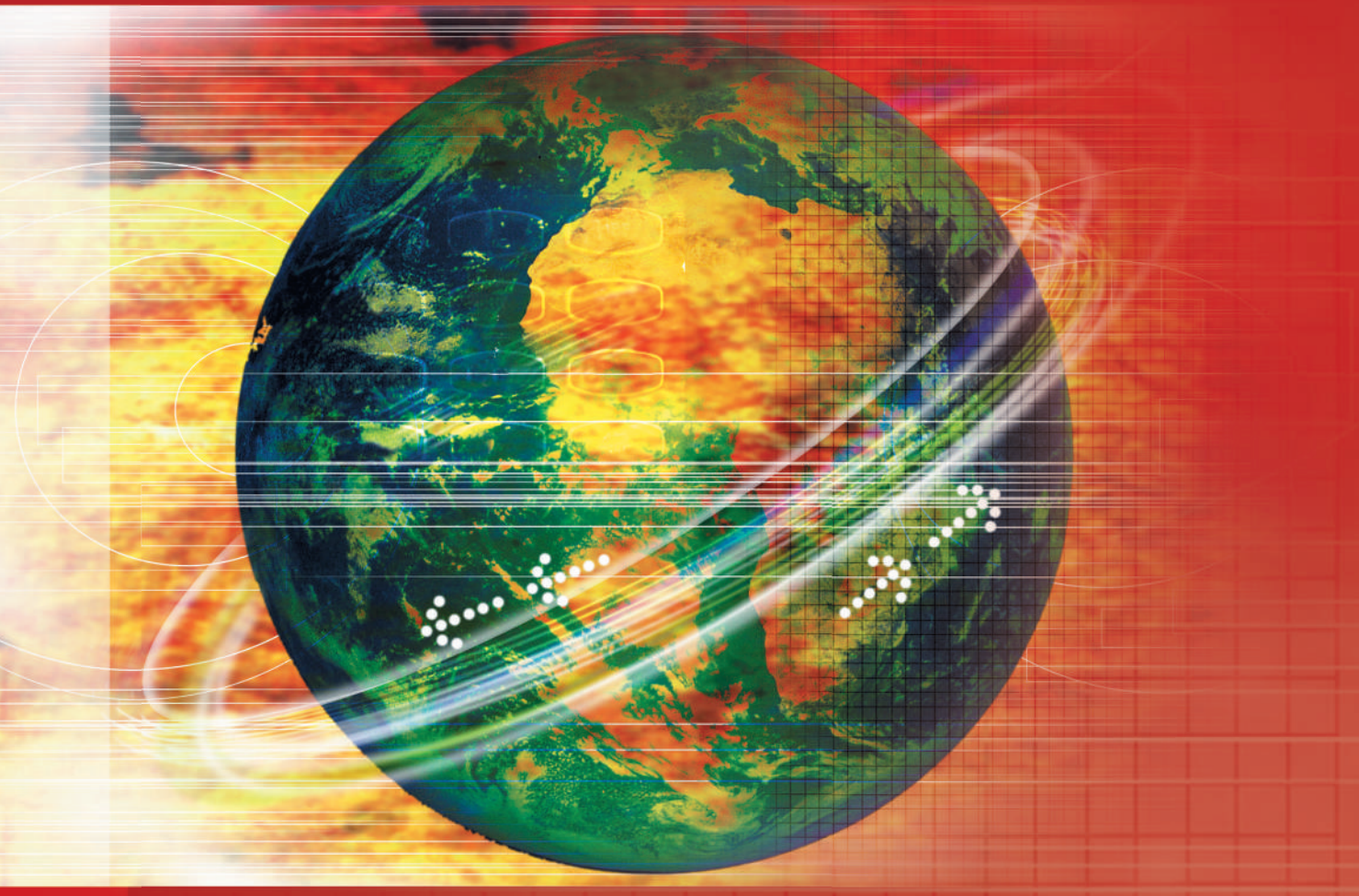
FIFTH EDITION

Data Communications AND Networking



BEHROUZ A. FOROUZAN

Data Communications and Networking

McGraw-Hill Forouzan Networking Series

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Data Communications and Networking
TCP/IP Protocol Suite
Computer Networks: A Top-Down Approach
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Data Communications and Networking

FIFTH EDITION

Behrouz A. Forouzan

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To my beloved grandson, William.

BRIEF CONTENTS

CONTENTS

PREFACE

Technologies related to data communication and networking may be the fastest growing in our culture today. The appearance of some new social networking applications every year is a testimony to this claim. People use the Internet more and more every day. They use the Internet for research, shopping, airline reservations, checking the latest news and weather, and so on.

In this Internet-oriented society, specialists need be trained to run and manage the Internet, part of the Internet, or an organization's network that is connected to the Internet. This book is designed to help students understand the basics of data communications and networking in general and the protocols used in the Internet in particular.

Features

Although the main goal of the book is to teach the principles of networking, it is designed to teach these principles using the following goals:

Protocol Layering

The book is designed to teach the principles of networking by using the protocol layering of the Internet and the TCP/IP protocol suite. Some of the networking principles may have been duplicated in some of these layers, but with their own special details. Teaching these principles using protocol layering is beneficial because these principles are repeated and better understood in relation to each layer. For example, although *addressing* is an issue that is applied to four layers of the TCP/IP suite, each layer uses a different addressing format for different purposes. In addition, addressing has a different domain in each layer. Another example is *framing and packetizing*, which is repeated in several layers, but each layer treats the principle differently.

Bottom-Up Approach

This book uses a bottom-up approach. Each layer in the TCP/IP protocol suite is built on the services provided by the layer below. We learn how bits are moving at the physical layer before learning how some programs exchange messages at the application layer.

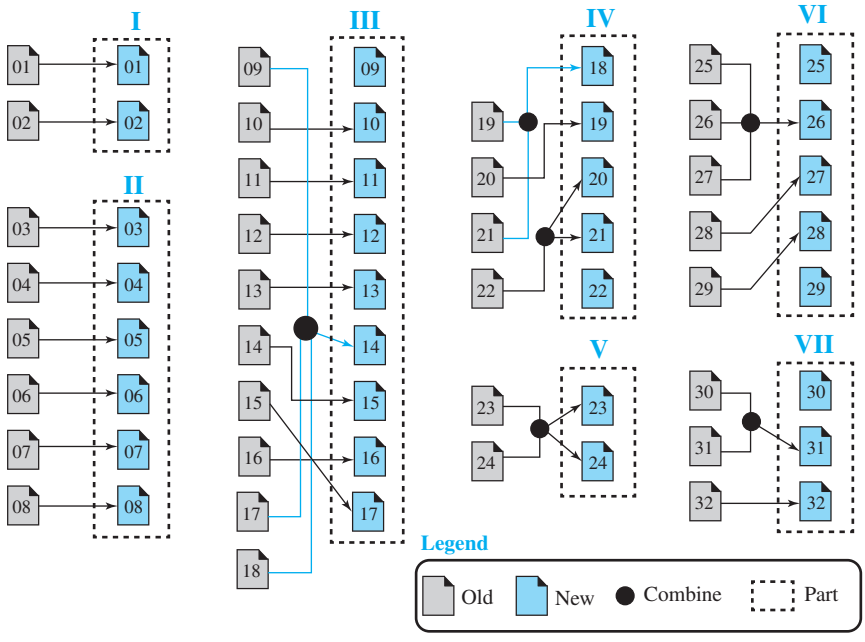
Changes in the Fifth Edition

I have made several categories of changes in this edition.

Changes in the Organization

Although the book is still made of seven parts, the contents and order of chapters have been changed. Some chapters have been combined, some have been moved, some are

new. Sometimes part of a chapter is eliminated because the topic is deprecated. The following shows the relationship between chapters in the fourth and fifth editions.



- Some chapters have been combined into one chapter. Chapters 9, 17, and 18 are combined into one chapter because some topics in each chapter have been deprecated. Chapters 19 and 21 are combined into Chapter 18. Chapters 25, 26, and 27 are also combined into one chapter because the topics are related to each other. Chapters 30 and 31 are also combined because they cover the same issue.
- Some chapters have been split into two chapters because of content augmentation. For example, Chapter 22 is split into Chapters 20 and 21.
- Some chapters have been first combined, but then split for better organization. For example, Chapters 23 and 24 are first combined and then split into two chapters again.
- Some chapters have been moved to better fit in the organization of the book. Chapter 15 now becomes Chapter 17. Chapters 28 and 29 now become Chapters 27 and 28.
- Some chapters have been moved to fit better in the sequence. For example, Chapter 15 has become Chapter 17 to cover more topics.
- Some chapters are new. Chapter 9 is an introduction to the data-link layer. Chapter 25 is an introduction to the application layer and includes socket-interface programming in C and Java. Chapter 30 is almost new. It covers QoS, which was part of other chapters in the previous edition.

New and Augmented Materials

Although the contents of each chapter have been updated, some new materials have also been added to this edition:

- *Peer-to-Peer paradigm* has been added as a new chapter (Chapter 29).
- *Quality of service* (QoS) has been augmented and added as a new chapter (Chapter 30).
- Chapter 10 is augmented to include the *forward error correction*.
- WiMAX, as the wireless access network, has been added to Chapter 16.
- The coverage of the transport-layer protocol has been augmented (Chapter 23).
- Socket-interface programming in Java has been added to Chapter 25.
- Chapter 28, on multimedia, has been totally revised and augmented.
- Contents of unicast and multicast routing (Chapters 20 and 21) have gone through a major change and have been augmented.
- The next generation IP is augmented and now belongs to Chapter 22.

Changes in the End-Chapter Materials

The end-chapter materials have gone through a major change:

- The practice set is augmented; it has many new problems in some appropriate chapters.
- Lab assignments have been added to some chapters to allow students to see some data in motion.
- Some applets have been posted on the book website to allow students to see some problems and protocols in action.
- Some programming assignments allow the students to write some programs to solve problems.

Extra Materials

Some extra materials, which could not be fit in the contents and volume of the book, have been posted on the book website for further study.

New Organization

This edition is divided into seven parts, which reflects the structure of the Internet model.

Part One: Overview

The first part gives a general overview of data communications and networking. Chapter 1 covers introductory concepts needed for the rest of the book. Chapter 2 introduces the Internet model.

Part Two: Physical Layer

The second part is a discussion of the physical layer of the Internet model. It is made of six chapters. Chapters 3 to 6 discuss telecommunication aspects of the physical layer.

Chapter 7 introduces the transmission media, which, although not part of the physical layer, is controlled by it. Chapter 8 is devoted to switching, which can be used in several layers.

Part Three: Data-Link Layer

The third part is devoted to the discussion of the data-link layer of the Internet model. It is made of nine chapters. Chapter 9 introduces the data-link layer. Chapter 10 covers error detection and correction, which can also be used in some other layers. Chapters 11 and 12 discuss issues related to two sublayers in the data-link layer. Chapters 13 and 14 discuss wired networks. Chapters 15 and 16 discuss wireless networks. Chapter 17 shows how networks can be combined to create larger or virtual networks.

Part Four: Network Layer

The fourth part is devoted to the discussion of the network layer of the Internet model. Chapter 18 introduces this layer and discusses the network-layer addressing. Chapter 19 discusses the protocols in the current version. Chapters 20 and 21 are devoted to routing (unicast and multicast). Chapter 22 introduces the next generation protocol.

Part Five: Transport Layer

The fifth part is devoted to the discussion of the transport layer of the Internet model. Chapter 23 gives an overview of the transport layer and discusses the services and duties of this layer. Chapter 24 discusses the transport-layer protocols in the Internet: UDP, TCP, and SCTP.

Part Six: Application Layer

Chapter 25 introduces the application layer and discusses some network programming in both C and Java. Chapter 26 discusses most of the standard client-server programming in the Internet. Chapter 27 discusses network management. Chapter 28 is devoted to the multimedia, an issue which is very hot today. Finally, Chapter 29 is an introduction to the peer-to-peer paradigm, a trend which is on the rise in the today's Internet.

Part Seven: Topics Related to All Layers

The last part of the book discusses the issues that belong to some or all layers. Chapter 30 discusses the quality of service. Chapters 31 and 32 discuss security.

Appendices

The appendices (available online at http://www.mhhe.com/forouzan) are intended to provide a quick reference or review of materials needed to understand the concepts discussed in the book. There are eight appendices that can be used by the students for reference and study:

- Appendix A: Unicode
- Appendix B: Positional Numbering System
- Appendix C: HTML,CSS, XML, and XSL
- Appendix D: A Touch of Probability
- Appendix E: Mathematical Review

- Appendix F: 8B/6T Code
- Appendix G: Miscellaneous Information
- Appendix H: Telephone History

References

The book contains a list of references for further reading.

Glossary and Acronyms

The book contains an extensive glossary and a list of acronyms for finding the corresponding term quickly.

Pedagogy

Several pedagogical features of this text are designed to make it particularly easy for students to understand data communication and networking.

Visual Approach

The book presents highly technical subject matter without complex formulas by using a balance of text and figures. More than 830 figures accompanying the text provide a visual and intuitive opportunity for understanding the material. Figures are particularly important in explaining networking concepts. For many students, these concepts are more easily grasped visually than verbally.

Highlighted Points

I have repeated important concepts in boxes for quick reference and immediate attention.

Examples and Applications

Whenever appropriate, I have included examples that illustrate the concepts introduced in the text. Also, I have added some real-life applications throughout each chapter to motivate students.

End-of-Chapter Materials

Each chapter ends with a set of materials that includes the following:

Key Terms

The new terms used in each chapter are listed at the end of the chapter and their definitions are included in the glossary.

Recommended Reading

This section gives a brief list of references relative to the chapter. The references can be used to quickly find the corresponding literature in the reference section at the end of the book.

Summary

Each chapter ends with a summary of the material covered by that chapter. The summary glues the important materials together to be seen in one shot.

Practice Set

Each chapter includes a practice set designed to reinforce salient concepts and encourage students to apply them. It consists of three parts: quizzes, questions, and problems.

Quizzes

Quizzes, which are posted on the book website, provide quick concept checking. Students can take these quizzes to check their understanding of the materials. The feedback to the students' responses is given immediately.

Questions

This section contains simple questions about the concepts discussed in the book. Answers to the odd-numbered questions are posted on the book website to be checked by the student. There are more than 630 questions at the ends of chapters.

Problems

This section contains more difficult problems that need a deeper understanding of the materials discussed in the chapter. I strongly recommend that the student try to solve all of these problems. Answers to the odd-numbered problems are also posted on the book website to be checked by the student. There are more than 600 problems at the ends of chapters.

Simulation Experiments

Network concepts and the flow and contents of the packets can be better understood if they can be analyzed in action. Some chapters include a section to help students experiment with these. This section is divided into two parts: applets and lab assignments.

Applets

Java applets are interactive experiments that are created by the authors and posted on the website. Some of these applets are used to better understand the solutions to some problems; others are used to better understand the network concepts in action.

Lab Assignments

Some chapters include lab assignments that use Wireshark simulation software. The instructions for downloading and using Wireshark are given in Chapter 1. In some other chapters, there are a few lab assignments that can be used to practice sending and receiving packets and analyzing their contents.

Programming Assignments

Some chapters also include programming assignments. Writing a program about a process or procedure clarifies many subtleties and helps the student better understand the concept behind the process. Although the student can write and test programs in any computer language she or he is comfortable with, the solutions are given in Java language at the book website for the use of professors.

Audience

This book is written for both academic and professional audiences. The book can be used as a self-study guide for interested professionals. As a textbook, it can be used for a one-semester or one-quarter course. It is designed for the last year of undergraduate study or the first year of graduate study. Although some problems at the end of the chapters require some knowledge of probability, the study of the text needs only general mathematical knowledge taught in the first year of college.

Instruction Resources

The book contains complete instruction resources that can be downloaded from the book site **http://www.mhhe.com/forouzan**. They include:

Presentations

The site includes a set of colorful and animated PowerPoint presentations for teaching the course.

Solutions to Practice Set

Solutions to all questions and problems are provided on the book website for the use of professors who teach the course.

Solution to Programming Assignments

Solutions to programming assignments are also provided on the book website. The programs are mostly in Java language.

Student Resources

The book contains complete student resources that can be downloaded from the book website **http://www.mhhe.com/forouzan**. They include:

Quizzes

There are quizzes at the end of each chapter that can be taken by the students. Students are encouraged to take these quizzes to test their general understanding of the materials presented in the corresponding chapter.

Solution to Odd-Numbered Practice Set

Solutions to all odd-numbered questions and problems are provided on the book website for the use of students.

Lab Assignments

The descriptions of lab assignments are also included in the student resources.

Applets

There are some applets for each chapter. Applets can either show the solution to some examples and problems or show some protocols in action. It is strongly recommended that students activate these applets.

Extra Materials

Students can also access the extra materials at the book website for further study.

How to Use the Book

The chapters in the book are organized to provide a great deal of flexibility. I suggest the following:

- Materials provided in Part I are essential for understanding the rest of the book.
- Part II (physical layer) is essential to understand the rest of the book, but the professor can skip this part if the students already have the background in engineering and the physical layer.
- Parts III to VI are based on the Internet model. They are required for understanding the use of the networking principle in the Internet.
- Part VII (QoS and Security) is related to all layers of the Internet mode. It can be partially or totally skipped if the students will be taking a course that covers these materials.

Website

The McGraw-Hill website contains much additional material, available at **www.mhhe.com/forouzan.** As students read through *Data Communications and Networking,* they can go online to take self-grading quizzes. They can also access lecture materials such as PowerPoint slides, and get additional review from animated figures from the book. Selected solutions are also available over the Web. The solutions to odd-numbered problems are provided to students, and instructors can use a password to access the complete set of solutions.

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Behrouz A. Forouzan
Los Angeles, CA.
January 2012

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PART I

Overview

In the first part of the book, we discuss some general ideas related to both data communications and networking. This part lays the plan for the rest of the book. The part is made of two chapters that prepare the reader for the long journey ahead.

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.

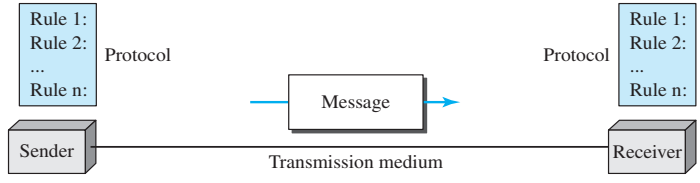
Data communications are the exchange of data between two devices via some form of transmission medium such as a wire cable. For data communications to occur, the communicating devices must be part of a communication system made up of a combination of hardware (physical equipment) and software (programs). 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.

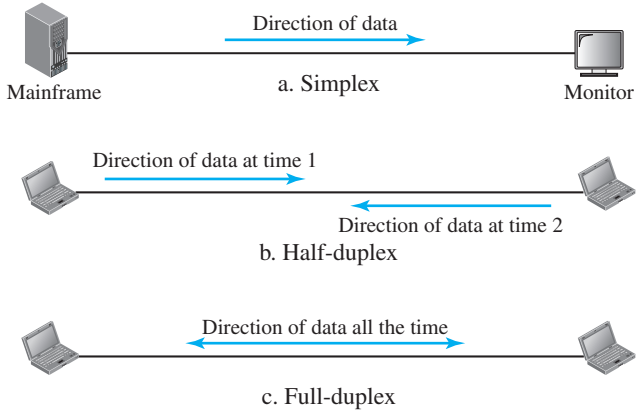
Video

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)*



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.

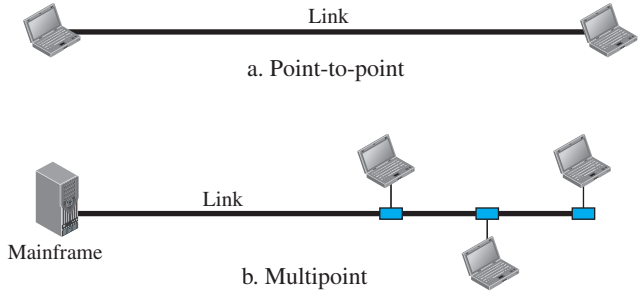
Point-to-Point

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

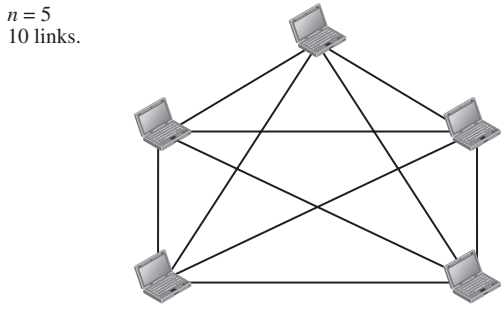
The term ***physical topology*** refers to the way in which a network is laid out physically. Two or more devices connect to a link; two or more links form a topology. The topology of a network is the geometric representation of the relationship of all the links and linking devices (usually called ***nodes***) 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)*



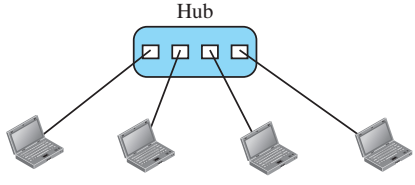
The main disadvantages of a mesh are related to the amount of cabling and the number of I/O ports required. First, because every device must be connected to every other device, installation and reconnection are difficult. Second, the sheer bulk of the wiring can be greater than the available space (in 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.

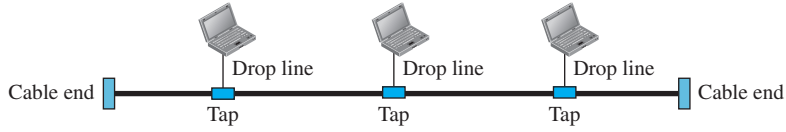
Although a star requires far less cable than a mesh, each node must be linked to a central hub. For this reason, often more cabling is required in a star than in some other topologies (such as ring or bus).

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.

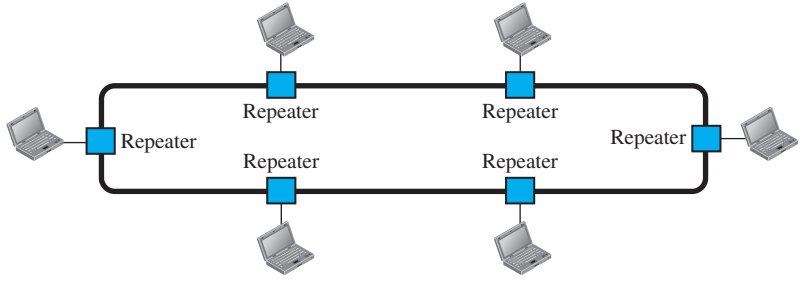
In addition, a fault or break in the bus cable stops all transmission, even between devices on the same side of the problem. The damaged area reflects signals back in the direction of origin, creating noise in both directions.

