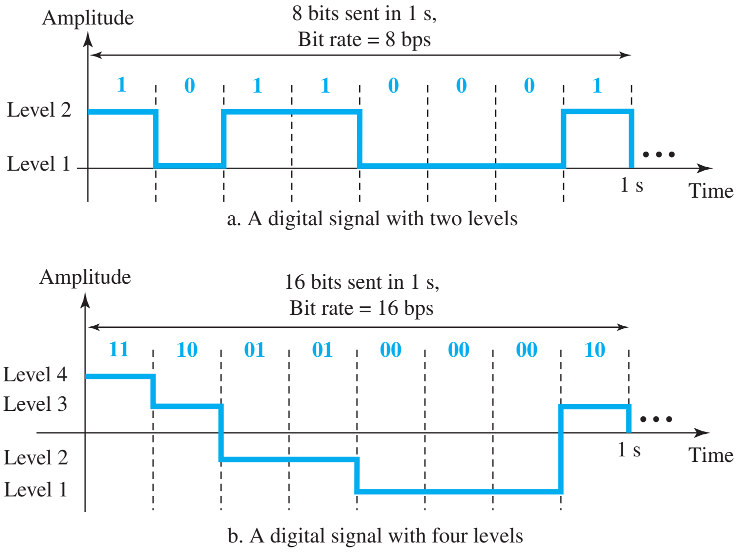
Another example of a nonperiodic composite signal is the signal received by an old-fashioned analog black-and-white TV. A TV screen is made up of pixels (picture elements) with each pixel being either white or black. The screen is scanned 30 times per second. (Scanning is actually 60 times per second, 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 × 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 3.17 shows two signals, one with two levels and the other with four. 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. For this reason, we can send $\log_2 4 = 2$ bits in part b.

Figure 3.17 *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

Example 3.16

A digital signal has eight levels. How many bits are needed per level? We calculate the number of bits from the following formula. Each signal level is represented by 3 bits.

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

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.

3.3.1 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 3.17 shows the bit rate for two signals.

Example 3.18

Assume we need to download text documents at the rate of 100 pages per second. 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

$$100 \times 24 \times 80 \times 8 = 1{,}536{,}000 \text{ bps} = 1.536 \text{ Mbps}$$

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

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

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

3.3.2 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.

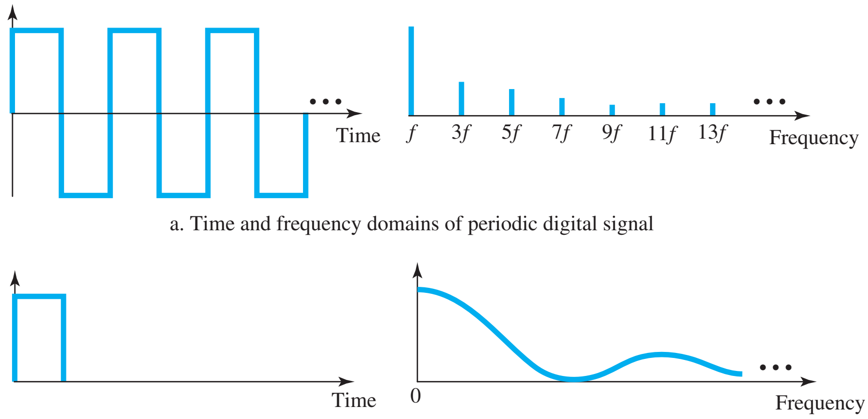
$$\textbf{Bit length} = \textbf{propagation speed} \times \textbf{bit duration}$$

3.3.3 Digital Signal as a Composite Analog Signal

Based on Fourier analysis (See Appendix E), 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 3.18 shows a periodic and a nonperiodic digital signal and their bandwidths.

Figure 3.18 *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.

3.3.4 Transmission of Digital Signals

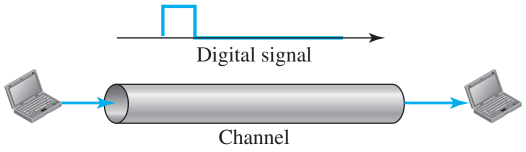
The previous discussion asserts that a digital signal, periodic or nonperiodic, is a composite analog signal with frequencies between zero and infinity. For the remainder of the discussion, let us consider the case of a nonperiodic digital signal, similar to the ones we encounter in data communications. 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).