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

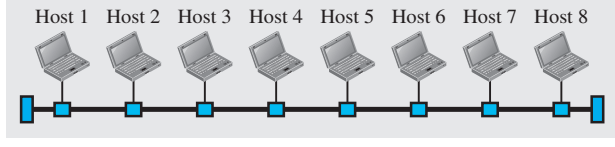
After defining networks in the previous section and discussing their physical structures, we need to discuss different types of networks we encounter in the world today. The criteria of distinguishing one type of network from another is difficult and sometimes confusing. We use a few criteria such as 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

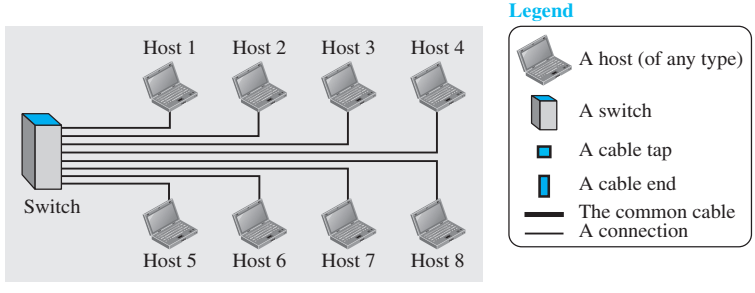
A **local area network** (**LAN**) is usually privately owned and connects some hosts in a single office, building, or campus. Depending on the needs of an organization, a LAN can be as simple as two PCs and a printer in someone's home office, or it can extend throughout a company and include audio and 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.

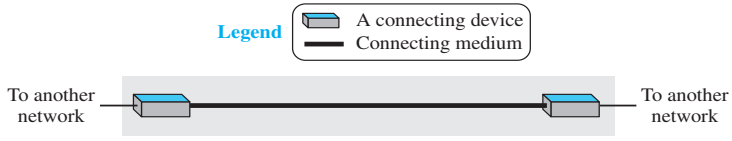
1.3.2 Wide Area Network

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.

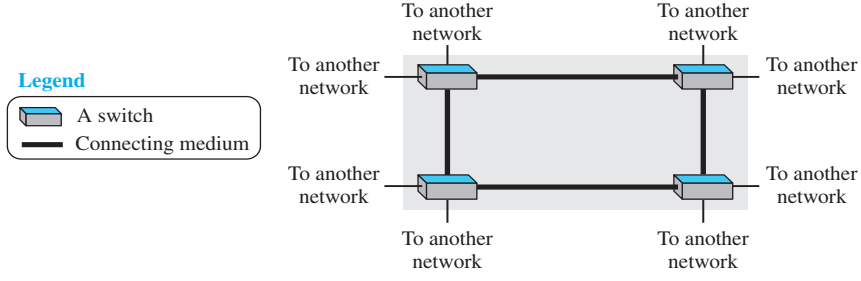
Figure 1.9 *A point-to-point WAN*



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*

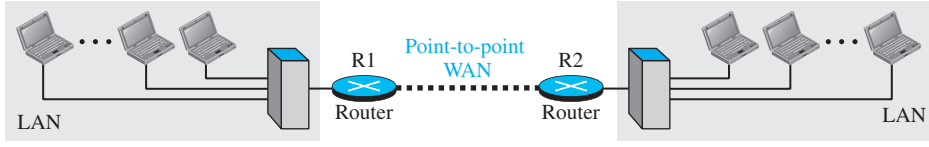


WANs are discussed in more detail in Part II of the book.

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*

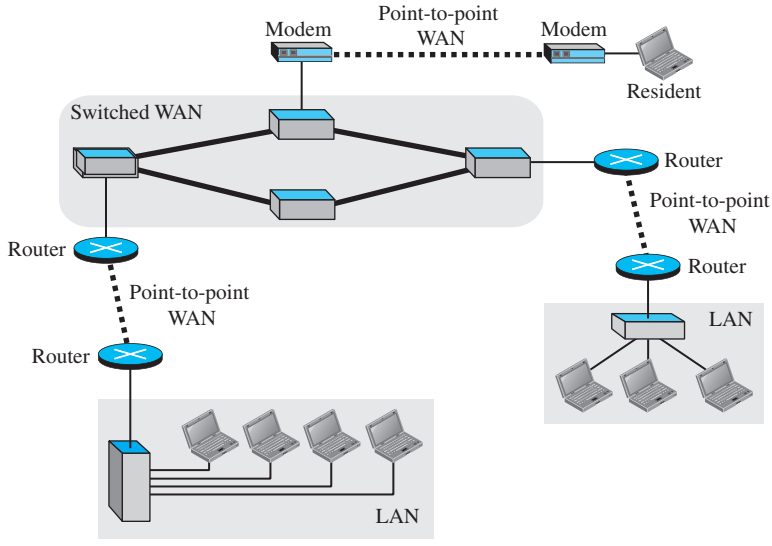
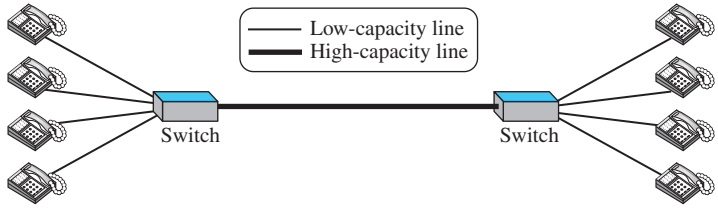


Figure 1.13 *A circuit-switched network*



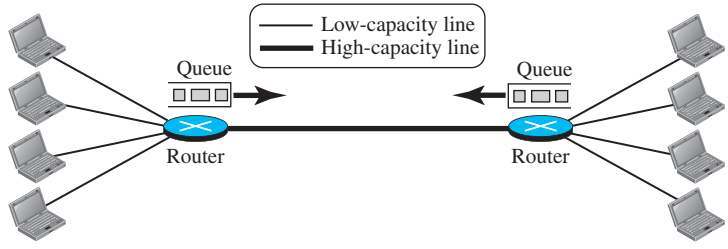
line connecting two switches is a high-capacity communication line that can handle four voice communications at the same time; the capacity can be shared between all pairs of telephone sets. The switches used in this example have forwarding tasks but no storing capability.

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.

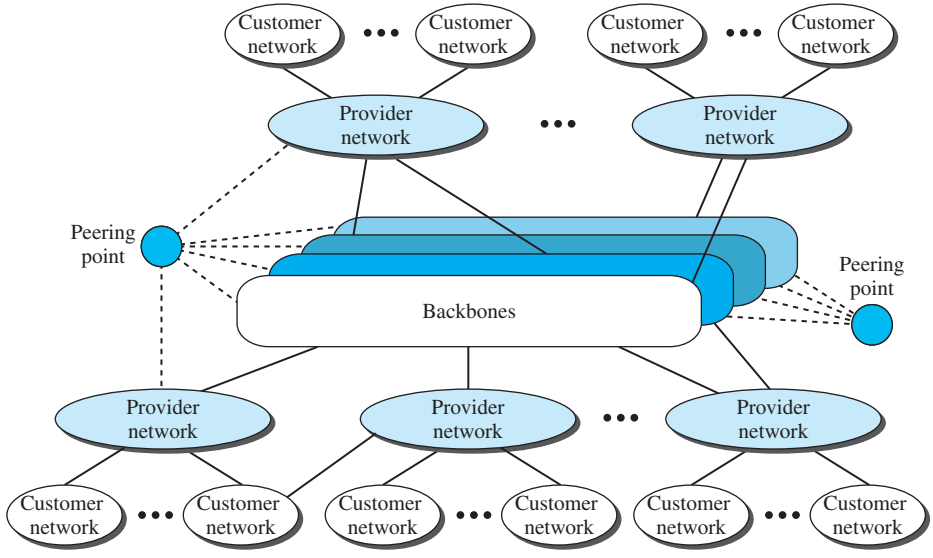
In this book, we mostly discuss packet-switched networks. In Chapter 18, we discuss packet-switched networks in more detail and discuss the performance of these networks.

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

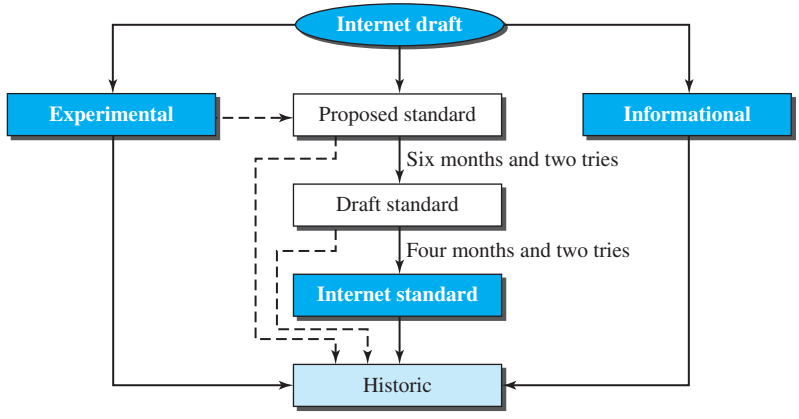
An **Internet standard** is a thoroughly tested specification that is useful to and adhered to by those who work with the Internet. It is a formalized regulation that must be followed. There is a strict procedure by which a specification attains Internet standard status. A specification begins as an 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.

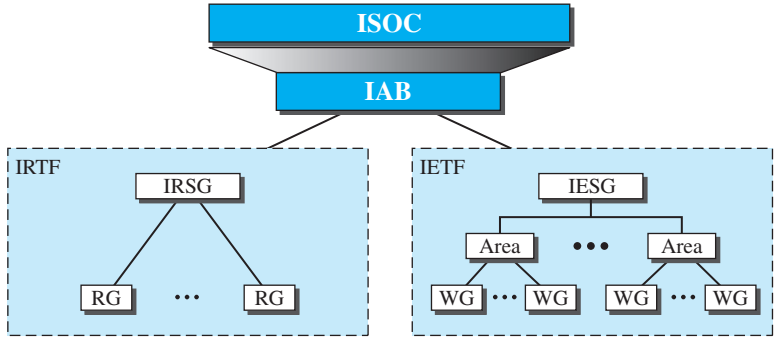
- ***Limited Use.*** An RFC labeled limited use should be used only in limited situations. Most of the experimental RFCs fall under this category.
- ***Not Recommended.*** An RFC labeled not recommended is inappropriate for general use. Normally a historic (deprecated) RFC may fall under this category.

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 *PingPlotter,* 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.

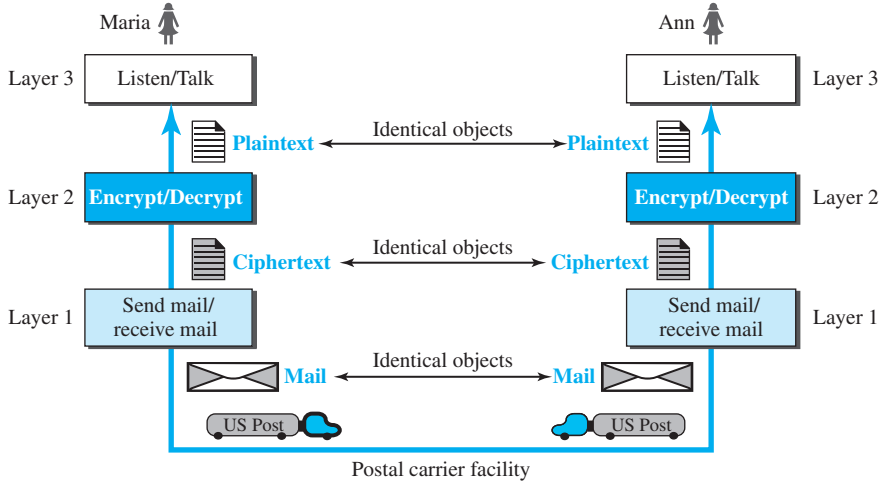
We can see that the protocol used by Maria and Ann is different from the communication between a professor and the students in a lecture hall. The communication in the second case is mostly monolog; the professor talks most of the time unless a student has a question, a situation in which the protocol 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

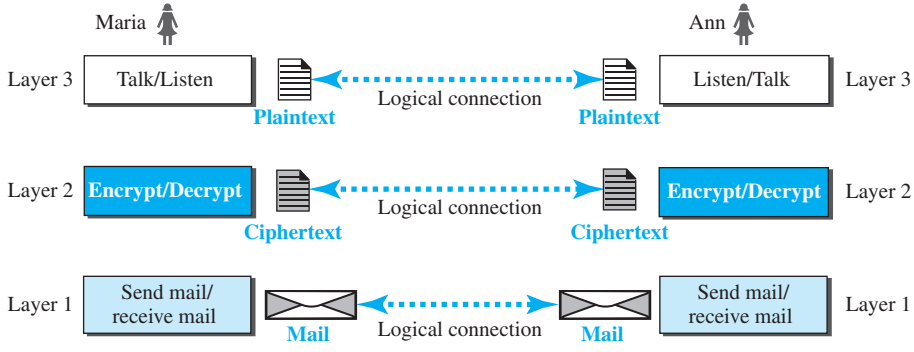
The second principle that we need to follow in protocol layering is that the two objects under each layer at both sites should be identical. For example, the object under layer 3 at both sites should be a plaintext letter. The object under layer 2 at

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*

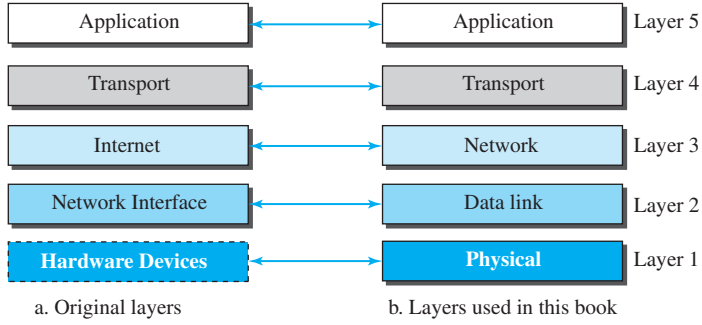
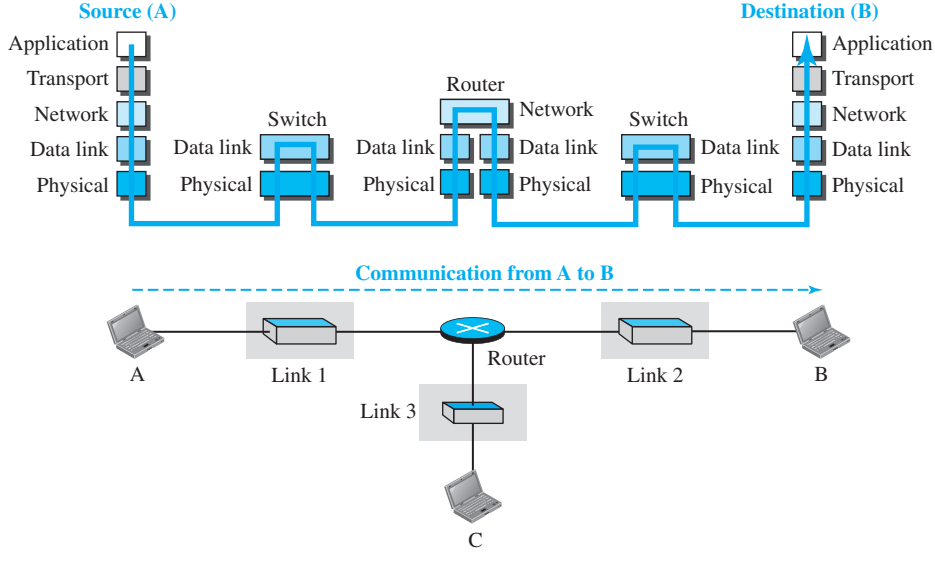


Figure 2.5 *Communication through an internet*



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.

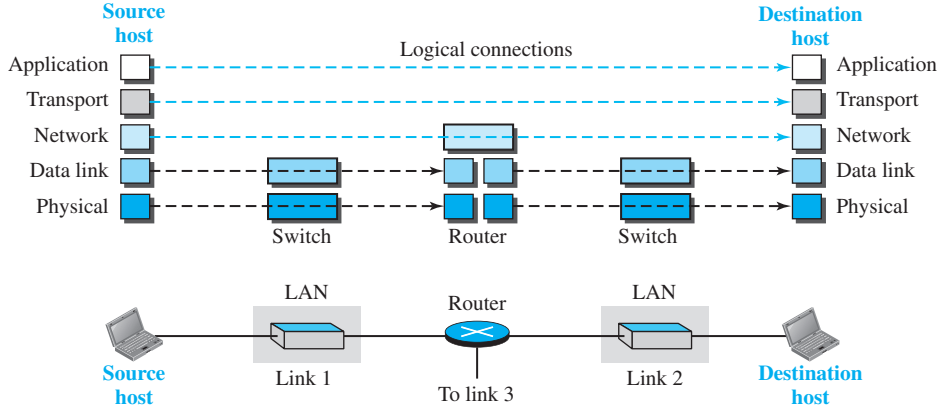
The router is involved in only three layers; there is no transport or application layer in a router as long as the router is used only for routing. Although a router is always involved in one network layer, it is involved in *n* combinations of link and physical layers in which *n* is the number of links 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*



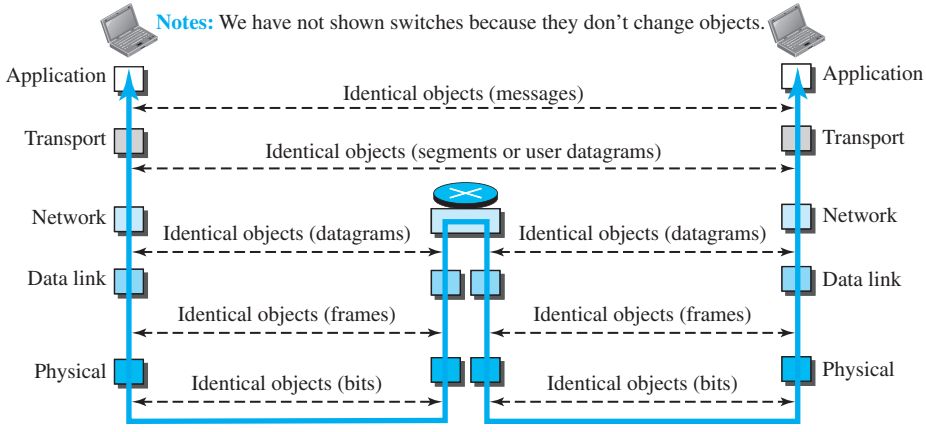
Using logical connections makes it easier for us to think about the duty of each layer. As the figure shows, the duty of the application, transport, and network layers is end-to-end. However, the duty of the data-link and physical layers is hop-to-hop, in which a hop is a host or router. In other words,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



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

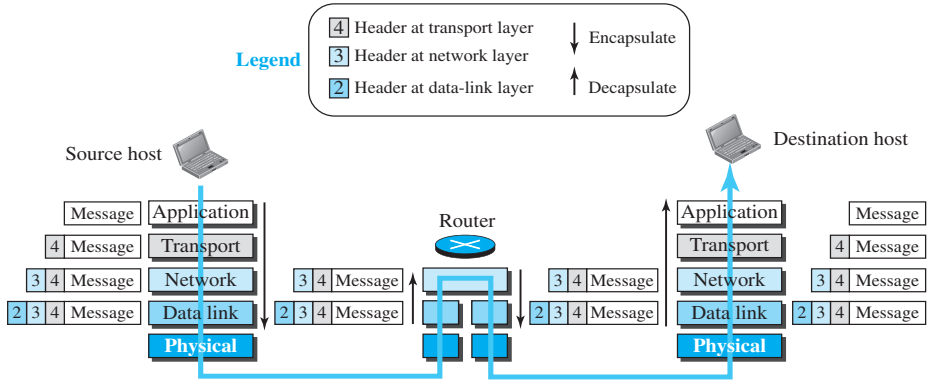
a user can also create a pair of processes to be run at the two hosts. In Chapter 25, we explore this situation.

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

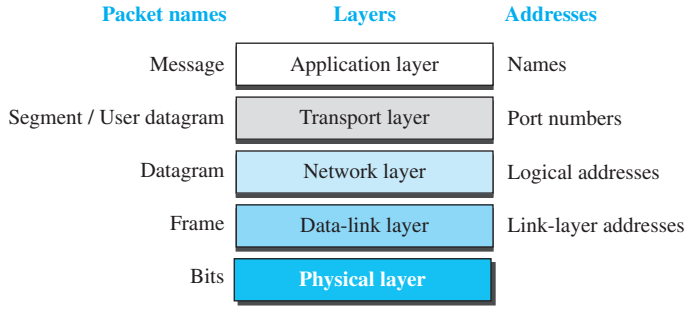
At the destination host, each layer only decapsulates the packet received, removes the payload, and delivers the payload to the next-higher layer protocol until the message reaches the application layer. It is necessary to say that decapsulation in the host involves error checking.

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*

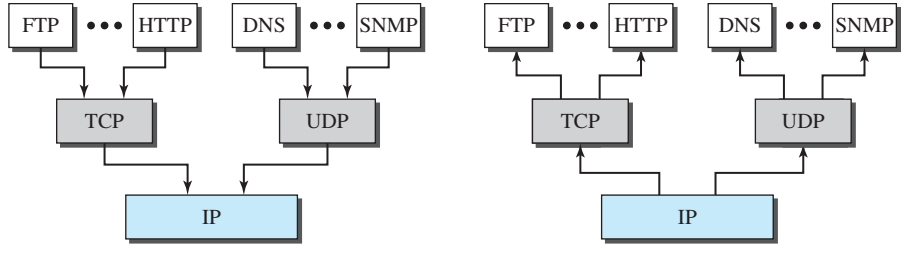


address, such as *somebody@coldmail.com.* At the transport layer, addresses are called port numbers, and these define the application-layer programs at the source and destination. Port numbers are local addresses that distinguish between several programs running at the same time. At the network-layer, 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*



a. Multiplexing at source

b. Demultiplexing at destination

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

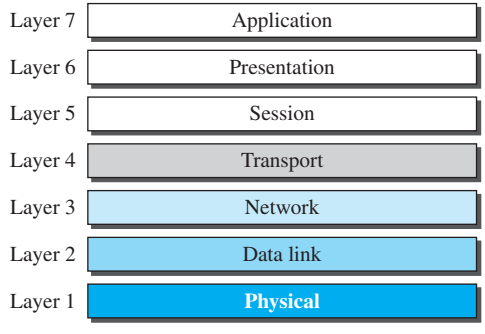
Although, when speaking of the Internet, everyone talks about the TCP/IP protocol suite, this suite is not the only suite of protocols defined. Established in 1947, the **International Organization for Standardization (ISO)** is a multinational body dedicated to worldwide agreement on international standards. 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).

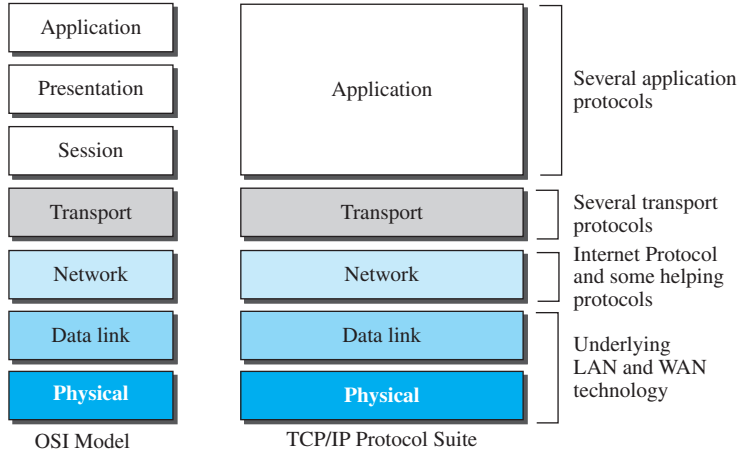
Figure 2.11 *The OSI model*



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.

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.**

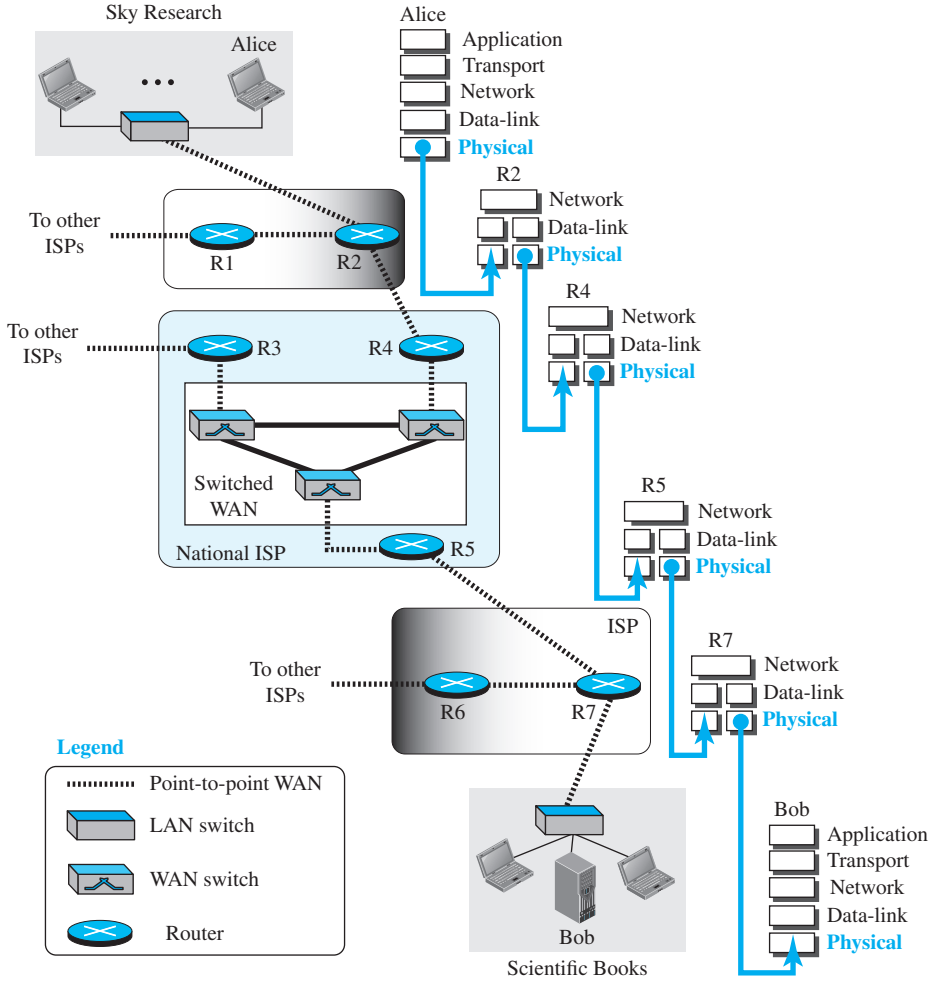
This chapter is divided into six sections:

- 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.

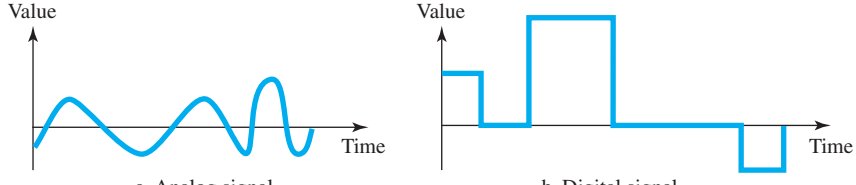
Digital data take on discrete values. For example, data are stored in computer memory in the form of 0s and 1s. They can be converted to a digital signal or modulated into an analog signal for transmission across a medium.

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.

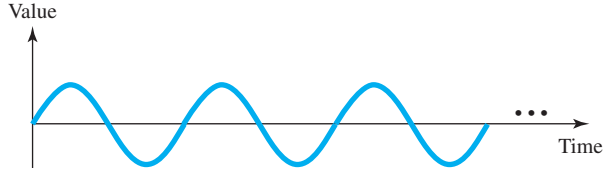
3.2 PERIODIC ANALOG SIGNALS

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*



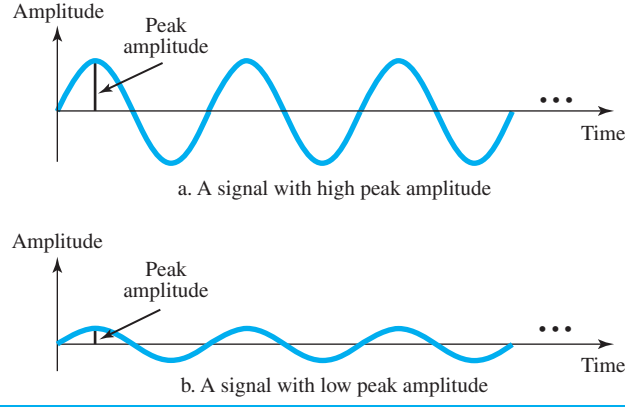
We discuss a mathematical approach to sine waves in Appendix E.

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*



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 $2^{1/2} \times$ 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.

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

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*

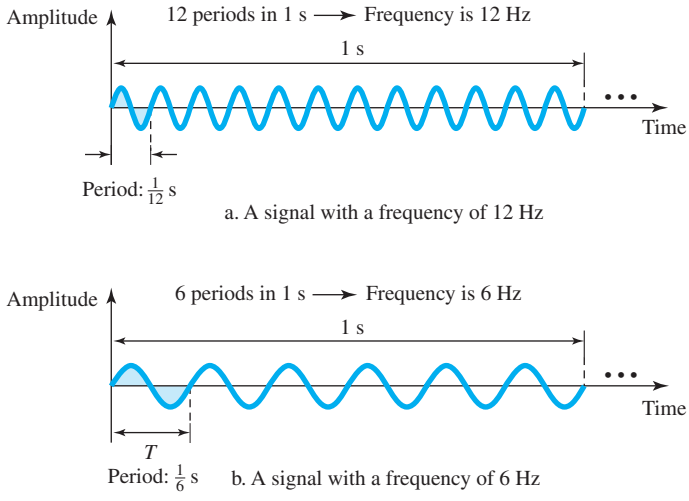


Table 3.1 *Units of period and frequency*

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:

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

Example 3.5

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

Solution

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

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

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

More About Frequency

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

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

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

Two Extremes

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

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

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

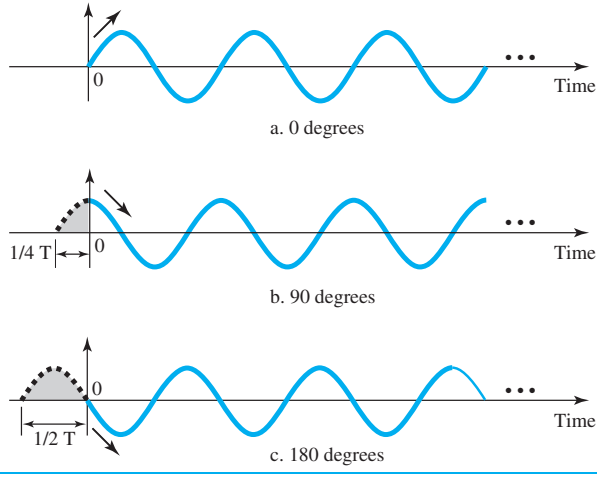
3.2.2 Phase

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; 1º 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

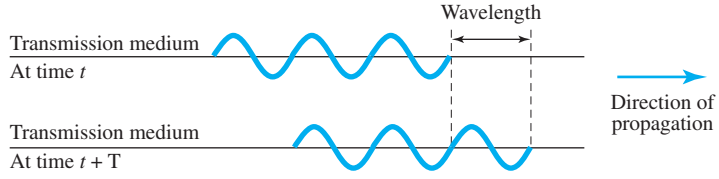
We know that 1 complete cycle is 360°. Therefore, $\frac{1}{6}$ cycle is

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

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

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

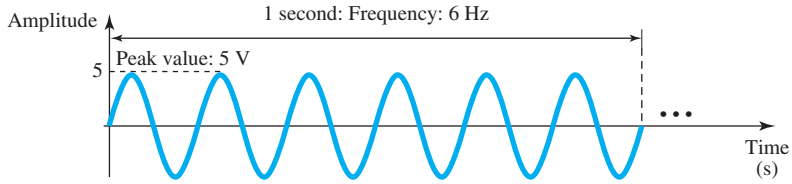
In a coaxial or fiber-optic cable, however, the wavelength is shorter (0.5 μm) because the propagation speed in the cable is decreased.

3.2.4 Time and Frequency Domains

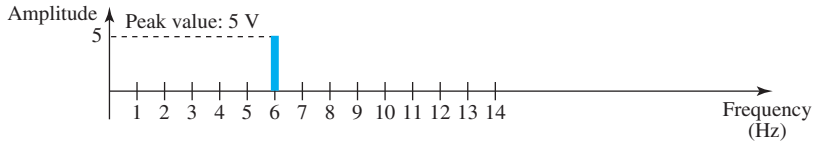
A sine wave is comprehensively defined by its amplitude, frequency, and phase. We have been showing a sine wave by using what is called a **time-domain** plot. The time-domain plot shows changes in signal amplitude with respect to time (it is an amplitude-versus-time plot). Phase is not explicitly shown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

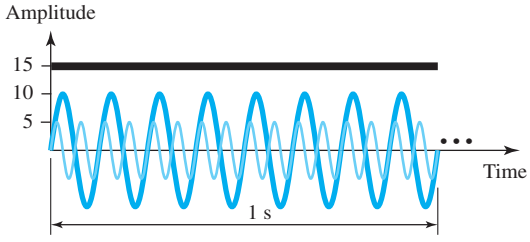
It is obvious that the frequency domain is easy to plot and conveys the information that one can find in a time domain plot. The advantage of the frequency domain is that we can immediately see the values of the frequency and peak amplitude. A complete sine wave is represented by one spike. The position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

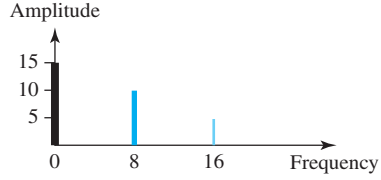
Example 3.7

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 $3f$ is one-third of that of

Figure 3.10 *A composite periodic signal*

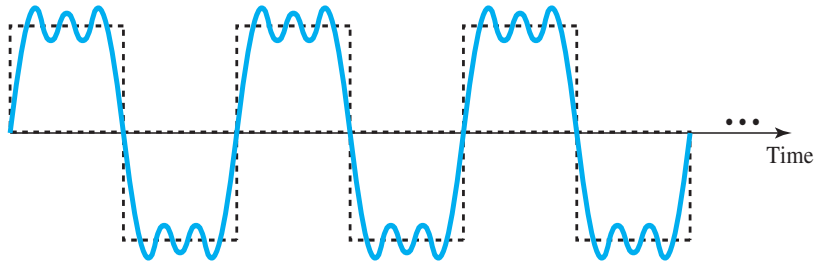
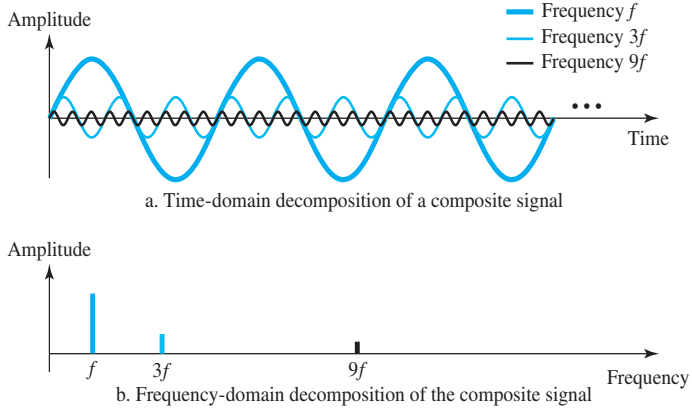


Figure 3.11 *Decomposition of a composite periodic signal in the time and frequency domains*



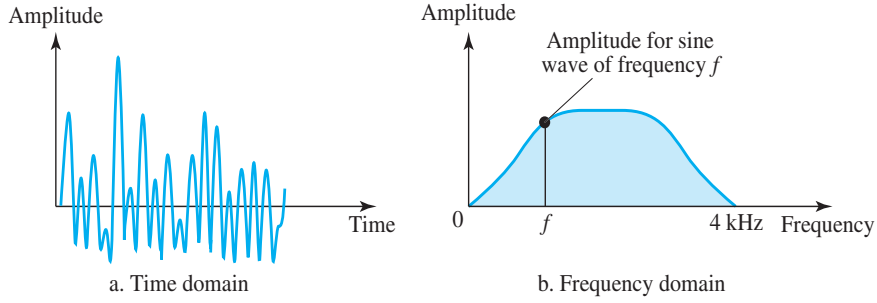
the first, and the amplitude of the sine wave with frequency $9f$ is one-ninth of the first. The frequency of the sine wave with frequency f is the same as the frequency of the composite signal; it is called the **fundamental frequency,** or first **harmonic.** The sine wave with frequency $3f$ has a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

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

Example 3.9

Figure 3.12 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*



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.

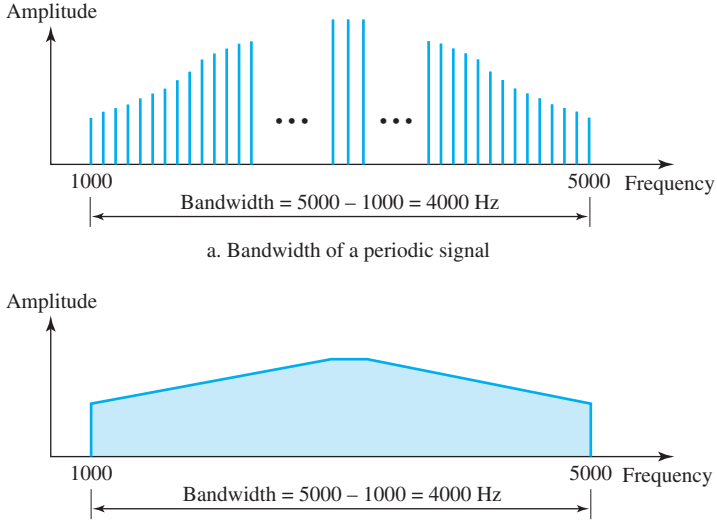
Solution

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

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

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

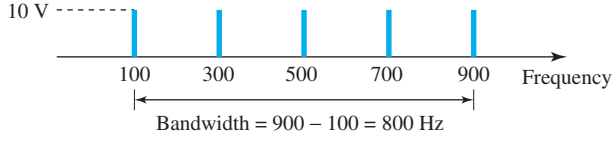
Figure 3.13 *The bandwidth of periodic and nonperiodic composite signals*



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

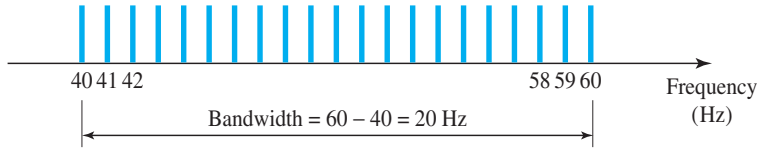
$$B = f_h - f_l \longrightarrow 20 = 60 - f_l \longrightarrow f_l = 60 - 20 = 40 \text{ Hz}$$

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.

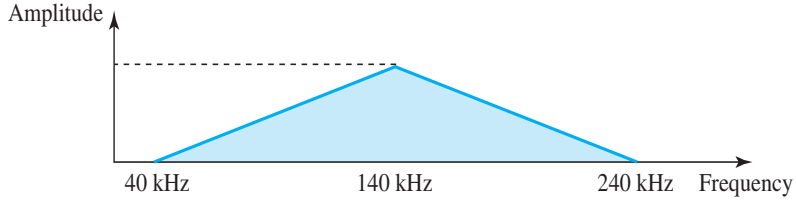
Figure 3.15 *The bandwidth for Example 3.11*



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

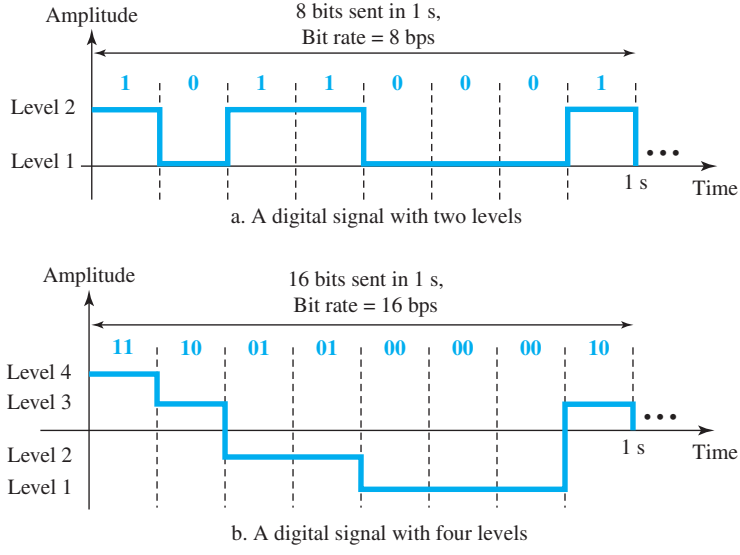
Another example of a nonperiodic composite signal is the signal received by an old-fashioned analog black-and-white TV. A TV screen is made up of pixels (picture elements) with each pixel being either white or black. The screen is scanned 30 times per second. (Scanning is actually 60 times per second, 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 $367{,}500 \times 30 = 11{,}025{,}000$ pixels per second. The worst-case scenario is alternating black and white pixels. In this case, we need to represent one color by the minimum amplitude and the other color by the maximum amplitude. We can send 2 pixels per cycle. Therefore, we need $11{,}025{,}000 / 2 = 5{,}512{,}500$ cycles per second, or Hz. The bandwidth needed is 5.5124 MHz.

This worst-case scenario has such a low probability of occurrence that the assumption is that we need only 70 percent of this bandwidth, which is 3.85 MHz. Since audio and synchronization signals are also needed, a 4-MHz bandwidth has been set aside for each black and white TV channel. An analog color TV channel has a 6-MHz bandwidth.

3.3 DIGITAL SIGNALS

In addition to being represented by an analog signal, information can also be represented by a digital signal. For example, a 1 can be encoded as a positive voltage and a 0 as zero voltage. A digital signal can have more than two levels. In this case, we can send more than 1 bit for each level. Figure 3.17 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*



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.

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

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.

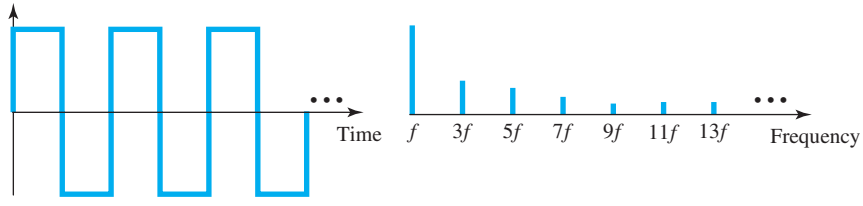
$$\text{Bit length} = \text{propagation speed} \times \text{bit duration}$$

3.3.3 Digital Signal as a Composite Analog Signal

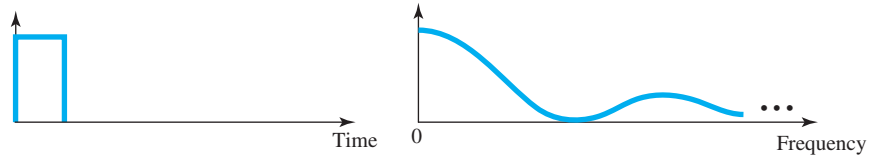
Based on Fourier analysis (See Appendix E), a digital signal is a composite analog signal. The bandwidth is infinite, as you may have guessed. We can intuitively come up with this concept when we consider a digital signal. A digital signal, in the time domain, comprises connected vertical and horizontal 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

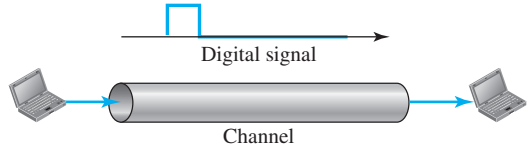
The previous discussion asserts that a digital signal, periodic or nonperiodic, is a composite analog signal with frequencies between zero and infinity. For the remainder of the discussion, let us consider the case of a nonperiodic digital signal, similar to the ones we encounter in data communications.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

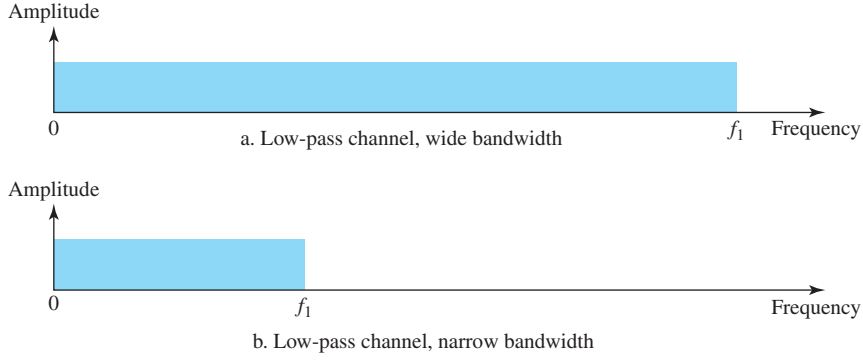
Baseband transmission means sending a digital signal over a channel without changing the digital signal to an analog signal. Figure 3.19 shows **baseband** transmission.