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

Baseband Transmission

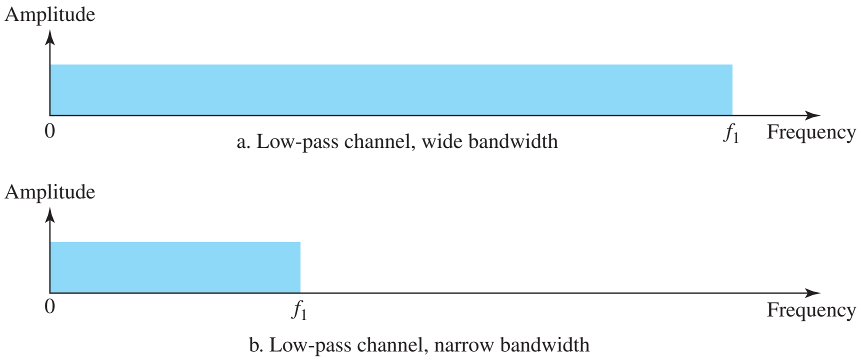
Baseband transmission means sending a digital signal over a channel without changing the digital signal to an analog signal. Figure 3.19 shows **baseband** transmission.

Figure 3.19 *Baseband transmission*



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 3.20 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 3.20 *Bandwidths of two low-pass channels*



a. Low-pass channel, wide bandwidth

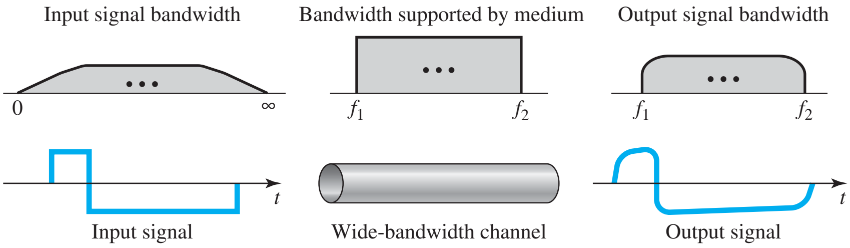
b. Low-pass channel, narrow bandwidth

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 or fiber optic cable, with a very wide bandwidth, two stations can communicate by using digital signals with very good accuracy, as shown in Figure 3.21. Note that f_1 is close to zero, and f_2 is very high.

Figure 3.21 *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 13.

Case 2: Low-Pass Channel with Limited Bandwidth

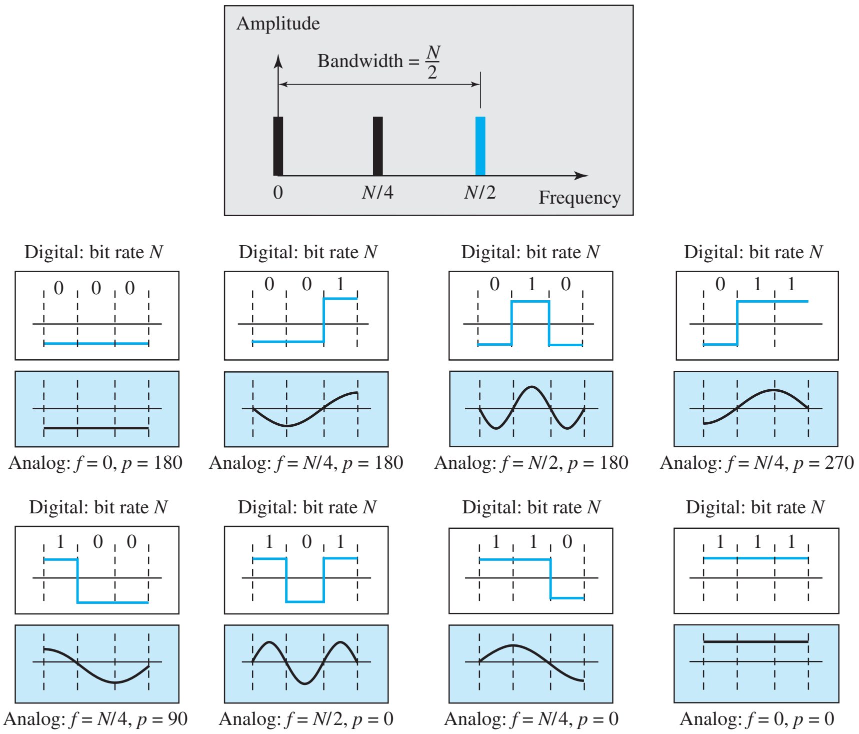
In a low-pass channel with limited bandwidth, we approximate the digital signal with an analog signal. The level of approximation depends on the bandwidth available.

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 3.22 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

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

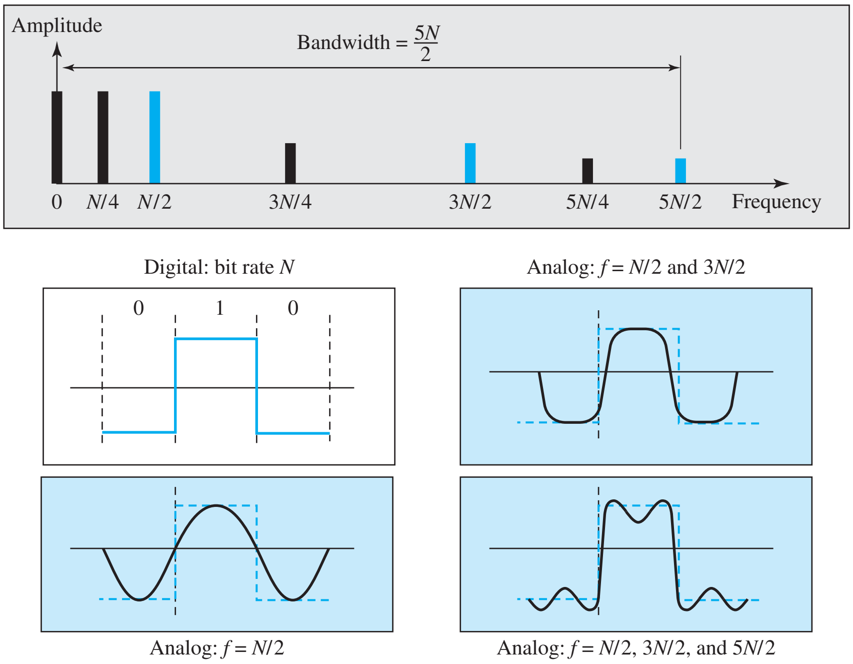
Figure 3.22 *Rough approximation of a digital signal using the first harmonic for worst case*



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 3.23 shows the effect of

Figure 3.23 *Simulating a digital signal with first three harmonics*



this increase for one of the worst cases, the pattern 010. 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. A still 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?

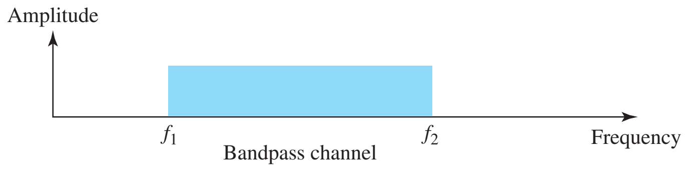
Solution

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 3.24 shows a bandpass channel.

Figure 3.24 *Bandwidth of a bandpass channel*

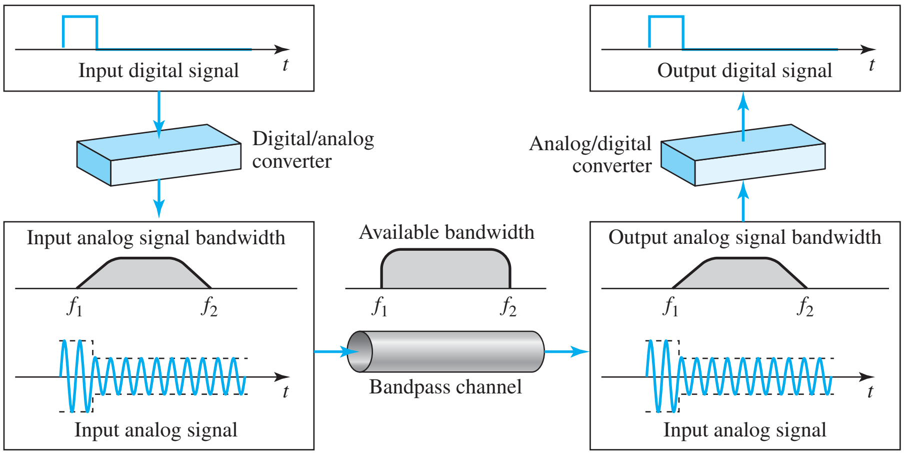


Note that a low-pass channel can be considered a bandpass channel with the lower frequency starting at zero.