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*

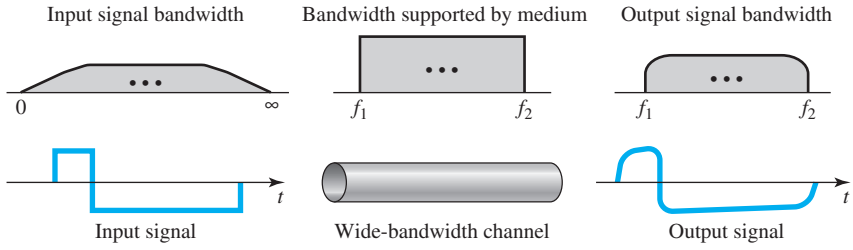


Let us study two cases of a baseband communication: a low-pass channel with a wide bandwidth and one with a limited bandwidth.

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.

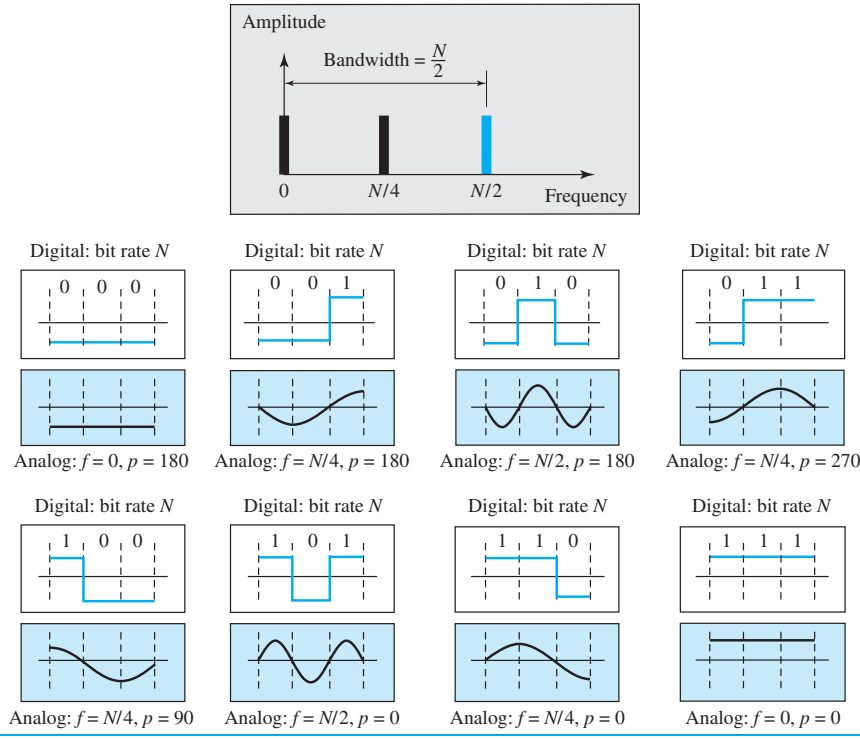
Rough Approximation

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

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

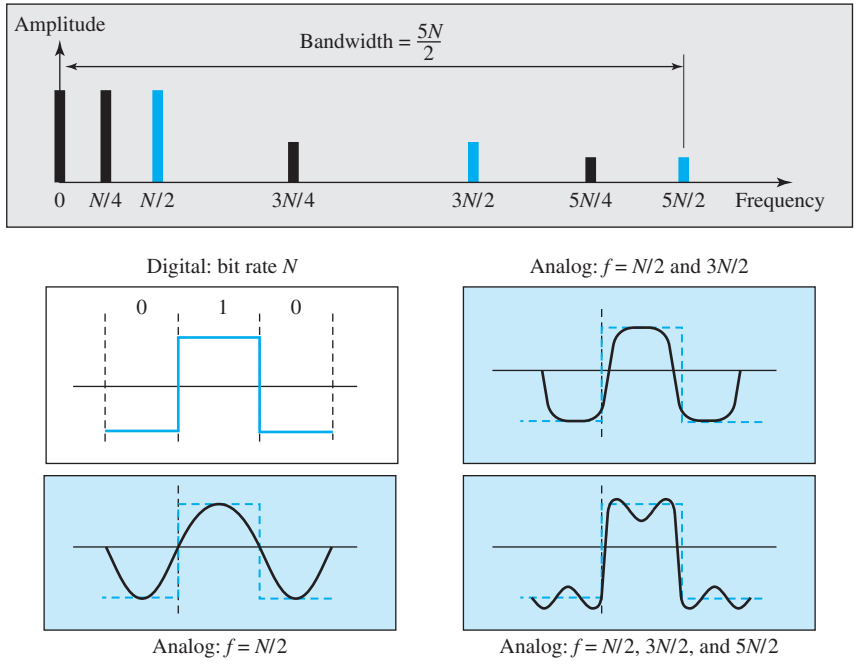
Figure 3.22 *Rough approximation of a digital signal using the first harmonic for worst case*



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?

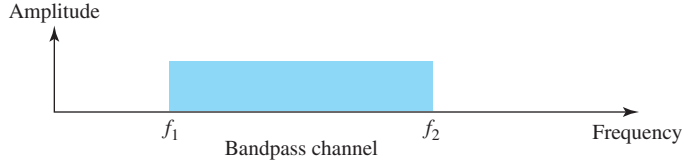
Solution

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*

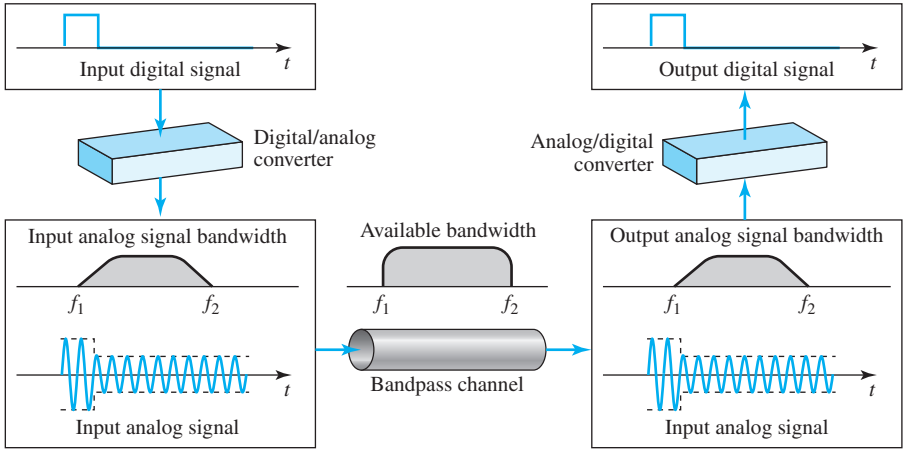


Note that a low-pass channel can be considered a bandpass channel with the lower frequency starting at zero.

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

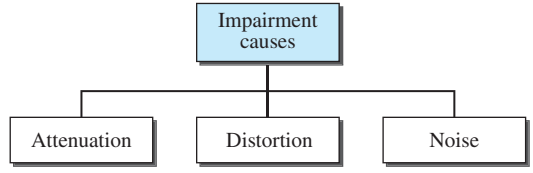
A second example is the digital cellular telephone. For better reception, digital cellular phones convert the analog voice signal to a digital signal (see Chapter 16). Although the bandwidth allocated to a company providing digital cellular phone service is very wide, we still cannot send the digital 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).

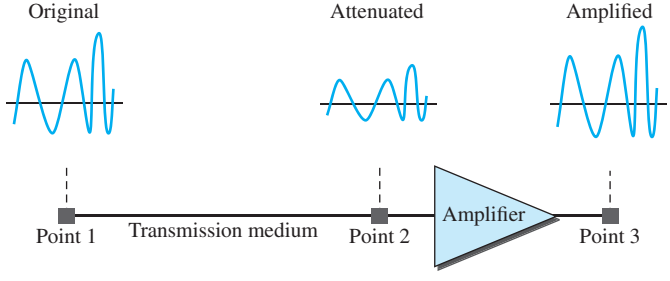
Figure 3.26 *Causes of impairment*



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.

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

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

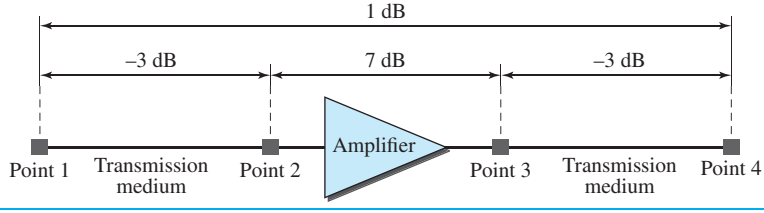
A signal travels through an amplifier, and its power is increased 10 times. This means that $P_2 = 10P_1$. In this case, the amplification (gain of power) can be calculated as

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

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 $5 \times (-0.3) = -1.5$ dB. We can calculate the power as

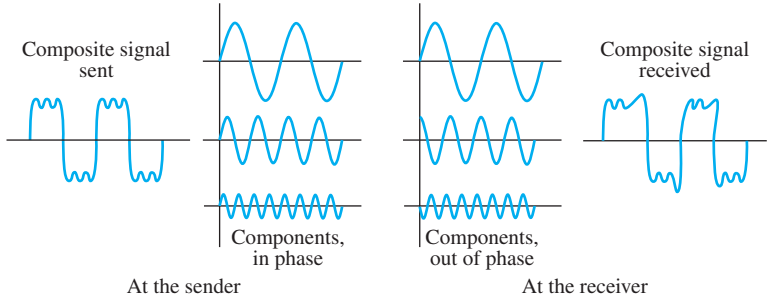
$$\text{dB} = 10 \log_{10} (P_2 / P_1) = -1.5 \quad \longrightarrow \quad (P_2 / P_1) = 10^{-0.15} = 0.71$$

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

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.

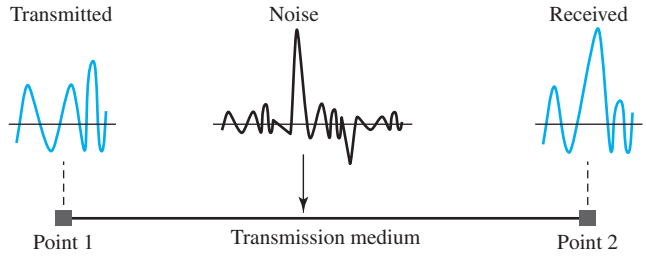
Figure 3.29 *Distortion*



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 appliances. These devices act as a sending antenna, and the transmission medium acts as the receiving antenna. Crosstalk is the effect of one wire on the other. One wire acts as a sending antenna and the other as the receiving antenna. Impulse noise is a spike (a signal with high energy in a very short time) that comes from power lines, lightning, and so on. Figure 3.30 shows the effect of noise on a signal. We discuss error in Chapter 10.

Figure 3.30 *Noise*



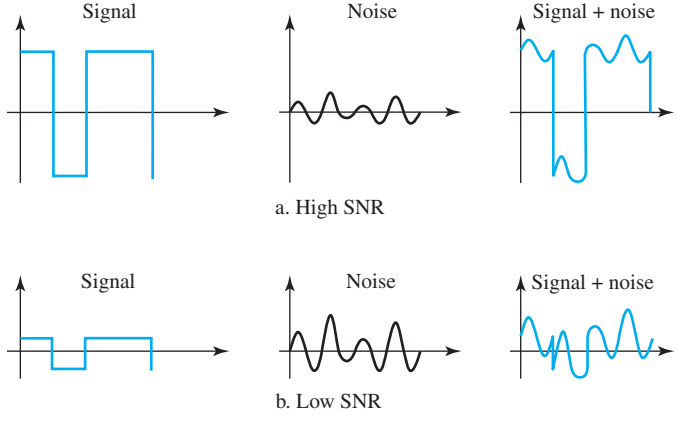
Signal-to-Noise Ratio (SNR)

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

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

We need to consider the average signal power and the average noise power because these may change with time. Figure 3.31 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, SNR_{dB}, defined as

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

Example 3.31

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

Solution

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

$$\mathbf{SNR = (10{,}000\ \mu w) / (1\ \mu w) = 10{,}000 \quad SNR_{dB} = 10 \log_{10} 10{,}000 = 10 \log_{10} 10^4 = 40}$$

Example 3.32

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

$$\mathbf{SNR = (signal\ power) / 0 = \infty \longrightarrow SNR_{dB} = 10 \log_{10} \infty = \infty}$$

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

3.5 DATA RATE LIMITS

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

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

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

3.5.1 Noiseless Channel: Nyquist Bit Rate

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

$$\mathbf{BitRate = 2 \times bandwidth \times \log_2 \mathit{L}}$$

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

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

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

Example 3.33

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

Solution

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

Example 3.34

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

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

Example 3.35

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

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

Example 3.36

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

Solution

We can use the Nyquist formula as shown:

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

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

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:

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

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

Example 3.37

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

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

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

Example 3.38

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

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

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

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

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

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 $C = B \times \text{SNR}_{\text{dB}}$. 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.

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

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.

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

Propagation Time

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

$$\textbf{Propagation time} = \textbf{Distance} / \textbf{(Propagation Speed)}$$

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

Example 3.45

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

Solution

We can calculate the propagation time as

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

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

Transmission Time

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

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

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

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

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

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

Example 3.47

What are the propagation time and the transmission time for a 5-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

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

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

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

Queuing Time

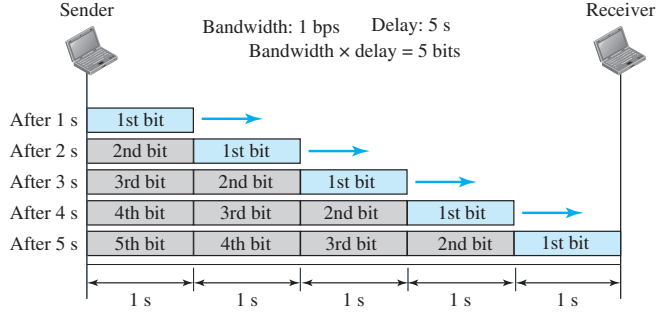
The third component in latency is the **queuing time**, the time needed for each intermediate or end device to hold the message before it can be processed. The queuing time is not a fixed factor; it changes with the load imposed on the network. When there is heavy traffic on the network, the queuing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

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*

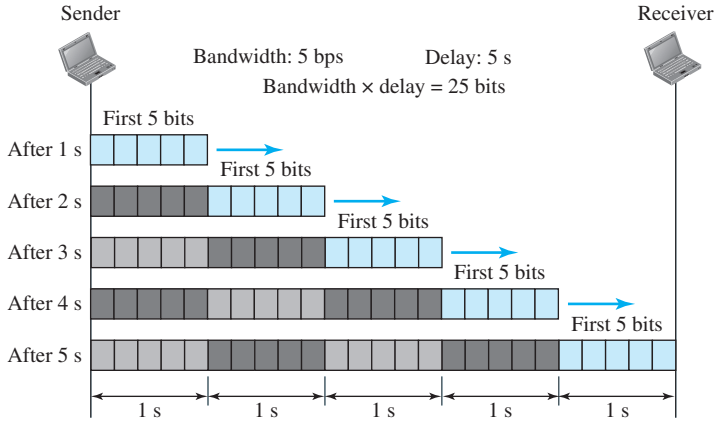


Let us assume that we have a link with a bandwidth of 1 bps (unrealistic, but good for demonstration purposes). We also assume that the delay of the link is 5 s (also unrealistic). We want to see what the bandwidth-delay product means in this case. Looking at the figure, we can say that this product 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 $5 \times 5 = 25$ 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*



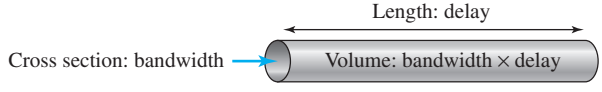
of bandwidth and delay; we need to fill up the full-duplex channel (two directions). The sender should send a burst of data of (2 × bandwidth × delay) bits. The sender then waits for receiver acknowledgment for part of the burst before sending another burst. The amount 2 × bandwidth × delay is the number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

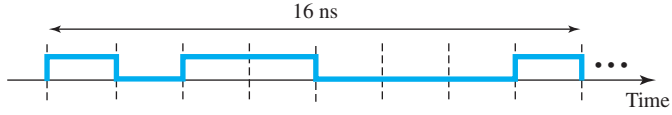
a. How long does it take to send out 10 bits?

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?

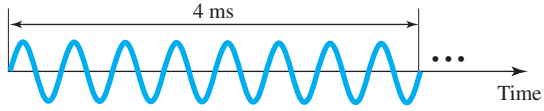
P3-9. What is the bit rate for the signal in Figure 3.35?

Figure 3.35 *Problem P3-9*



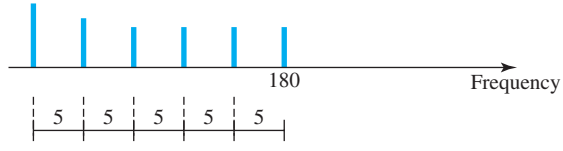
P3-10. What is the frequency of the signal in Figure 3.36?

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 $SNR_{dB} = 40$

b. Bandwidth: 200 KHz $SNR_{dB} = 4$

c. Bandwidth: 1 MHz $SNR_{dB} = 20$

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.

CHAPTER 4

Digital Transmission

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

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

- The first section discusses digital-to-digital conversion. Line coding is used to convert digital data to a digital signal. Several common schemes are discussed. The section also describes block coding, which is used to create redundancy in the digital data before they are encoded as a digital signal. Redundancy is used as an inherent error detecting tool. The last topic in this section discusses scrambling, a technique used for digital-to-digital conversion in long-distance transmission.
- The second section discusses analog-to-digital conversion. Pulse code modulation is described as the main method used to sample an analog signal. Delta modulation is used to improve the efficiency of the pulse code modulation.
- The third section discusses transmission modes. When we want to transmit data digitally, we need to think about parallel or serial transmission. In parallel transmission, we send multiple bits at a time; in serial transmission, we send one bit at a time.

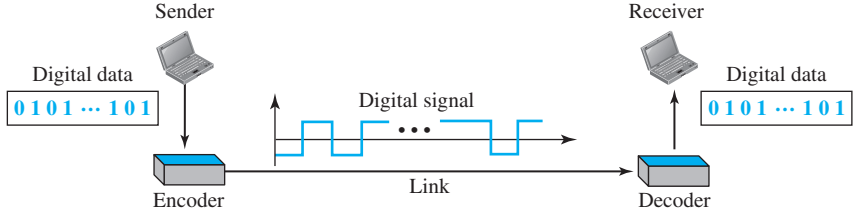
4.1 DIGITAL-TO-DIGITAL CONVERSION

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

4.1.1 Line Coding

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

Figure 4.1 *Line coding and decoding*



Characteristics

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

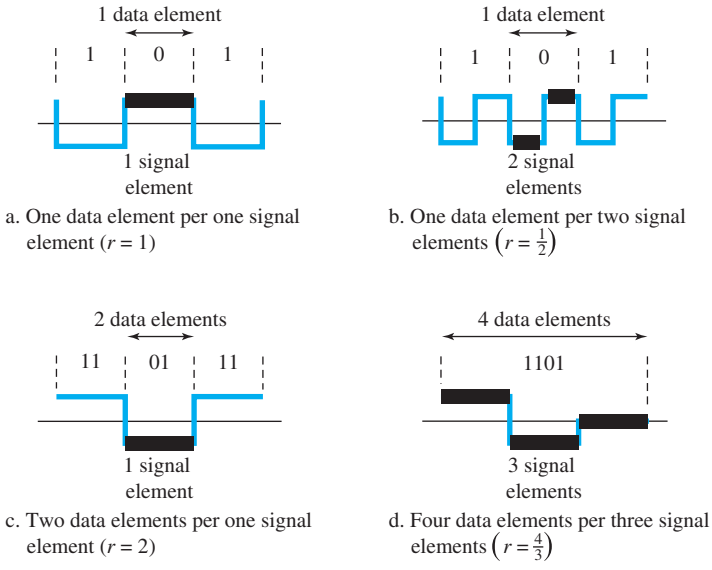
Signal Element Versus Data Element

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

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

In part a of the figure, one data element is carried by one signal element ($r = 1$). In part b of the figure, we need two signal elements (two transitions) to carry each data element ($r = \frac{1}{2}$). We will see later that the extra signal element is needed to guarantee synchronization. In part c of the figure, a signal element carries two data elements ($r = 2$).

Figure 4.2 *Signal element versus data element*



Finally, in part d, a group of 4 bits is being carried by a group of three signal elements ($r = 4/3$). For every line coding scheme we discuss, we will give the value of r.

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

Data Rate Versus Signal Rate

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

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

We now need to consider the relationship between data rate (N) and signal rate (S)

$$S = N/r$$

in which r has been previously defined. This relationship, of course, depends on the value of r. It also depends on the data pattern. If we have a data pattern of all 1s or all 0s, the signal rate may be different from a data pattern of alternating 0s and 1s. To

derive a formula for the relationship, we need to define three cases: the worst, best, and average. The worst case is when we need the maximum signal rate; the best case is when we need the minimum. In data communications, we are usually interested in the average case. We can formulate the relationship between data rate and signal rate as

$$S_{ave} = c \times N \times (1/r) \qquad \textbf{baud}$$

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

Example 4.1

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

Solution

We assume that the average value of c is 1/2. The baud rate is then

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

Bandwidth

We discussed in Chapter 3 that a digital signal that carries information is nonperiodic. We also showed that the bandwidth of a nonperiodic signal is continuous with an infinite range. However, most digital signals we encounter in real life have a bandwidth with finite values. In other words, the bandwidth is theoretically infinite, but many of the components have such a small amplitude that they can be ignored. The effective bandwidth is finite. From now on, when we talk about the bandwidth of a digital signal, we need to remember that we are talking about this effective bandwidth.

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

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

$$B_{min} = c \times N \times (1 / r)$$

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

$$N_{max} = (1 / c) \times B \times r$$

Example 4.2

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

Solution

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

$$N_{max} = (1/c) \times B \times r = 2 \times B \times \log_2 L$$

Baseline Wandering

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

DC Components

When the voltage level in a digital signal is constant for a while, the spectrum creates very low frequencies (results of Fourier analysis). These frequencies around zero, called DC (direct-current) *components,* present problems for a system that cannot pass low frequencies or a system that uses electrical coupling (via a transformer). We can say that DC component means 0/1 parity that can cause base-line wondering. For example, a telephone line cannot pass frequencies below 200 Hz. Also a long-distance link may use one or more transformers to isolate different parts of the line electrically. For these systems, we need a scheme with no **DC component.**

Self-synchronization

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

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

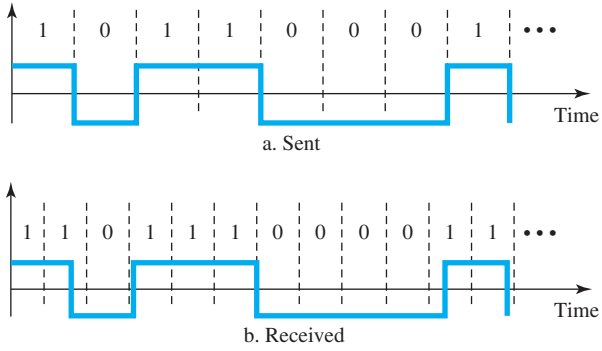
Example 4.3

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

Solution

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

Figure 4.3 *Effect of lack of synchronization*



1000 bits sent → 1001 bits received → 1 extra bps

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

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

Built-in Error Detection

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

Immunity to Noise and Interference

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

Complexity

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

4.1.2 Line Coding Schemes

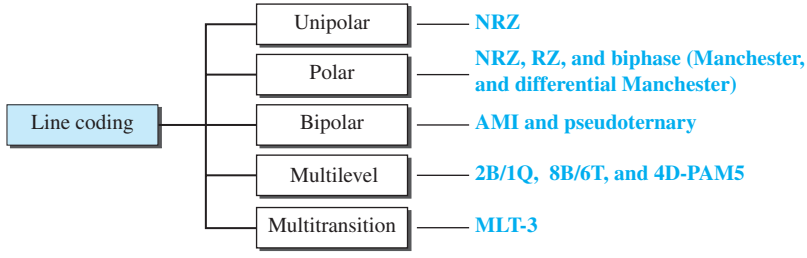
We can roughly divide line coding schemes into five broad categories, as shown in Figure 4.4.