Figure 3.25 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 3.25 *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.

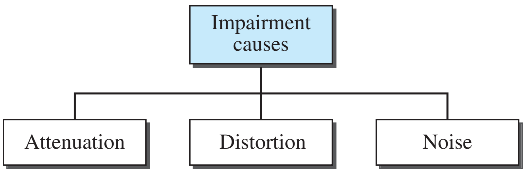
Example 3.25

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 3.26).

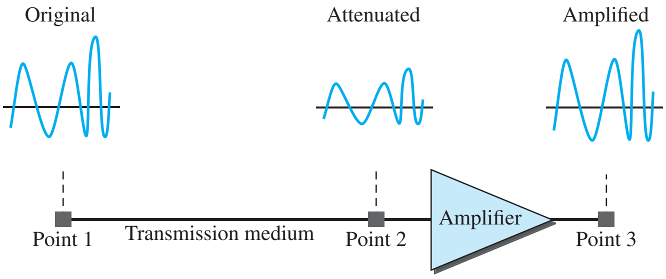
Figure 3.26 *Causes of impairment*



3.4.1 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 3.27 shows the effect of attenuation and amplification.

Figure 3.27 *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.

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

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

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

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

Example 3.27

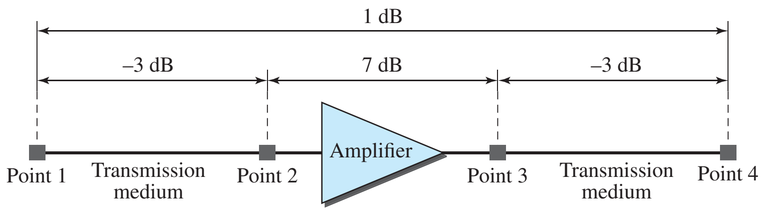
A signal travels through an amplifier, and its power is increased 10 times. This means that $P_2 = 10P_1$. In this case, the amplification (gain of power) can be calculated as

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

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 3.28 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 3.28 *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

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

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 × (−0.3) = −1.5 dB. We can calculate the power as

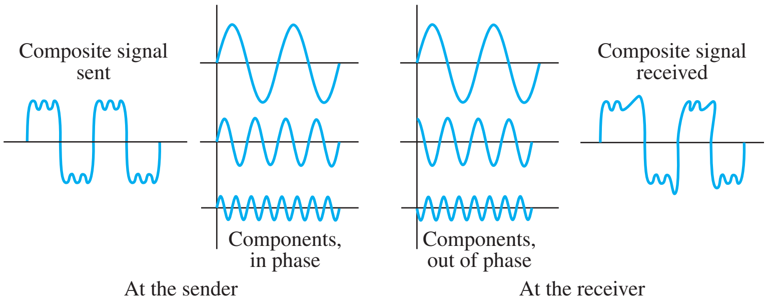
$$\mathbf{dB = 10 \log_{10} (P_2 / P_1) = -1.5 \quad \longrightarrow \quad (P_2 / P_1) = 10^{-0.15} = 0.71}$$

$$\mathbf{P_2 = 0.71P_1 = 0.7 \times 2\ mW = 1.4\ mW}$$

3.4.2 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 3.29 shows the effect of distortion on a composite signal.

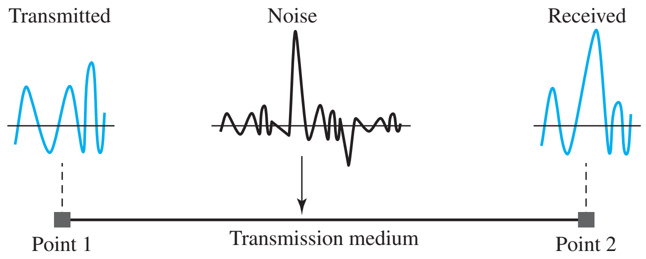
Figure 3.29 *Distortion*



3.4.3 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 appliancses. 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 3.30 shows the effect of noise on a signal. We discuss error in Chapter 10.