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

Unipolar Scheme

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

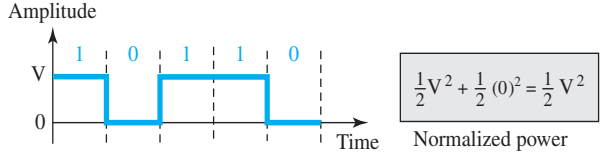
Figure 4.4 *Line coding schemes*



NRZ (Non-Return-to-Zero)

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

Figure 4.5 *Unipolar NRZ scheme*



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

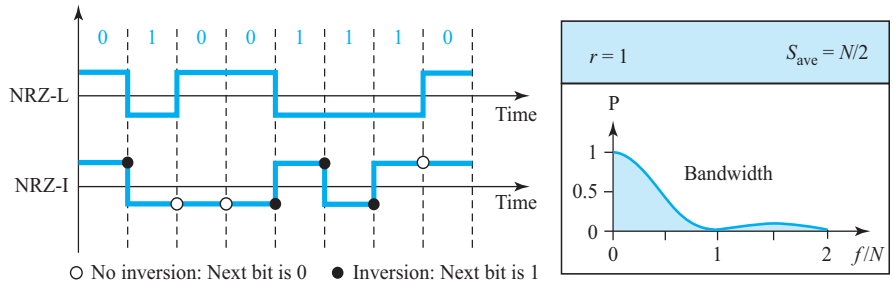
Polar Schemes

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

Non-Return-to-Zero (NRZ)

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

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



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

Let us compare these two schemes based on the criteria we previously defined. Although baseline wandering is a problem for both variations, it is twice as severe in NRZ-L. If there is a long sequence of 0s or 1s in NRZ-L, the average signal power becomes skewed. The receiver might have difficulty discerning the bit value. In NRZ-I this problem occurs only for a long sequence of 0s. If somehow we can eliminate the long sequence of 0s, we can avoid baseline wandering. We will see shortly how this can be done.

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

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

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

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

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

Example 4.4

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

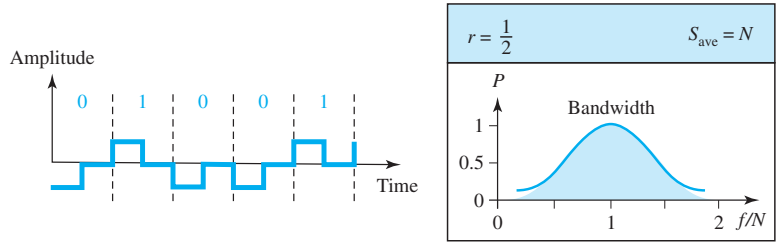
Solution

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

Return-to-Zero (RZ)

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

Figure 4.7 *Polar RZ scheme*

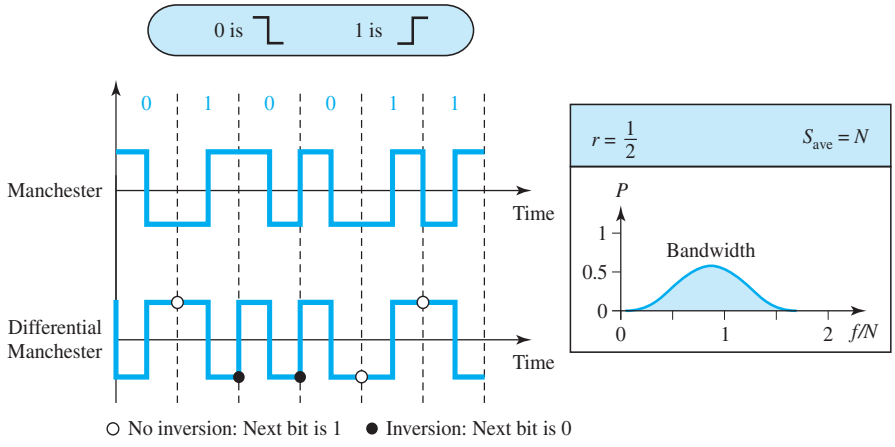


Biphase: Manchester and Differential Manchester

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

The Manchester scheme overcomes several problems associated with NRZ-L, and differential Manchester overcomes several problems associated with NRZ-I. First, there

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



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

is no baseline wandering. There is no DC component because each bit has a positive and negative voltage contribution. The only drawback is the signal rate. The signal rate for Manchester and differential Manchester is double that for NRZ. The reason is that there is always one transition at the middle of the bit and maybe one transition at the end of each bit. Figure 4.8 shows both Manchester and differential Manchester encoding schemes. Note that Manchester and differential Manchester schemes are also called **biphase** schemes.

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

Bipolar Schemes

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

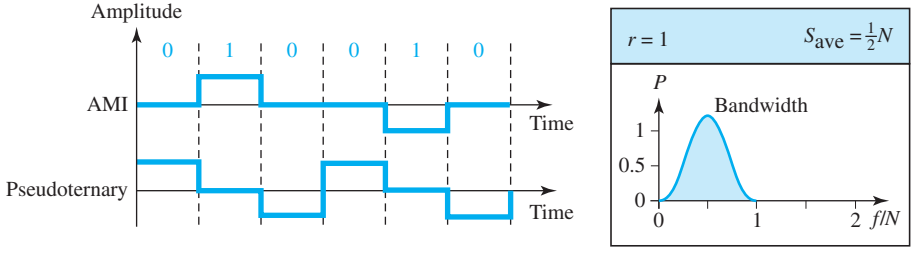
In bipolar encoding, we use three levels: positive, zero, and negative.

AMI and Pseudoternary

Figure 4.9 shows two variations of bipolar encoding: AMI and pseudoternary. A common bipolar encoding scheme is called bipolar **alternate mark inversion (AMI).** In the term *alternate mark inversion,* the word *mark* comes from telegraphy and means 1. So AMI means alternate 1 inversion. A neutral zero voltage represents binary 0. Binary

1s are represented by alternating positive and negative voltages. A variation of AMI encoding is called **pseudoternary** in which the 1 bit is encoded as a zero voltage and the 0 bit is encoded as alternating positive and negative voltages.

Figure 4.9 *Bipolar schemes: AMI and pseudoternary*



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

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

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

Multilevel Schemes

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

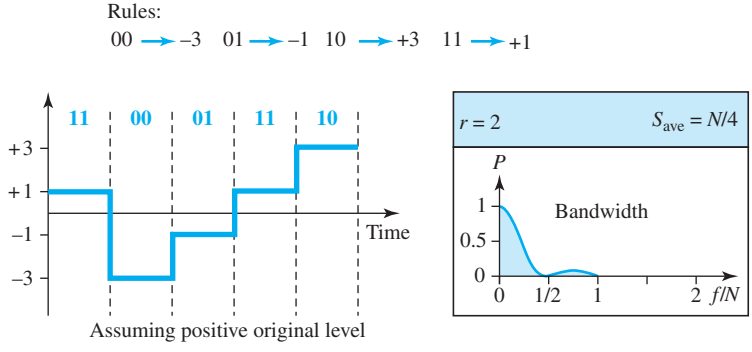
The code designers have classified these types of coding as *mBnL*, where *m* is the length of the binary pattern, *B* means binary data, *n* is the length of the signal pattern, and *L* is the number of levels in the signaling. A letter is often used in place of *L*: *B* (binary) for $L = 2$, *T* (ternary)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

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

2B1Q

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

Figure 4.10 *Multilevel: 2B1Q scheme*



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

The 2B1Q scheme is used in DSL (Digital Subscriber Line) technology to provide a high-speed connection to the Internet by using subscriber telephone lines (see Chapter 14).

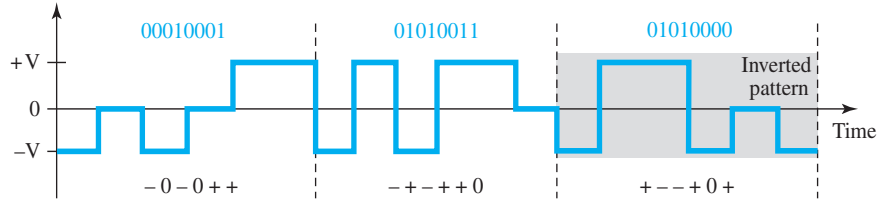
8B6T

A very interesting scheme is **eight binary, six ternary (8B6T).** This code is used with 100BASE-4T cable, as we will see in Chapter 13. The idea is to encode a pattern of 8 bits as a pattern of six signal elements, where the signal has three levels (ternary). In this type of scheme, we can have $2^8 = 256$ different data patterns and $3^6 = 729$ different signal patterns. The mapping table is shown in Appendix F. There are $729 - 256 = 473$ redundant signal elements that provide synchronization and error detection. Part of the

redundancy is also used to provide DC balance. Each signal pattern has a weight of 0 or +1 DC values. This means that there is no pattern with the weight −1. To make the whole stream DC-balanced, the sender keeps track of the weight. If two groups of weight 1 are encountered one after another, the first one is sent as is, while the next one is totally inverted to give a weight of −1.

Figure 4.11 shows an example of three data patterns encoded as three signal patterns. The three possible signal levels are represented as −, 0, and +. The first 8-bit pattern 00010001 is encoded as the signal pattern − 0 − 0 + + with weight 0; the second 8-bit pattern 01010011 is encoded as − + − + + 0 with weight +1. The third 8-bit pattern 01010000 should be encoded as + − − + 0 + with weight +1. To create DC balance, the sender inverts the actual signal. The receiver can easily recognize that this is an inverted pattern because the weight is −1. The pattern is inverted before decoding.

Figure 4.11 *Multilevel: 8B6T scheme*



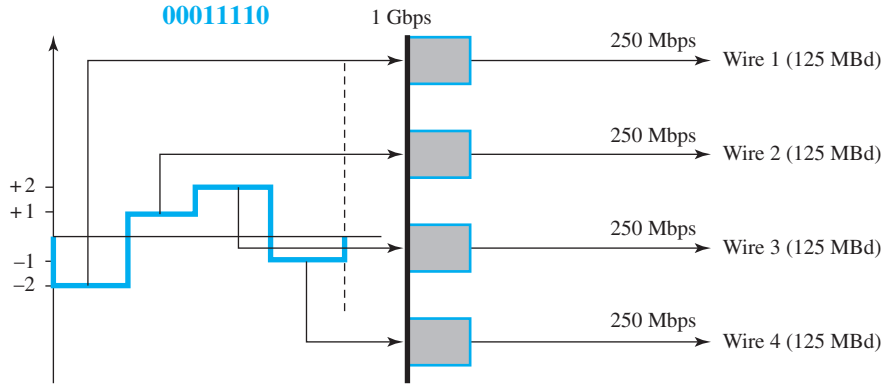
The average signal rate of the scheme is theoretically $S_{ave} = \frac{1}{2} \times N \times \frac{6}{8}$; in practice the minimum bandwidth is very close to 6*N*/8.

4D-PAM5

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

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

Figure 4.12 *Multilevel: 4D-PAM5 scheme*



Multitransition: MLT-3

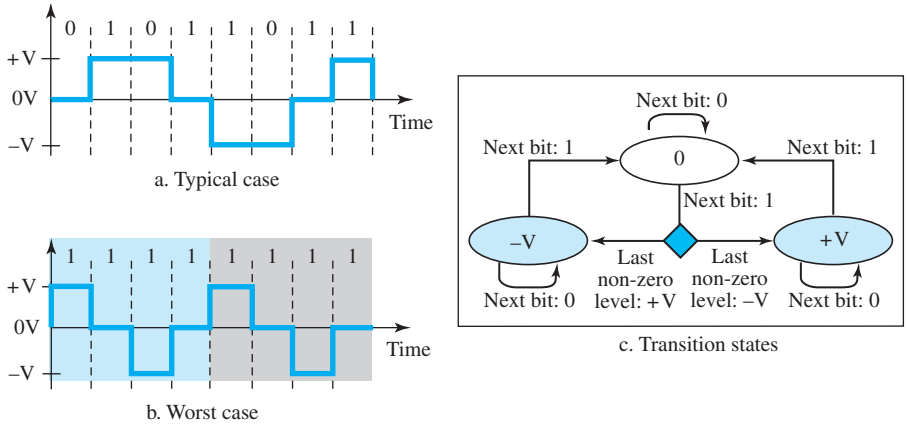
NRZ-I and differential Manchester are classified as differential encoding but use two transition rules to encode binary data (no inversion, inversion). If we have a signal with more than two levels, we can design a differential encoding scheme with more than two transition rules. MLT-3 is one of them.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

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

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

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

Figure 4.13 *Multitransition: MLT-3 scheme*



Summary of Line Coding Schemes

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

Table 4.1 *Summary of line coding schemes*

Category	*Scheme*	*Bandwidth (average)*	*Characteristics*
Unipolar	NRZ	$B = N/2$	Costly, no self-synchronization if long 0s or 1s, DC
Polar	NRZ-L	$B = N/2$	No self-synchronization if long 0s or 1s, DC
	NRZ-I	$B = N/2$	No self-synchronization for long 0s, DC
	Biphase	$B = N$	Self-synchronization, no DC, high bandwidth
Bipolar	AMI	$B = N/2$	No self-synchronization for long 0s, DC
Multilevel	2B1Q	$B = N/4$	No self-synchronization for long same double bits
	8B6T	$B = 3N/4$	Self-synchronization, no DC
	4D-PAM5	$B = N/8$	Self-synchronization, no DC
Multitransition	MLT-3	$B = N/3$	No self-synchronization for long 0s

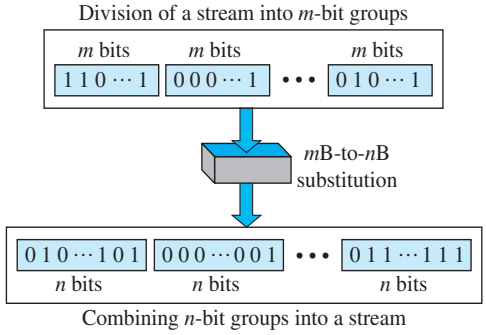
4.1.3 Block Coding

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

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

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

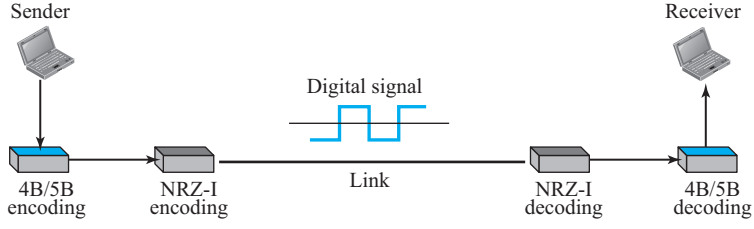
Figure 4.14 *Block coding concept*



4B/5B

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

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



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

Table 4.2 *4B/5B mapping codes*

Data Sequence	*Encoded Sequence*	*Control Sequence*	*Encoded Sequence*
0000	11110	Q (Quiet)	00000
0001	01001	I (Idle)	11111
0010	10100	H (Halt)	00100
0011	10101	J (Start delimiter)	11000
0100	01010	K (Start delimiter)	10001
0101	01011	T (End delimiter)	01101
0110	01110	S (Set)	11001
0111	01111	R (Reset)	00111
1000	10010		
1001	10011		
1010	10110		
1011	10111		
1100	11010		
1101	11011		
1110	11100		
1111	11101		

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

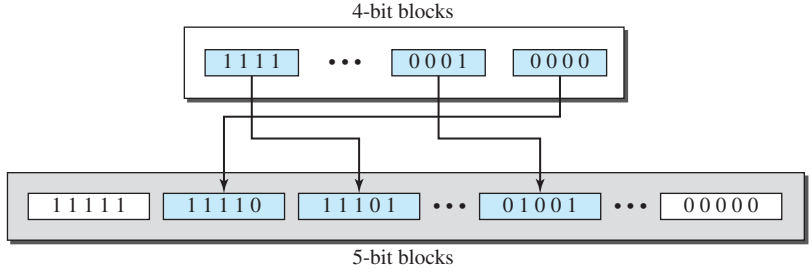
Example 4.5

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

Solution

First 4B/5B block coding increases the bit rate to 1.25 Mbps. The minimum bandwidth using NRZ-I is $N/2$ or 625 kHz. The Manchester scheme needs a minimum bandwidth of 1 MHz. The

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

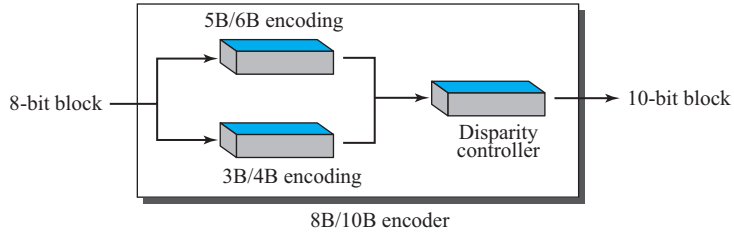


first choice needs a lower bandwidth, but has a DC component problem; the second choice needs a higher bandwidth, but does not have a DC component problem.

8B/10B

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

Figure 4.17 *8B/10B block encoding*

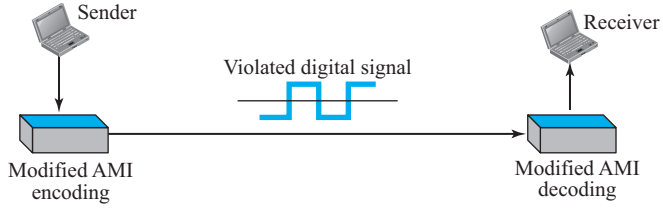


The five most significant bits of a 10-bit block are fed into the 5B/6B encoder; the three least significant bits are fed into a 3B/4B encoder. The split is done to simplify the mapping table. To prevent a long run of consecutive 0s or 1s, the code uses a disparity controller which keeps track of excess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

4.1.4 Scrambling

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

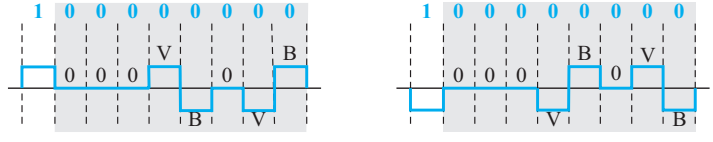
Figure 4.18 *AMI used with scrambling*



B8ZS

Bipolar with 8-zero substitution (B8ZS) is commonly used in North America. In this technique, eight consecutive zero-level voltages are replaced by the sequence **000VB0VB.** The V in the sequence denotes *violation;* this is a nonzero voltage that breaks an AMI rule of encoding (opposite polarity from the previous). The B in the sequence denotes *bipolar,* which means a nonzero level voltage in accordance with the AMI rule. There are two cases, as shown in Figure 4.19.

Figure 4.19 *Two cases of B8ZS scrambling technique*



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

B8ZS substitutes eight consecutive zeros with 000VB0VB.

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

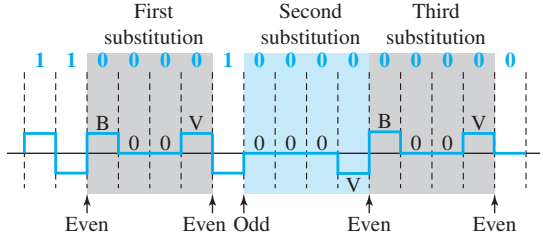
HDB3

High-density bipolar 3-zero (HDB3) is commonly used outside of North America. In this technique, which is more conservative than B8ZS, four consecutive zero-level voltages are replaced with a sequence of **000V** or **B00V.** The reason for two different substitutions is to maintain the even number of nonzero pulses after each substitution. The two rules can be stated as follows:

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

Figure 4.20 shows an example.

Figure 4.20 *Different situations in HDB3 scrambling technique*



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

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

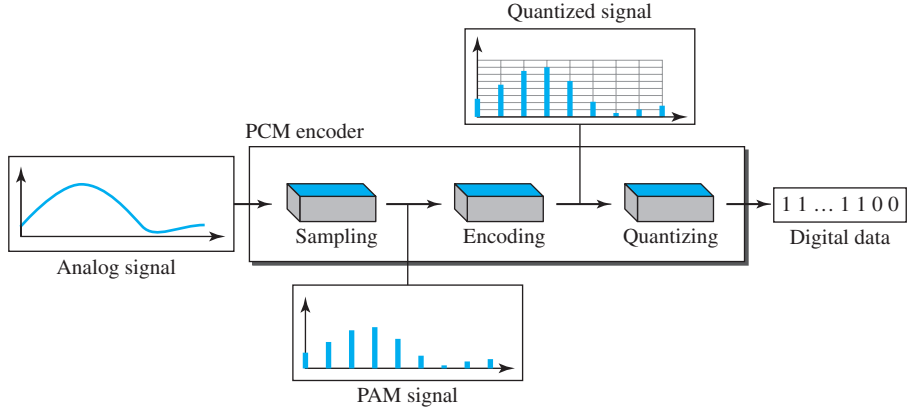
4.2 ANALOG-TO-DIGITAL CONVERSION

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

4.2.1 Pulse Code Modulation (PCM)

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

Figure 4.21 *Components of PCM encoder*



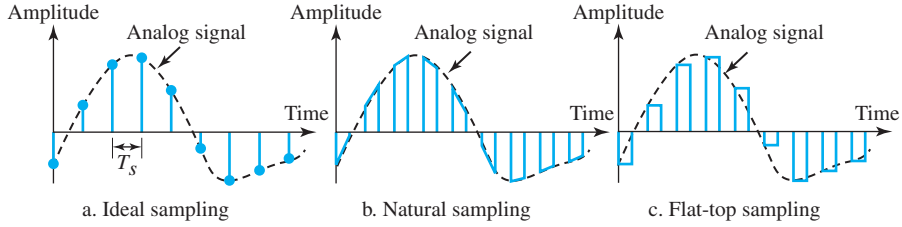
1. The analog signal is sampled.
2. The sampled signal is quantized.
3. The quantized values are encoded as streams of bits.

Sampling

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

In ideal sampling, pulses from the analog signal are sampled. This is an ideal sampling method and cannot be easily implemented. In natural sampling, a high-speed switch is turned on for only the small period of time when the sampling occurs. The result is a sequence of samples that retains the shape of the analog signal. The most

Figure 4.22 *Three different sampling methods for PCM*



common sampling method, called ***sample and hold,*** however, creates flat-top samples by using a circuit.

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

Sampling Rate

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

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

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

Example 4.6

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

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

Example 4.7

As an interesting example, let us see what happens if we sample a periodic event such as the revolution of a hand of a clock. The second hand of a clock has a period of 60 s. According to the

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

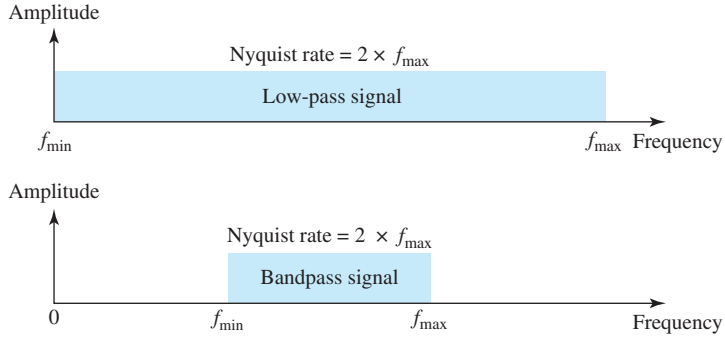
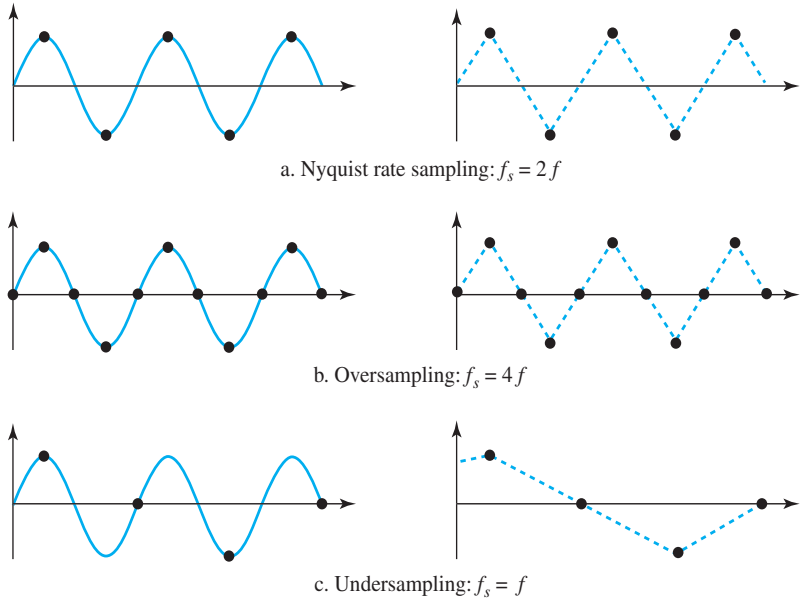
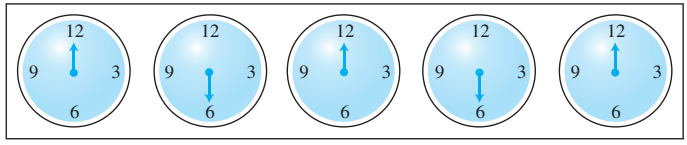


Figure 4.24 *Recovery of a sampled sine wave for different sampling rates*



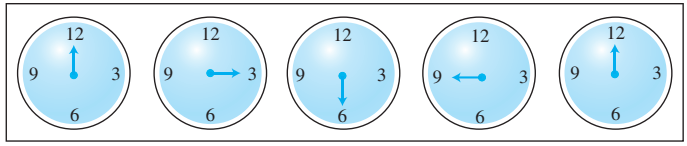
Nyquist theorem, we need to sample the hand (take and send a picture) every 30 s ($T_s = \frac{1}{2}T$ or $f_s = 2f$). In Figure 4.25a, the sample points, in order, are 12, 6, 12, 6, 12, and 6. The receiver of the samples cannot tell if the clock is moving forward or backward. In part b, we sample at double the Nyquist rate (every 15 s). The sample points, in order, are 12, 3, 6, 9, and 12. The clock is moving forward. In part c, we sample below the Nyquist rate ($T_s = \frac{3}{4}T$ or $f_s = \frac{4}{3}f$). The sample

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



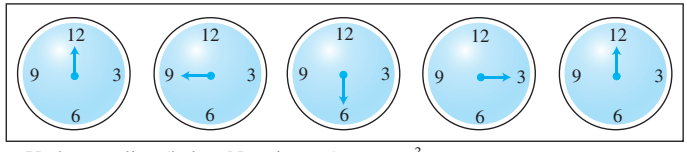
Samples can mean that the clock is moving either forward or backward. (12-6-12-6-12)

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



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

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



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

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

points, in order, are 12, 9, 6, 3, and 12. Although the clock is moving forward, the receiver thinks that the clock is moving backward.

Example 4.8

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

Example 4.9

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

Example 4.10

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

Solution

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

Example 4.11

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

Solution

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

Quantization

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

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

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

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

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

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

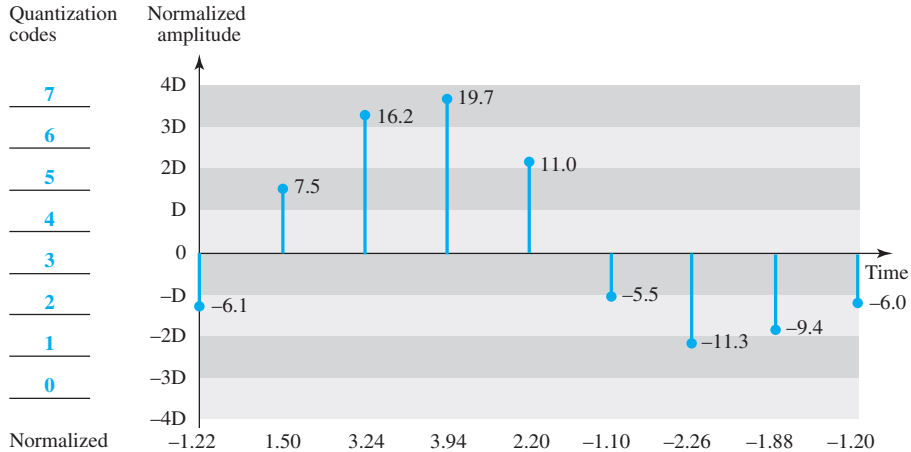
Quantization Levels

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

Quantization Error

One important issue is the error created in the quantization process. (Later, we will see how this affects high-speed modems.) Quantization is an approximation process. The

Figure 4.26 *Quantization and encoding of a sampled signal*



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

input values to the quantizer are the real values; the output values are the approximated values. The output values are chosen to be the middle value in the zone. If the input value is also at the middle of the zone, there is no quantization error; otherwise, there is an error. In the previous example, the normalized amplitude of the third sample is 3.24, but the normalized quantized value is 3.50. This means that there is an error of +0.26. The value of the error for any sample is less than $\Delta/2$. In other words, we have $-\Delta/2 \leq \text{error} \leq \Delta/2$.

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

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

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

Example 4.12

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

Solution

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

Example 4.13

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

Solution

We can calculate the number of bits as

$$\mathbf{SNR_{dB} = 6.02}n_b \mathbf{+ 1.76 = 40} \rightarrow n = 6.35$$

Telephone companies usually assign 7 or 8 bits per sample.

Uniform Versus Nonuniform Quantization

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

Encoding

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

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

Example 4.14

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

Solution

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

$$\textbf{Sampling rate} = 4000 \times 2 = \textbf{8000 samples/s}$$

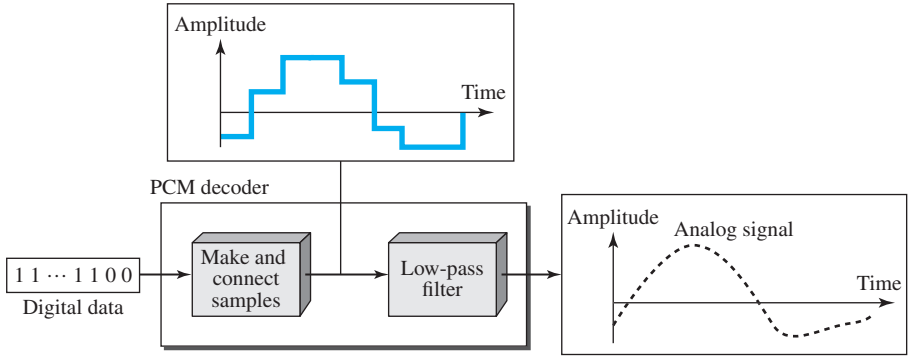
$$\textbf{Bit rate} = 8000 \times 8 = \textbf{64,000 bps} = \textbf{64 kbps}$$

Original Signal Recovery

The recovery of the original signal requires the PCM decoder. The decoder first uses circuitry to convert the code words into a pulse that holds the amplitude until the next

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

Figure 4.27 *Components of a PCM decoder*



PCM Bandwidth

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

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

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

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

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

Example 4.15

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

Maximum Data Rate of a Channel

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

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

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

Minimum Required Bandwidth

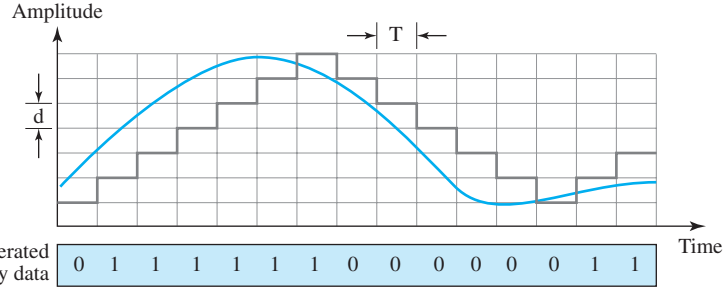
The previous argument can give us the minimum bandwidth if the data rate and the number of signal levels are fixed. We can say

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

4.2.2 Delta Modulation (DM)

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

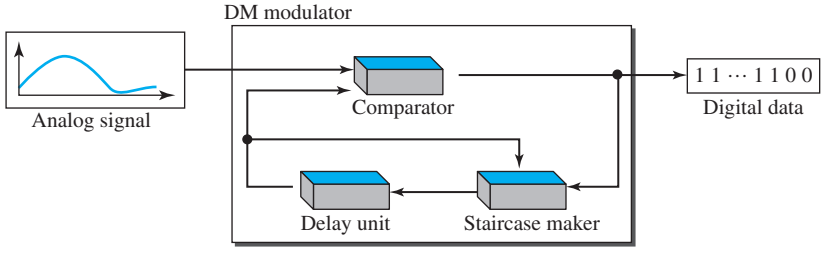
Figure 4.28 *The process of delta modulation*



Modulator

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

Figure 4.29 *Delta modulation components*



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

Demodulator

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

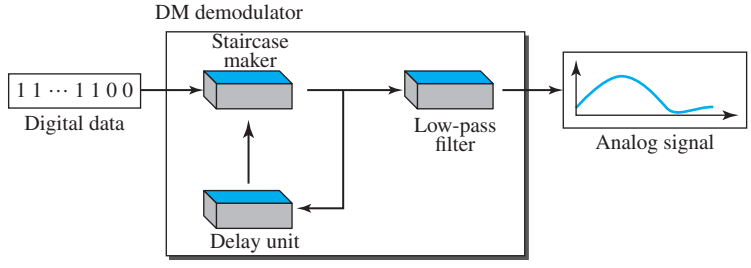
Adaptive DM

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

Quantization Error

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

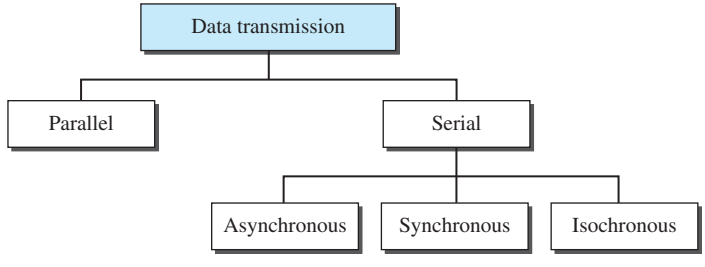
Figure 4.30 *Delta demodulation components*



4.3 TRANSMISSION MODES

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

Figure 4.31 *Data transmission and modes*



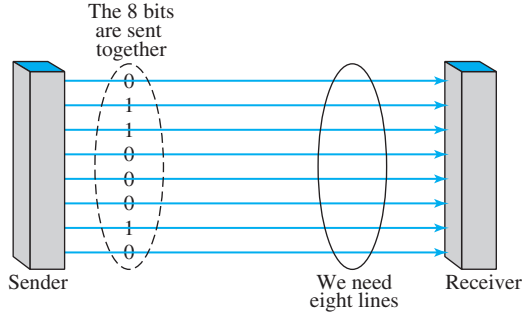
4.3.1 Parallel Transmission

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

The mechanism for parallel transmission is a conceptually simple one: Use n wires to send n bits at one time. That way each bit has its own wire, and all n bits of one

group can be transmitted with each clock tick from one device to another. Figure 4.32 shows how parallel transmission works for $n = 8$. Typically, the eight wires are bundled in a cable with a connector at each end.

Figure 4.32 *Parallel transmission*

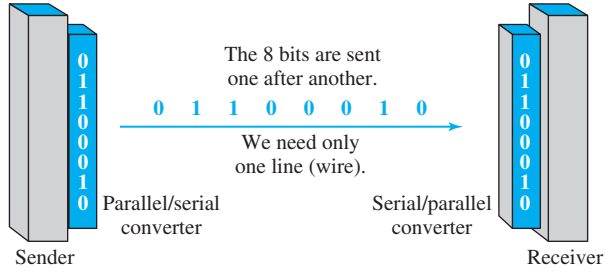


The advantage of parallel transmission is speed. All else being equal, parallel transmission can increase the transfer speed by a factor of n over serial transmission. But there is a significant disadvantage: cost. Parallel transmission requires n communication lines (wires in the example) just to transmit the data stream. Because this is expensive, parallel transmission is usually limited to short distances.

4.3.2 Serial Transmission

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

Figure 4.33 *Serial transmission*



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

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

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

Asynchronous Transmission

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

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

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

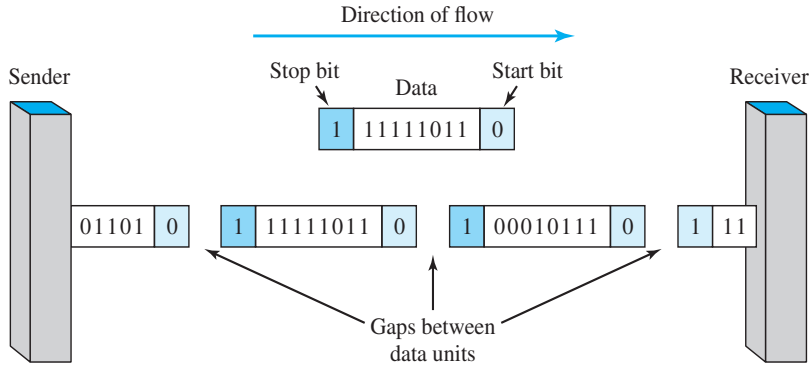
The start and stop bits and the gap alert the receiver to the beginning and end of each byte and allow it to synchronize with the data stream. This mechanism is called *asynchronous* because, at the byte level, the sender and receiver do not have to be synchronized. But within each byte, the receiver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

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

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

The addition of stop and start bits and the insertion of gaps into the bit stream make asynchronous transmission slower than forms of transmission that can operate

Figure 4.34 *Asynchronous transmission*



without the addition of control information. But it is cheap and effective, two advantages that make it an attractive choice for situations such as low-speed communication. For example, the connection of a keyboard to a computer is a natural application for asynchronous transmission. A user types only one character at a time, types extremely slowly in data processing terms, and leaves unpredictable gaps of time between characters.

Synchronous Transmission

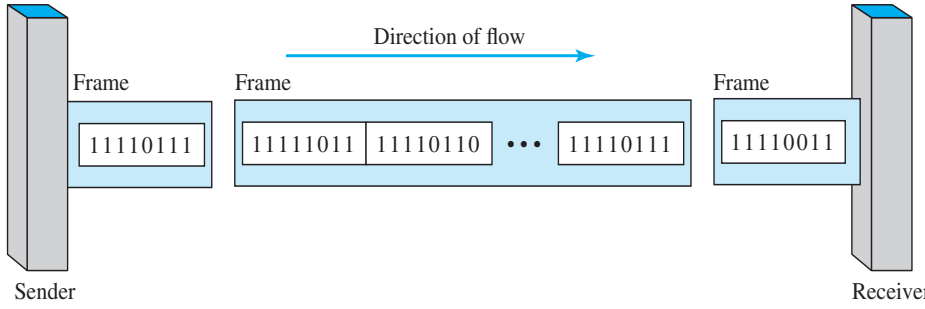
In **synchronous transmission,** the bit stream is combined into longer "frames," which may contain multiple bytes. Each byte, however, is introduced onto the transmission link without a gap between it and the next one. It is left to the receiver to separate the bit stream into bytes for decoding purposes.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

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

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

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

Figure 4.35 *Synchronous transmission*



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

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

Isochronous

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

4.4 END-CHAPTER MATERIALS

4.4.1 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

Digital to digital conversion is discussed in [Pea92], [Cou01], and [Sta04]. Sampling is discussed in [Pea92], [Cou01], and [Sta04]. [Hsu03] gives a good mathematical approach to modulation and sampling. More advanced materials can be found in [Ber96].

4.4.2 Key Terms

adaptive delta modulation
alternate mark inversion (AMI)
analog-to-digital conversion
asynchronous transmission
baseline
baseline wandering
baud rate
biphase
bipolar
bipolar with 8-zero substitution (B8ZS)
bit rate
block coding
companding and expanding
data element
data rate
DC component
delta modulation (DM)
differential Manchester
digital-to-digital conversion
digitization
eight binary/ten binary (8B/10B)
eight-binary, six-ternary (8B6T)
four binary/five binary (4B/5B)
four dimensional, five-level pulse amplitude modulation (4D-PAM5)
high-density bipolar 3-zero (HDB3)
isochronous transmission
line coding
Manchester
modulation rate
multilevel binary
multiline transmission, three-level (MLT-3)
non-return-to-zero (NRZ)
non-return-to-zero, invert (NRZ-I)
non-return-to-zero, level (NRZ-L)
Nyquist theorem
parallel transmission
polar
pseudoternary
pulse amplitude modulation (PAM)
pulse code modulation (PCM)
pulse rate
quantization
quantization error
return-to-zero (RZ)
sample and hold
sampling
sampling rate
scrambling
self-synchronizing
serial transmission
signal element
signal rate
start bit
stop bit
synchronous transmission
transmission mode
two-binary, one quaternary (2B1Q)
unipolar

4.4.3 Summary

Digital-to-digital conversion involves three techniques: line coding, block coding, and scrambling. Line coding is the process of converting digital data to a digital signal. We can roughly divide line coding schemes into five broad categories: unipolar, polar, bipolar, multilevel, and multitransition. Block coding provides redundancy to ensure synchronization and inherent error detection. Block coding is normally referred to as mB/nB coding; it replaces each m-bit group with an n-bit group. Scrambling provides synchronization without increasing the number of bits. Two common scrambling techniques are B8ZS and HDB3.

The most common technique to change an analog signal to digital data (digitization) is called pulse code modulation (PCM). The first step in PCM is sampling. The analog signal is sampled every T_s second, where T_s is the sample interval or period. The inverse of the sampling interval is called the *sampling rate* or *sampling frequency* and denoted by f_s, where $f_s = 1/T_s$. There are three sampling methods—ideal, natural, and flat-top. According to the *Nyquist theorem,* to reproduce the original analog signal, one necessary condition is that the *sampling rate* be at least twice the highest frequency in

the original signal. Other sampling techniques have been developed to reduce the complexity of PCM. The simplest is delta modulation. PCM finds the value of the signal amplitude for each sample; DM finds the change from the previous sample.

While there is only one way to send parallel data, there are three subclasses of serial transmission: asynchronous, synchronous, and isochronous. In asynchronous transmission, we send 1 start bit (0) at the beginning and 1 or more stop bits (1s) at the end of each byte. In synchronous transmission, we send bits one after another without start or stop bits or gaps. It is the responsibility of the receiver to group the bits. The isochronous mode provides synchronization for the entire stream of bits. In other words, it guarantees that the data arrive at a fixed rate.

4.5 PRACTICE SET

4.5.1 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.

4.5.2 Questions

Q4-1. List three techniques of digital-to-digital conversion.

Q4-2. Distinguish between a signal element and a data element.

Q4-3. Distinguish between data rate and signal rate.

Q4-4. Define baseline wandering and its effect on digital transmission.

Q4-5. Define a DC component and its effect on digital transmission.

Q4-6. Define the characteristics of a self-synchronizing signal.

Q4-7. List five line coding schemes discussed in this book.

Q4-8. Define block coding and give its purpose.

Q4-9. Define scrambling and give its purpose.

Q4-10. Compare and contrast PCM and DM.

Q4-11. What are the differences between parallel and serial transmission?

Q4-12. List three different techniques in serial transmission and explain the differences.

4.5.3 Problems

P4-1. Calculate the value of the signal rate for each case in Figure 4.2 if the data rate is 1 Mbps and c = 1/2.

P4-2. In a digital transmission, the sender clock is 0.2 percent faster than the receiver clock. How many extra bits per second does the sender send if the data rate is 1 Mbps?

P4-3. Draw the graph of the NRZ-L scheme using each of the following data streams, assuming that the last signal level has been positive. From the graphs, guess the bandwidth for this scheme using the average number of changes

in the signal level. Compare your guess with the corresponding entry in Table 4.1.

a. 00000000 **b.** 11111111 **c.** 01010101 **d.** 00110011

P4-4. Repeat Problem P4-3 for the NRZ-I scheme.

P4-5. Repeat Problem P4-3 for the Manchester scheme.

P4-6. Repeat Problem P4-3 for the differential Manchester scheme.