Figure 3.30 *Noise*



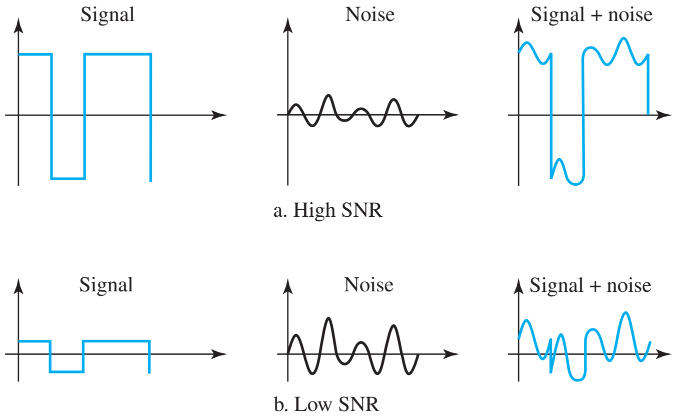
Signal-to-Noise Ratio (SNR)

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

$$\mathbf{SNR} = \frac{\textbf{average signal power}}{\textbf{average noise power}}$$

We need to consider the average signal power and the average noise power because these may change with time. Figure 3.31 shows the idea of SNR.

Figure 3.31 *Two cases of SNR: a high SNR and a low 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, $\mathrm{SNR_{dB}}$, defined as

$$\mathbf{SNR_{dB} = 10 \log_{10} SNR}$$

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}?

Solution

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

$$\textbf{SNR} = (10{,}000\ \mu\text{w}) / (1\ \mu\text{w}) = 10{,}000 \qquad \textbf{SNR}_{\text{dB}} = 10 \log_{10} 10{,}000 = 10 \log_{10} 10^4 = 40$$

Example 3.32

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

$$\textbf{SNR} = (\text{signal power}) / 0 = \infty \quad \longrightarrow \quad \textbf{SNR}_{\text{dB}} = 10 \log_{10} \infty = \infty$$

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.

3.5.1 Noiseless Channel: Nyquist Bit Rate

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

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

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

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

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

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

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:

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

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.

3.5.2 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:

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

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

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

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

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

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 $\mathrm{SNR}_{\mathrm{dB}} = 36$ and the channel bandwidth is 2 MHz. The theoretical channel capacity can be calculated as

$$\mathrm{SNR}_{\mathrm{dB}} = 10 \log_{10}\mathrm{SNR} \longrightarrow \mathrm{SNR} = 10^{\mathrm{SNR}_{\mathrm{dB}}/10} \longrightarrow \mathrm{SNR} = 10^{3.6} = 3981$$

$$C = B \log_2(1 + \mathrm{SNR}) = 2 \times 10^6 \times \log_2 3982 = 24 \text{ Mbps}$$

Example 3.40

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 $\boldsymbol{C = B \times \mathrm{SNR}_{\mathrm{dB}}}$. For example, we can calculate the theoretical capacity of the previous example as

$$C = 2 \text{ MHz} \times (36 / 3) = 24 \text{ Mbps}$$

3.5.3 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.

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

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.

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

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 30. In this section, we introduce terms that we need for future chapters.

3.6.1 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 second. 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.

3.6.2 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

$$\textbf{Throughput} = (12{,}000 \times 10{,}000) / 60 = 2 \text{ Mbps}$$

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

3.6.3 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.

Latency = propagation time + transmission time + queuing time + 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.

Propagation time = Distance / (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

$$\textbf{Propagation time} = (12{,}000 \times 10{,}000) / (2.4 \times 2^8) = 50 \textbf{ ms}$$

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 **transmission time** of a message depends on the size of the message and the bandwidth of the channel.

$$\textbf{Transmission time} = \textbf{(Message size) / Bandwidth}$$

Example 3.46

What are the propagation time and the transmission time for a 2.5-KB (kilobyte) 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

$$\textbf{Propagation time} = (12{,}000 \times 1000) / (2.4 \times 10^8) = 50 \textbf{ ms}$$

$$\textbf{Transmission time} = (2500 \times 8) / 10^9 = 0.020 \textbf{ ms}$$

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-MB (megabyte) 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

$$\textbf{Propagation time} = (12{,}000 \times 1000) / (2.4 \times 10^8) = 50 \textbf{ ms}$$

$$\textbf{Transmission time} = (5{,}000{,}000 \times 8) / 10^6 = 40 \textbf{ s}$$