P4-7. Repeat Problem P4-3 for the 2B1Q scheme, but use the following data streams.

a. 0000000000000000

b. 1111111111111111

c. 0101010101010101

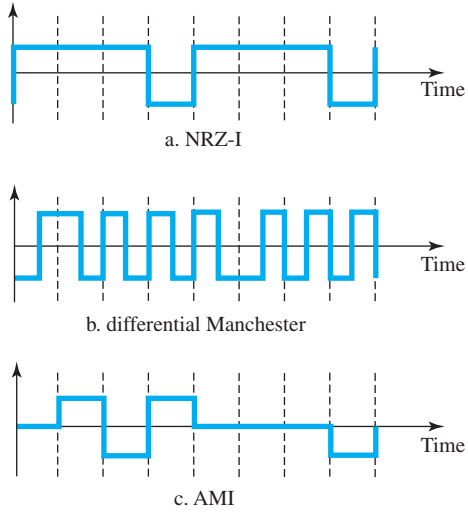
d. 0011001100110011

P4-8. Repeat Problem P4-3 for the MLT-3 scheme, but use the following data streams.

a. 00000000 **b.** 11111111 **c.** 01010101 **d.** 00011000

P4-9. Find the 8-bit data stream for each case depicted in Figure 4.36.

Figure 4.36 *Problem P4-9*



P4-10. An NRZ-I signal has a data rate of 100 Kbps. Using Figure 4.6, calculate the value of the normalized energy (P) for frequencies at 0 Hz, 50 KHz, and 100 KHz.

P4-11. A Manchester signal has a data rate of 100 Kbps. Using Figure 4.8, calculate the value of the normalized energy (P) for frequencies at 0 Hz, 50 KHz, 100 KHz.

P4-12. The input stream to a 4B/5B block encoder is

0100 0000 0000 0000 0000 0001

Answer the following questions:

a. What is the output stream?

b. What is the length of the longest consecutive sequence of 0s in the input?

c. What is the length of the longest consecutive sequence of 0s in the output?

P4-13. How many invalid (unused) code sequences can we have in 5B/6B encoding? How many in 3B/4B encoding?

P4-14. What is the result of scrambling the sequence 11100000000000 using each of the following scrambling techniques? Assume that the last non-zero signal level has been positive.

a. B8ZS

b. HDB3 (The number of nonzero pulses is odd after the last substitution.)

P4-15. What is the Nyquist sampling rate for each of the following signals?

a. A low-pass signal with bandwidth of 200 KHz?

b. A band-pass signal with bandwidth of 200 KHz if the lowest frequency is 100 KHz?

P4-16. We have sampled a low-pass signal with a bandwidth of 200 KHz using 1024 levels of quantization.

a. Calculate the bit rate of the digitized signal.

b. Calculate the SNR_{dB} for this signal.

c. Calculate the PCM bandwidth of this signal.

P4-17. What is the maximum data rate of a channel with a bandwidth of 200 KHz if we use four levels of digital signaling.

P4-18. An analog signal has a bandwidth of 20 KHz. If we sample this signal and send it through a 30 Kbps channel, what is the SNR_{dB}?

P4-19. We have a baseband channel with a 1-MHz bandwidth. What is the data rate for this channel if we use each of the following line coding schemes?

a. NRZ-L **b.** Manchester **c.** MLT-3 **d.** 2B1Q

P4-20. We want to transmit 1000 characters with each character encoded as 8 bits.

a. Find the number of transmitted bits for synchronous transmission.

b. Find the number of transmitted bits for asynchronous transmission.

c. Find the redundancy percent in each case.

4.6 SIMULATION EXPERIMENTS

4.6.1 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.

CHAPTER 5

Analog Transmission

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

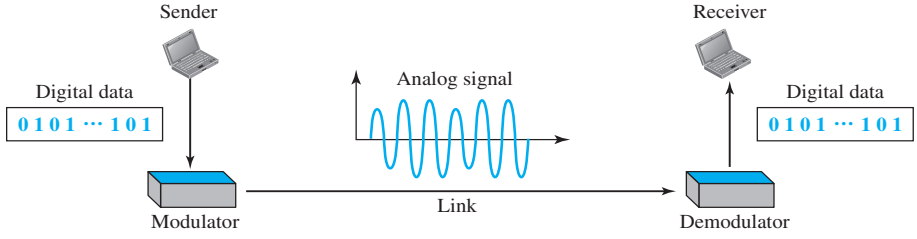
Converting digital data to a bandpass analog signal is traditionally called digital-to-analog conversion. Converting a low-pass analog signal to a bandpass analog signal is traditionally called analog-to-analog conversion. In this chapter, we discuss these two types of conversions in two sections:

- The first section discusses digital-to-analog conversion. The section shows how we can change digital data to an analog signal when a band-pass channel is available. The first method described is called amplitude shift keying (ASK), in which the amplitude of a carrier is changed using the digital data. The second method described is called frequency shift keying (FSK), in which the frequency of a carrier is changed using the digital data. The third method described is called phase shift keying (PSK), in which the phase of a carrier signal is changed to represent digital data. The fourth method described is called quadrature amplitude modulation (QAM), in which both amplitude and phase of a carrier signal are changed to represent digital data.
- The second section discusses analog-to-analog conversion. The section shows how we can change an analog signal to a new analog signal with a smaller bandwidth. The conversion is used when only a band-pass channel is available. The first method is called amplitude modulation (AM), in which the amplitude of a carrier is changed based on the changes in the original analog signal. The second method is called frequency modulation (FM), in which the phase of a carrier is changed based on the changes in the original analog signal. The third method is called phase modulation (PM), in which the phase of a carrier signal is changed to show the changes in the original signal.

5.1 DIGITAL-TO-ANALOG CONVERSION

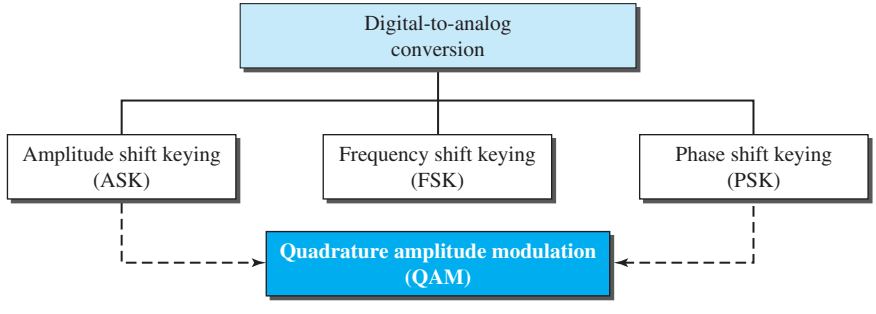
Digital-to-analog conversion is the process of changing one of the characteristics of an analog signal based on the information in digital data. Figure 5.1 shows the relationship between the digital information, the digital-to-analog modulating process, and the resultant analog signal.

Figure 5.1 *Digital-to-analog conversion*



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

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



5.1.1 Aspects of Digital-to-Analog Conversion

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

Data Element Versus Signal Element

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

Data Rate Versus Signal Rate

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

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

where N is the data rate (bps) and r is the number of data elements carried in one signal element. The value of r in analog transmission is $r = \log_2 L$, where L is the number of different signal elements. The same nomenclature is used to simplify the comparisons.

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

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

Example 5.1

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

Solution

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

$$S = N \times (1/r) \quad \text{or} \quad N = S \times r = 1000 \times 4 = 4000 \text{ bps}$$

Example 5.2

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

Solution

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

$$S = N \times 1/r \longrightarrow r = N / S = 8000 / 10{,}000 = 8 \text{ bits/baud}$$
$$r = \log_2 L \longrightarrow L = 2^r = 2^8 = 256$$

Bandwidth

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

Carrier Signal

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

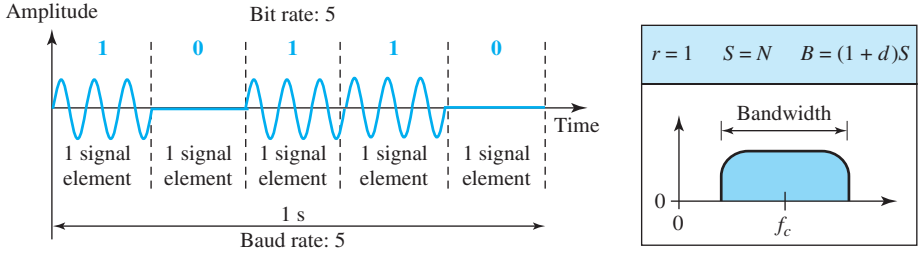
5.1.2 Amplitude Shift Keying

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

Binary ASK (BASK)

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

Figure 5.3 *Binary amplitude shift keying*



Bandwidth for ASK

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

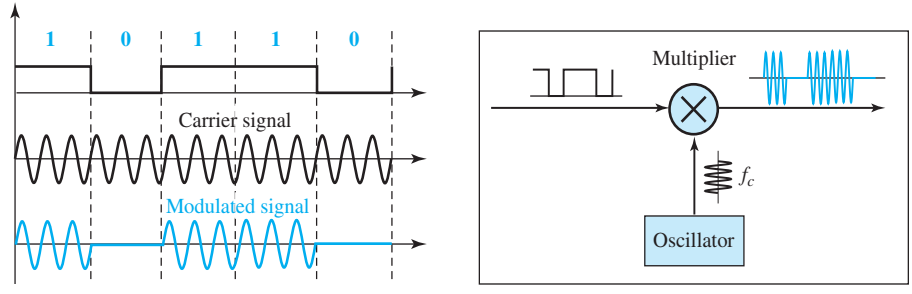
$$B = (1 + d) \times S$$

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

Implementation

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

Figure 5.4 *Implementation of binary ASK*



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

Example 5.3

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

Solution

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

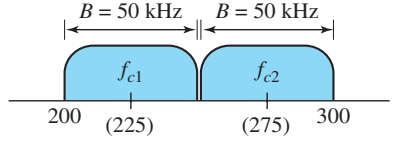
$$\boldsymbol{B = (1 + d) \times S = 2 \times N \times (1/r) = 2 \times N = 100 \text{ kHz} \quad \longrightarrow \quad N = 50 \text{ kbps}}$$

Example 5.4

In data communications, we normally use full-duplex links with communication in both directions. We need to divide the bandwidth into two with two carrier frequencies, as shown in Figure 5.5. The figure shows the positions of two carrier frequencies and the bandwidths. The

available bandwidth for each direction is now 50 kHz, which leaves us with a data rate of 25 kbps in each direction.

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



Multilevel ASK

The above discussion uses only two amplitude levels. We can have multilevel ASK in which there are more than two levels. We can use 4, 8, 16, or more different amplitudes for the signal and modulate the data using 2, 3, 4, or more bits at a time. In these cases, $r = 2$, $r = 3$, $r = 4$, and so on. Although this is not implemented with pure ASK, it is implemented with QAM (as we will see later).

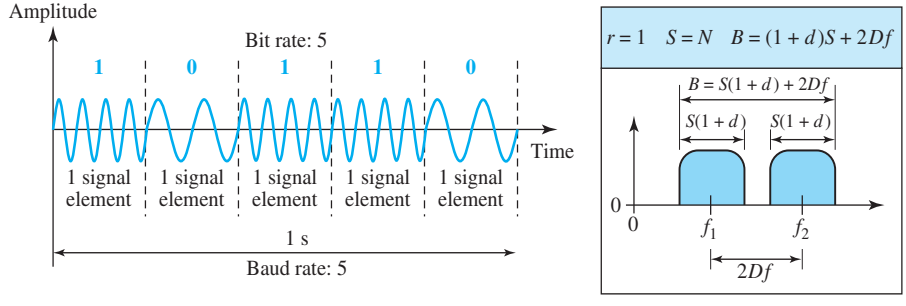
5.1.3 Frequency Shift Keying

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

Binary FSK (BFSK)

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

Figure 5.6 *Binary frequency shift keying*



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

Bandwidth for BFSK

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

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

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

Example 5.5

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

Solution

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

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

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

Implementation

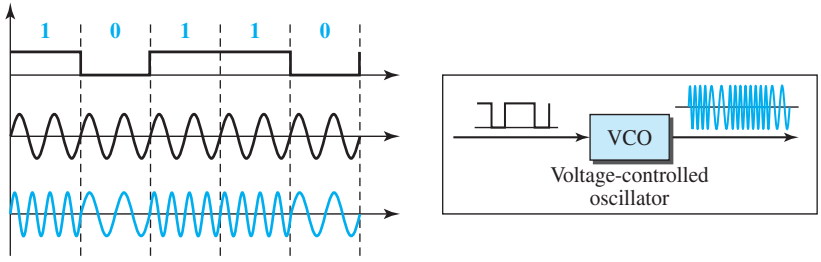
There are two implementations of BFSK: noncoherent and coherent. In noncoherent BFSK, there may be discontinuity in the phase when one signal element ends and the next begins. In coherent BFSK, the phase continues through the boundary of two signal elements. Noncoherent BFSK can be implemented by treating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

Multilevel FSK

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

$$B = (1 + d) \times S + (L - 1)2\Delta_f \longrightarrow B = L \times S$$

Figure 5.7 *Implementation of BFSK*



Note that MFSK uses more bandwidth than the other techniques; it should be used when noise is a serious issue.

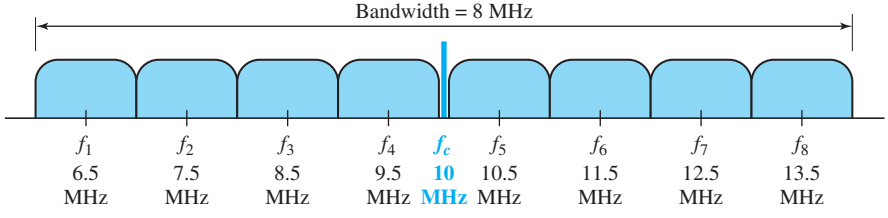
Example 5.6

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

Solution

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

Figure 5.8 *Bandwidth of MFSK used in Example 5.6*



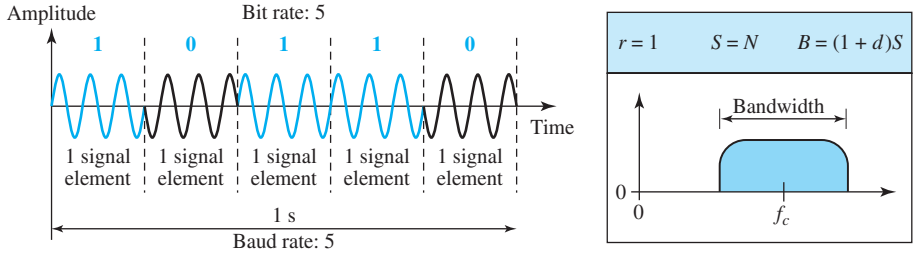
5.1.4 Phase Shift Keying

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

Binary PSK (BPSK)

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

Figure 5.9 *Binary phase shift keying*



signal; in PSK, it is the phase. Noise can change the amplitude easier than it can change the phase. In other words, PSK is less susceptible to noise than ASK. PSK is superior to FSK because we do not need two carrier signals. However, PSK needs more sophisticated hardware to be able to distinguish between phases.

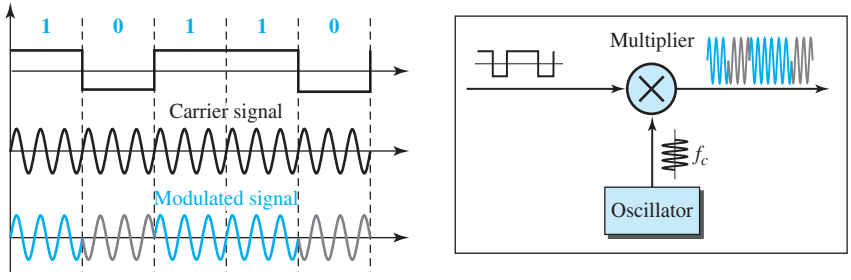
Bandwidth

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

Implementation

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

Figure 5.10 *Implementation of BASK*

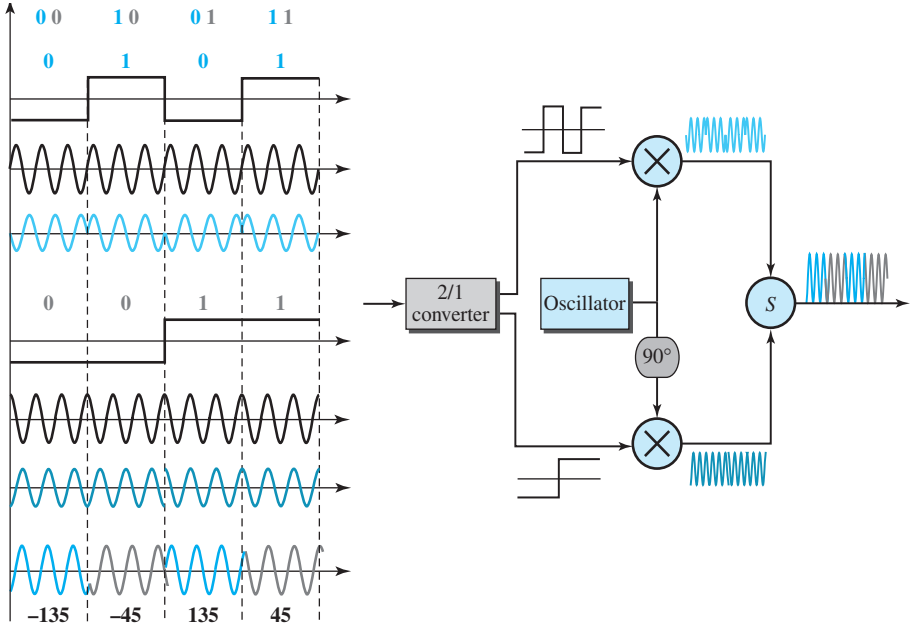


Quadrature PSK (QPSK)

The simplicity of BPSK enticed designers to use 2 bits at a time in each signal element, thereby decreasing the baud rate and eventually the required bandwidth. The scheme is

called *quadrature PSK* or *QPSK* because it uses two separate BPSK modulations; one is in-phase, the other quadrature (out-of-phase). The incoming bits are first passed through a serial-to-parallel conversion that sends one bit to one modulator and the next bit to the other modulator. If the duration of each bit in the incoming signal is T, the duration of each bit sent to the corresponding BPSK signal is $2T$. This means that the bit to each BPSK signal has one-half the frequency of the original signal. Figure 5.11 shows the idea.

Figure 5.11 *QPSK and its implementation*



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

Example 5.7

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

Solution

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

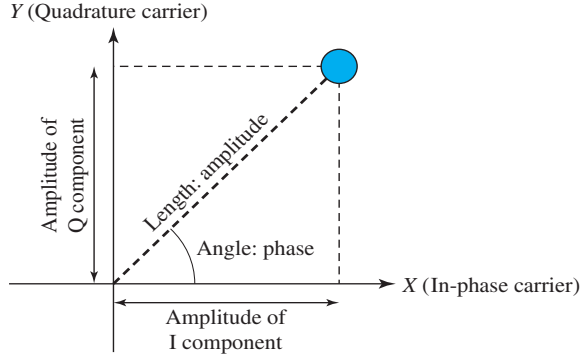
Constellation Diagram

A **constellation diagram** can help us define the amplitude and phase of a signal element, particularly when we are using two carriers (one in-phase and one quadrature). The

diagram is useful when we are dealing with multilevel ASK, PSK, or QAM (see next section). In a constellation diagram, a signal element type is represented as a dot. The bit or combination of bits it can carry is often written next to it.

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

Figure 5.12 *Concept of a constellation diagram*



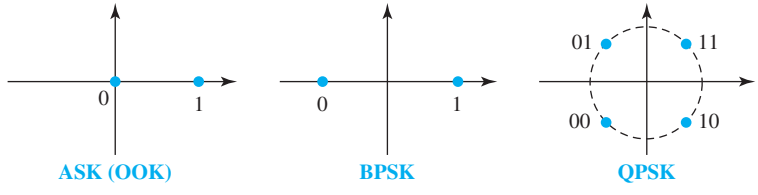
Example 5.8

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

Solution

Figure 5.13 shows the three constellation diagrams. Let us analyze each case separately:

Figure 5.13 *Three constellation diagrams*



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

- BPSK also uses only an in-phase carrier. However, we use a polar NRZ signal for modulation. It creates two types of signal elements, one with amplitude 1 and the other with amplitude −1. This can be stated in other words: BPSK creates two different signal elements, one with amplitude 1 V and in phase and the other with amplitude 1 V and 180° out of phase.
- QPSK uses two carriers, one in-phase and the other quadrature. The point representing 11 is made of two combined signal elements, both with an amplitude of 1 V. One element is represented by an in-phase carrier, the other element by a quadrature carrier. The amplitude of the final signal element sent for this 2-bit data element is $2^{1/2}$, and the phase is 45°. The argument is similar for the other three points. All signal elements have an amplitude of $2^{1/2}$, but their phases are different (45°, 135°, −135°, and −45°). Of course, we could have chosen the amplitude of the carrier to be $1/(2^{1/2})$ to make the final amplitudes 1 V.

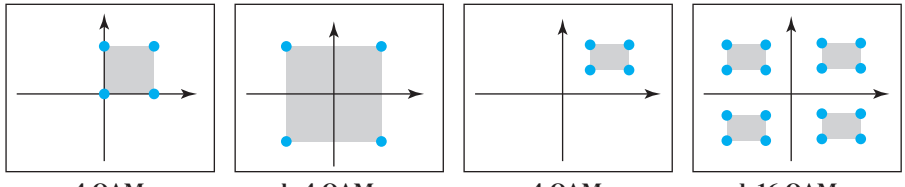
5.1.5 Quadrature Amplitude Modulation

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

Quadrature amplitude modulation is a combination of ASK and PSK.

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

Figure 5.14 *Constellation diagrams for some QAMs*



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

Bandwidth for QAM

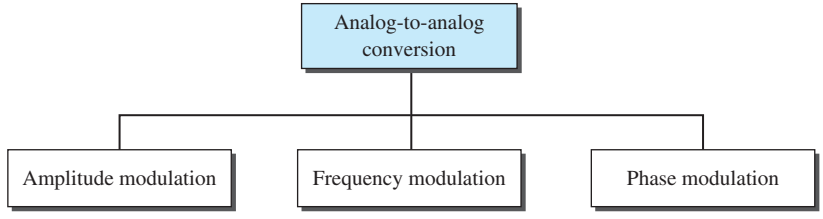
The minimum bandwidth required for QAM transmission is the same as that required for ASK and PSK transmission. QAM has the same advantages as PSK over ASK.