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

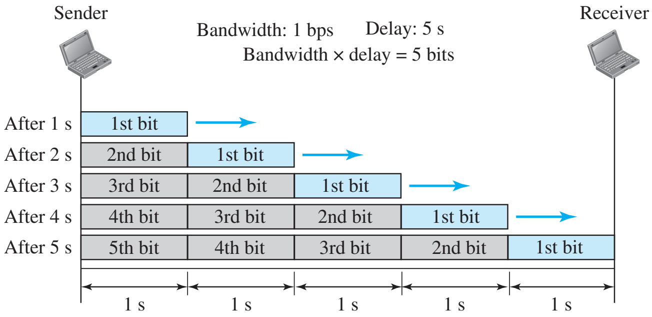
The third component in latency is the **queuing time**, the time needed for each intermediate or end device to hold the message before it can be processed. The queuing time is not a fixed factor; it changes with the load imposed on the network. When there is heavy traffic on the network, the queuing 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.

3.6.4 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 3.32 shows case 1.

Figure 3.32 *Filling the link with bits for case 1*

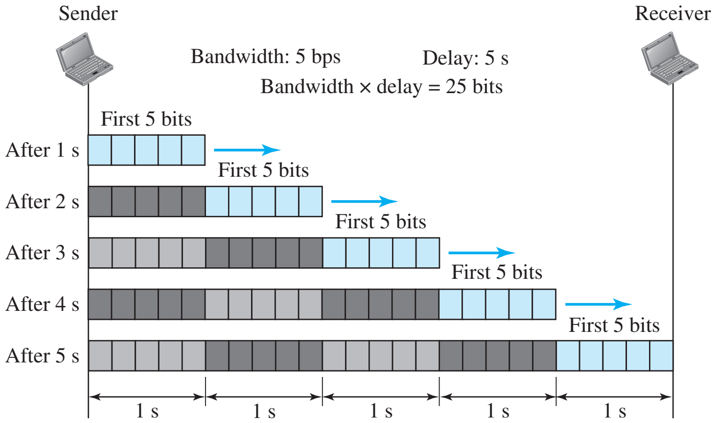


Let us assume that we have a link with a bandwidth of 1 bps (unrealistic, but good for demonstration purposes). We also assume that the delay of the link is 5 s (also unrealistic). We want to see what the bandwidth-delay product means in this case. Looking at the figure, we can say that this product 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 5 bps. Figure 3.33 shows that there can be maximum 5 × 5 = 25 bits on the line. The reason is that, at each second, there are 5 bits on the line; the duration of each bit is 0.20 s.

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

Figure 3.33 *Filling the link with bits in case 2*



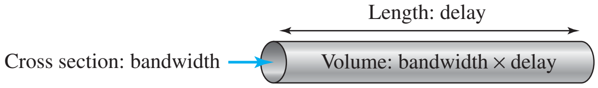
of bandwidth and delay; we need to fill up the full-duplex channel (two directions). The sender should send a burst of data of (2 × bandwidth × delay) bits. The sender then waits for receiver acknowledgment for part of the burst before sending another burst. The amount 2 × bandwidth × delay is the number 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 3.34.

Figure 3.34 *Concept of bandwidth-delay product*



3.6.5 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 28.

3.7 END-CHAPTER MATERIALS

3.7.1 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 discussed in [Pea92]. [Cou01] gives excellent coverage of signals. 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 [Sta04] and [Tan03].

3.7.2 Key Terms

analog
analog data
analog signal
attenuation
bandpass channel
bandwidth
baseband transmission
bit length
bit rate
bits per second (bps)
broadband transmission
composite signal
cycle
data
decibel (dB)
digital
digital data
digital signal
distortion
Fourier analysis
frequency
frequency-domain
fundamental frequency
harmonic
Hertz (Hz)
jitter
latency
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

3.7.3 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.

3.8 PRACTICE SET

3.8.1 Quizzes

A set of interactive quizzes for this chapter can be found on the book website. It is strongly recommended that the student take the quizzes to check his/her understanding of the materials before continuing with the practice set.

3.8.2 Questions

Q3-1. What is the relationship between period and frequency?

Q3-2. What does the amplitude of a signal measure? What does the frequency of a signal measure? What does the phase of a signal measure?

Q3-3. How can a composite signal be decomposed into its individual frequencies?

Q3-4. Name three types of transmission impairment.

Q3-5. Distinguish between baseband transmission and broadband transmission.

Q3-6. Distinguish between a low-pass channel and a band-pass channel.

Q3-7. What does the Nyquist theorem have to do with communications?

Q3-8. What does the Shannon capacity have to do with communications?

Q3-9. Why do optical signals used in fiber optic cables have a very short wave length?

Q3-10. Can we say whether a signal is periodic or nonperiodic by just looking at its frequency domain plot? How?

Q3-11. Is the frequency domain plot of a voice signal discrete or continuous?

Q3-12. Is the frequency domain plot of an alarm system discrete or continuous?

Q3-13. We send a voice signal from a microphone to a recorder. Is this baseband or broadband transmission?

Q3-14. We send a digital signal from one station on a LAN to another station. Is this baseband or broadband transmission?

Q3-15. We modulate several voice signals and send them through the air. Is this baseband or broadband transmission?

3.8.3 Problems

P3-1. Given the frequencies listed below, calculate the corresponding periods.

a. 24 Hz **b.** 8 MHz **c.** 140 KHz

P3-2. Given the following periods, calculate the corresponding frequencies.

a. 5 s **b.** 12 μs **c.** 220 ns

P3-3. 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

P3-4. 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.

P3-5. 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.

P3-6. 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?

P3-7. 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

P3-8. A device is sending out data at the rate of 1000 bps.

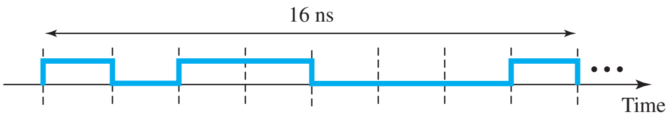
a. How long does it take to send out 10 bits?

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?

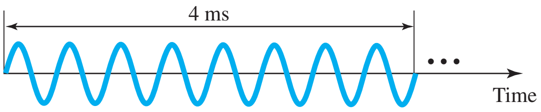
P3-9. What is the bit rate for the signal in Figure 3.35?

Figure 3.35 *Problem P3-9*



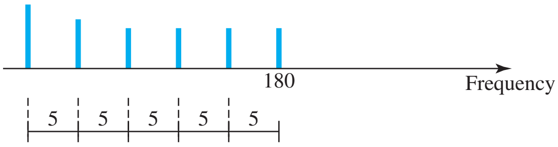
P3-10. What is the frequency of the signal in Figure 3.36?

Figure 3.36 *Problem P3-10*



P3-11. What is the bandwidth of the composite signal shown in Figure 3.37?

Figure 3.37 *Problem P3-11*



P3-12. A periodic composite signal contains frequencies from 10 to 30 KHz, each with an amplitude of 10 V. Draw the frequency spectrum.

P3-13. A nonperiodic 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.

P3-14. 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?

P3-15. 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?

P3-16. The attenuation of a signal is −10 dB. What is the final signal power if it was originally 5 W?

P3-17. 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?

P3-18. 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?

P3-19. The light of the sun takes approximately eight minutes to reach the earth. What is the distance between the sun and the earth?

P3-20. A signal has a wavelength of 1 μm in air. How far can the front of the wave travel during 1000 periods?

P3-21. 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?

P3-22. 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?

P3-23. A file contains 2 million bytes. How long does it take to download this file using a 56-Kbps channel? 1-Mbps channel?

P3-24. 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?

P3-25. 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 SNRdB?

P3-26. 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}?

P3-27. What is the theoretical capacity of a channel in each of the following cases?

a. Bandwidth: 20 KHz $\text{SNR}_{\text{dB}} = 40$

b. Bandwidth: 200 KHz $\text{SNR}_{\text{dB}} = 4$

c. Bandwidth: 1 MHz $\text{SNR}_{\text{dB}} = 20$

P3-28. 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?

P3-29. 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 the SNR?

P3-30. 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?

P3-31. 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?

P3-32. 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?

P3-33. 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?

3.9 SIMULATION EXPERIMENTS

3.9.1 Applets

We have created some Java applets to show some of the main concepts discussed in this chapter. It is strongly recommended that the students activate these applets on the book website and carefully examine the protocols in action.