5.2 ANALOG-TO-ANALOG CONVERSION

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

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

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



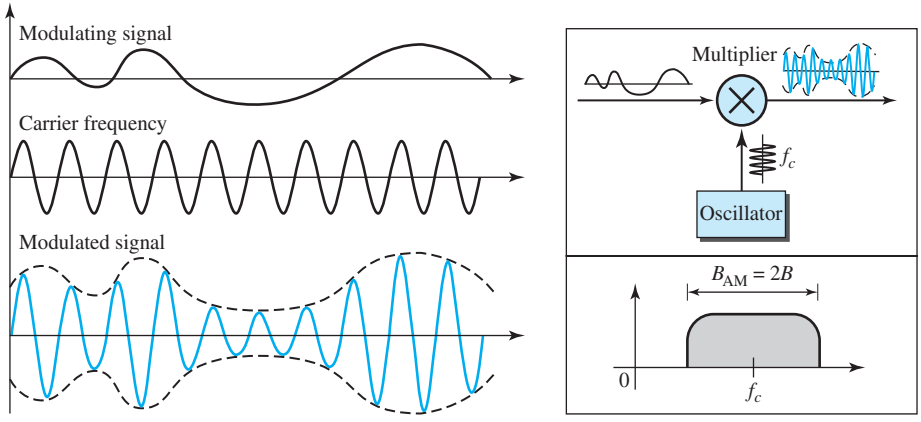
5.2.1 Amplitude Modulation (AM)

In AM transmission, the carrier signal is modulated so that its amplitude varies with the changing amplitudes of the modulating signal. The frequency and phase of the carrier remain the same; only the amplitude changes to follow variations in the information. Figure 5.16 shows how this concept works. The modulating signal is the envelope of the carrier. As Figure 5.16 shows, AM is normally implemented by using a simple multiplier because the amplitude of the carrier signal needs to be changed according to the amplitude of the modulating signal.

AM Bandwidth

Figure 5.16 also shows the bandwidth of an AM signal. The modulation creates a bandwidth that is twice the bandwidth of the modulating signal and covers a range centered on the carrier frequency. However, the signal components above and below the carrier frequency carry exactly the same information. For this reason, some implementations discard one-half of the signals and cut the bandwidth in half.

Figure 5.16 *Amplitude modulation*



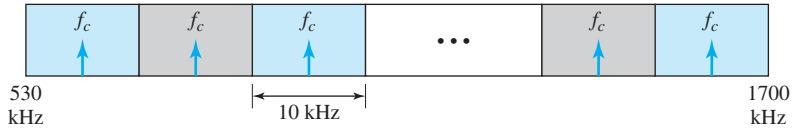
The total bandwidth required for AM can be determined from the bandwidth of the audio signal: $B_{AM} = 2B$.

Standard Bandwidth Allocation for AM Radio

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

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

Figure 5.17 *AM band allocation*



5.2.2 Frequency Modulation (FM)

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

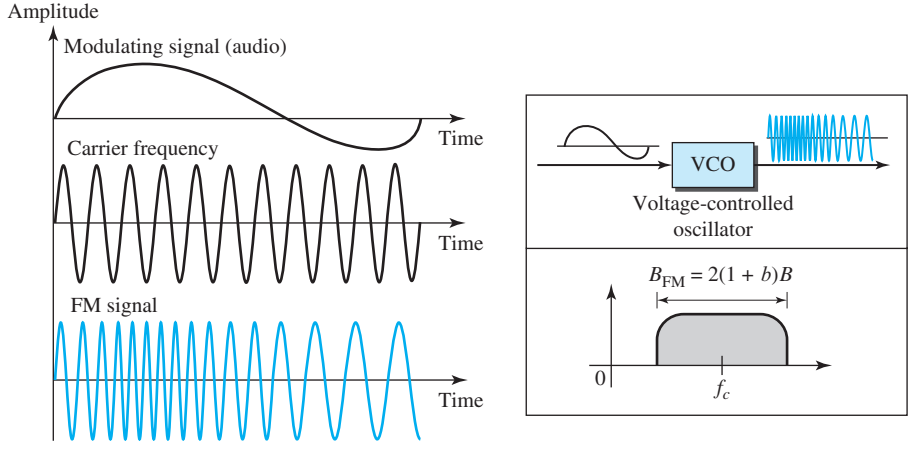
As Figure 5.18 shows, FM is normally implemented by using a voltage-controlled oscillator as with FSK. The frequency of the oscillator changes according to the input voltage which is the amplitude of the modulating signal.

FM Bandwidth

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

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

Figure 5.18 *Frequency modulation*



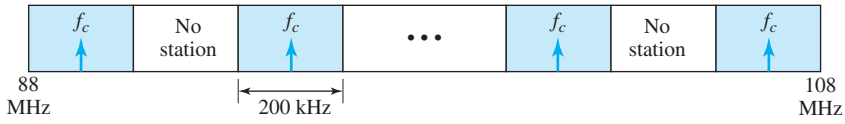
Standard Bandwidth Allocation for FM Radio

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

5.2.3 Phase Modulation (PM)

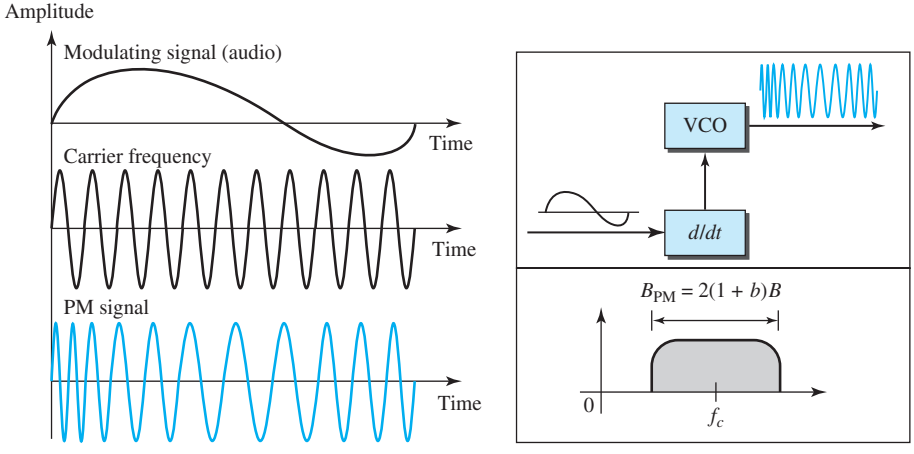
In PM transmission, the phase of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal. The peak amplitude and frequency

Figure 5.19 *FM band allocation*



of the carrier signal remain constant, but as the amplitude of the information signal changes, the phase of the carrier changes correspondingly. It can be proved mathematically (see Appendix E) that PM is the same as FM with one difference. In FM, the instantaneous change in the carrier frequency is proportional to the amplitude of the modulating signal; in PM the instantaneous change in the carrier frequency is proportional to the derivative of the amplitude of the modulating signal. Figure 5.20 shows the relationships of the modulating signal, the carrier signal, and the resultant PM signal.

Figure 5.20 *Phase modulation*



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

PM Bandwidth

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

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

5.3 END-CHAPTER MATERIALS

5.3.1 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

Digital-to-analog conversion is discussed in [Pea92], [Cou01], and [Sta04]. Analog-to-analog conversion is discussed in [Pea92], Chapter 5 of [Cou01], [Sta04]. [Hsu03] gives a good mathematical approach to all materials discussed in this chapter. More advanced materials can be found in [Ber96].

5.3.2 Key Terms

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

5.3.3 Summary

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

Analog-to-analog conversion can be accomplished in three ways: amplitude modulation (AM), frequency modulation (FM), and phase modulation (PM). In AM transmission, the carrier signal is modulated so that its amplitude varies with the changing amplitudes of the modulating signal. The frequency and phase of the carrier remain the same; only the amplitude changes to follow variations in the information. In FM transmission, the frequency of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal. The peak amplitude and phase of the carrier signal remain constant, but as the amplitude of the information signal changes, the frequency of the carrier changes correspondingly. In PM transmission, the phase of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal. The peak amplitude and frequency of the carrier signal remain constant, but as the amplitude of the information signal changes, the phase of the carrier changes correspondingly.

5.4 PRACTICE SET

5.4.1 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.

5.4.2 Questions

Q5-1. Define *analog transmission*.

Q5-2. Define *carrier signal* and explain its role in analog transmission.

Q5-3. Define *digital-to-analog conversion*.

Q5-4. Which characteristics of an analog signal are changed to represent the digital signal in each of the following digital-to-analog conversions?

a. ASK **b.** FSK **c.** PSK **d.** QAM

Q5-5. Which of the four digital-to-analog conversion techniques (ASK, FSK, PSK or QAM) is the most susceptible to noise? Defend your answer.

Q5-6. Define *constellation diagram* and explain its role in analog transmission.

Q5-7. What are the two components of a signal when the signal is represented on a constellation diagram? Which component is shown on the horizontal axis? Which is shown on the vertical axis?

Q5-8. Define *analog-to-analog conversion*.

Q5-9. Which characteristics of an analog signal are changed to represent the lowpass analog signal in each of the following analog-to-analog conversions?

a. AM **b.** FM **c.** PM

Q5-10. Which of the three analog-to-analog conversion techniques (AM, FM, or PM) is the most susceptible to noise? Defend your answer.

5.4.3 Problems

P5-1. Calculate the baud rate for the given bit rate and type of modulation.

a. 2000 bps, FSK

b. 4000 bps, ASK

c. 6000 bps, QPSK

d. 36,000 bps, 64-QAM

P5-2. Calculate the bit rate for the given baud rate and type of modulation.

a. 1000 baud, FSK

b. 1000 baud, ASK

c. 1000 baud, BPSK

d. 1000 baud, 16-QAM

P5-3. What is the number of bits per baud for the following techniques?

a. ASK with four different amplitudes

b. FSK with eight different frequencies

c. PSK with four different phases

d. QAM with a constellation of 128 points

P5-4. Draw the constellation diagram for the following:

a. ASK, with peak amplitude values of 1 and 3

b. BPSK, with a peak amplitude value of 2

c. QPSK, with a peak amplitude value of 3

d. 8-QAM with two different peak amplitude values, 1 and 3, and four different phases

P5-5. Draw the constellation diagram for the following cases. Find the peak amplitude value for each case and define the type of modulation (ASK, FSK, PSK, or QAM).The numbers in parentheses define the values of I and Q respectively.

a. Two points at (2, 0) and (3, 0)

b. Two points at (3, 0) and (–3, 0)

c. Four points at (2, 2), (–2, 2), (–2, –2), and (2, –2)

d. Two points at (0, 2) and (0, –2)

P5-6. How many bits per baud can we send in each of the following cases if the signal constellation has one of the following number of points?

a. 2 **b.** 4 **c.** 16 **d.** 1024

P5-7. What is the required bandwidth for the following cases if we need to send 4000 bps? Let d = 1.

a. ASK

b. FSK with $2\Delta f = 4$ KHz

c. QPSK

d. 16-QAM

P5-8. The telephone line has 4 KHz bandwidth. What is the maximum number of bits we can send using each of the following techniques? Let d = 0.

a. ASK **b.** QPSK **c.** 16-QAM **d.** 64-QAM

P5-9. A corporation has a medium with a 1-MHz bandwidth (lowpass). The corporation needs to create 10 separate independent channels each capable of sending at least 10 Mbps. The company has decided to use QAM technology. What is the minimum number of bits per baud for each channel? What is the number of points in the constellation diagram for each channel? Let d = 0.

P5-10. A cable company uses one of the cable TV channels (with a bandwidth of 6 MHz) to provide digital communication for each resident. What is the available data rate for each resident if the company uses a 64-QAM technique?

P5-11. Find the bandwidth for the following situations if we need to modulate a 5-KHz voice.

a. AM **b.** FM ($\beta = 5$) **c.** PM ($\beta = 1$)

P5-12. Find the total number of channels in the corresponding band allocated by FCC.

a. AM **b.** FM

5.5 SIMULATION EXPERIMENTS

5.5.1 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.

CHAPTER 6

Bandwidth Utilization: Multiplexing and Spectrum Spreading

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

This chapter is divided into two sections:

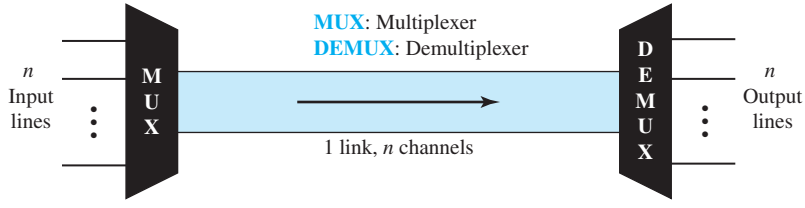
- The first section discusses multiplexing. The first method described in this section is called *frequency-division multiplexing* (FDM), which means to combine several analog signals into a single analog signal. The second method is called *wavelength-division multiplexing* (WDM), which means to combine several optical signals into one optical signal.The third method is called *time-division multiplexing* (TDM), which allows several digital signals to share a channel in time.
- The second section discusses spectrum spreading, in which we first spread the bandwidth of a signal to add redundancy for the purpose of more secure transmission before combining different channels. The first method described in this section is called *frequency hopping spread spectrum* (FHSS), in which different modulation frequencies are used in different periods of time. The second method is called *direct sequence spread spectrum* (DSSS), in which a single bit in the original signal is changed to a sequence before transmission.

6.1 MULTIPLEXING

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

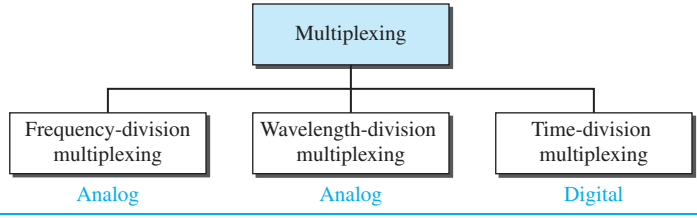
In a multiplexed system, *n* lines share the bandwidth of one link. Figure 6.1 shows the basic format of a multiplexed system. The lines on the left direct their transmission streams to a **multiplexer (MUX),** which combines them into a single stream (many-to-one). At the receiving end, that stream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

Figure 6.1 *Dividing a link into channels*



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

Figure 6.2 *Categories of multiplexing*



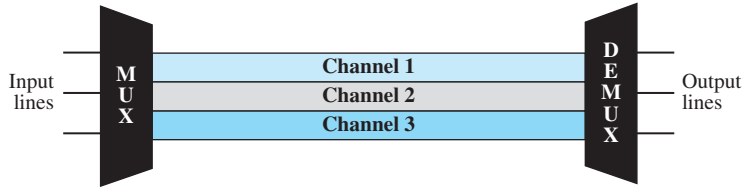
Although some textbooks consider carrier division multiple access (CDMA) as a fourth multiplexing category, we discuss CDMA as an access method (see Chapter 12).

6.1.1 Frequency-Division Multiplexing

Frequency-division multiplexing (FDM) is an analog technique that can be applied when the bandwidth of a link (in hertz) is greater than the combined bandwidths of the signals to be transmitted. In FDM, signals generated by each sending device modulate different carrier frequencies. These modulated signals are then combined into a single composite signal that can be transported by the link. Carrier frequencies are separated by sufficient bandwidth to accommodate the modulated signal. These bandwidth ranges are the channels through which the various signals travel. Channels can be separated by strips of unused bandwidth—**guard bands**—to prevent signals from overlapping. In addition, carrier frequencies must not interfere with the original data frequencies.

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

Figure 6.3 *Frequency-division multiplexing*



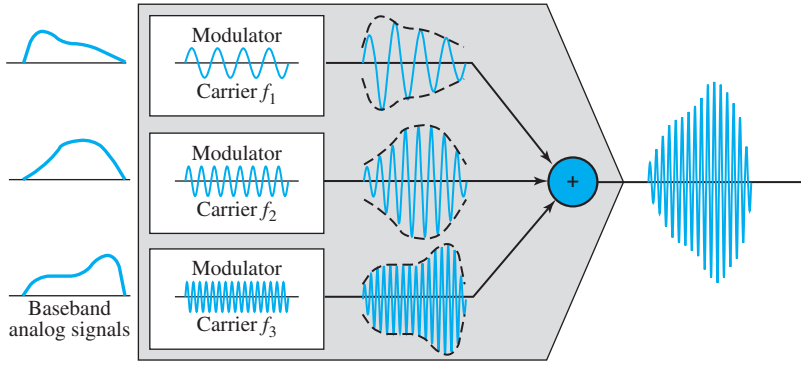
We consider FDM to be an analog multiplexing technique; however, this does not mean that FDM cannot be used to combine sources sending digital signals. A digital signal can be converted to an analog signal (with the techniques discussed in Chapter 5) before FDM is used to multiplex them.

FDM is an analog multiplexing technique that combines analog signals.

Multiplexing Process

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

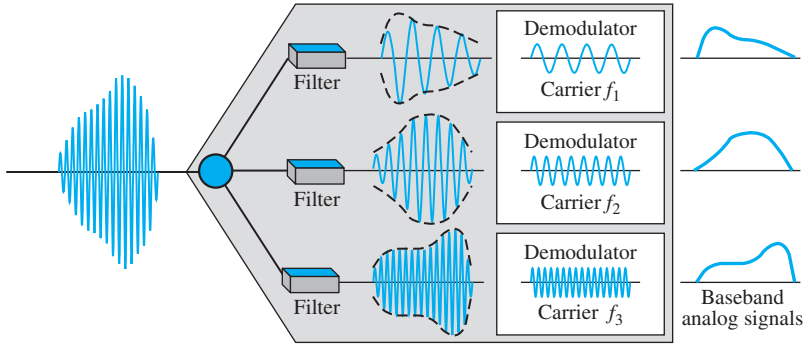
Figure 6.4 *FDM process*



Demultiplexing Process

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

Figure 6.5 *FDM demultiplexing example*



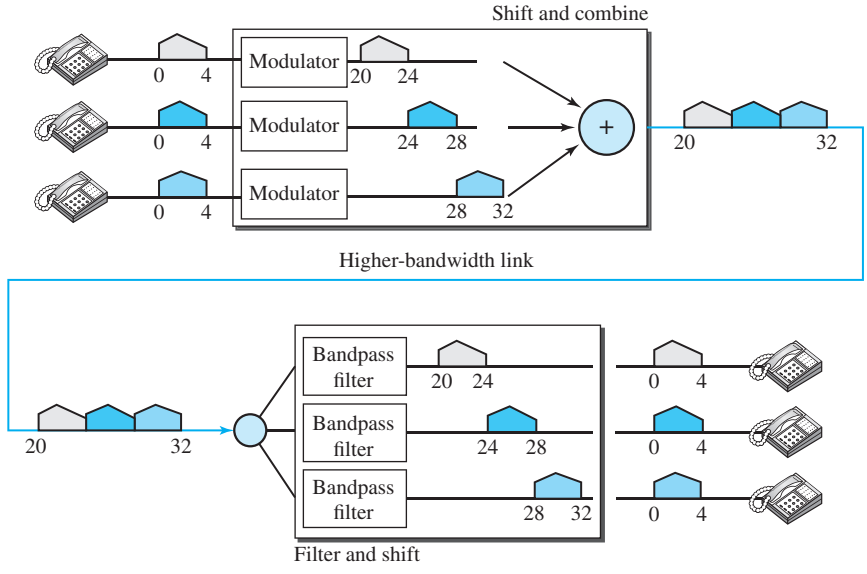
Example 6.1

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

Solution

We shift (modulate) each of the three voice channels to a different bandwidth, as shown in Figure 6.6. We use the 20- to 24-kHz bandwidth for the first channel, the 24- to 28-kHz bandwidth

Figure 6.6 *Example 6.1*



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

Example 6.2

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

Solution

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

Example 6.3

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

Solution

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

Figure 6.7 *Example 6.2*

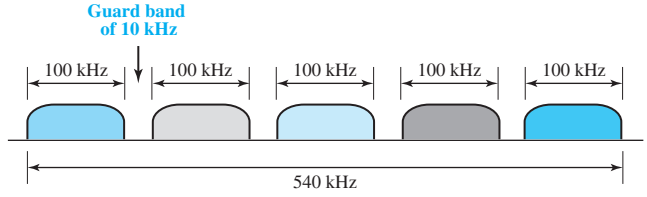
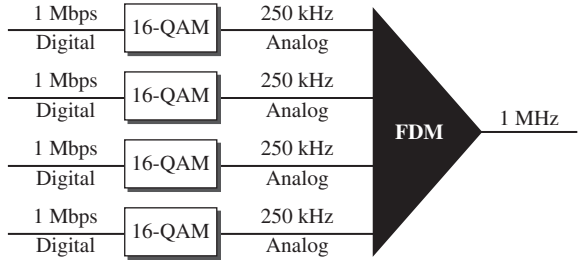


Figure 6.8 *Example 6.3*



The Analog Carrier System

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

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

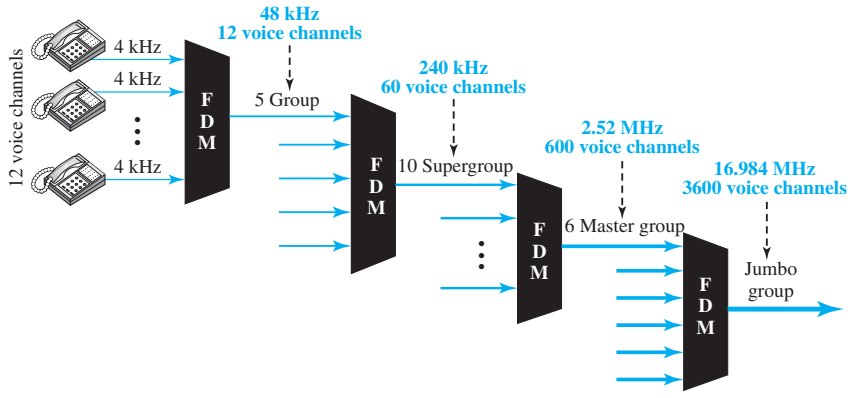
In this **analog hierarchy,** 12 voice channels are multiplexed onto a higher-bandwidth line to create a **group.** A group has 48 kHz of bandwidth and supports 12 voice channels.

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

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

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

Figure 6.9 *Analog hierarchy*



Other Applications of FDM

A very common application of FDM is AM and FM radio broadcasting. Radio uses the air as the transmission medium. A special band from 530 to 1700 kHz is assigned to AM radio. All radio stations need to share this band. As discussed in Chapter 5, each AM station needs 10 kHz of bandwidth. Each station uses a different carrier frequency, which means it is shifting its signal and multiplexing. The signal that goes to the air is a combination of signals. A receiver receives all these signals, but filters (by tuning) only the one which is desired. Without multiplexing, only one AM station could broadcast to the common link, the air. However, we need to know that there is no physical multiplexer or demultiplexer here. As we will see in Chapter 12, multiplexing is done at the data-link layer.

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

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

The first generation of cellular telephones (See Chapter 16) also uses FDM. Each user is assigned two 30-kHz channels, one for sending voice and the other for receiving. The voice signal, which has a bandwidth of 3 kHz (from 300 to 3300 Hz), is modulated by using FM. Remember that an FM signal has a bandwidth 10 times that of the modulating signal, which means each channel has 30 kHz (10×3) of bandwidth. Therefore, each user is given, by the base station, a 60-kHz bandwidth in a range available at the time of the call.

Example 6.4

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

Solution

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

Implementation

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

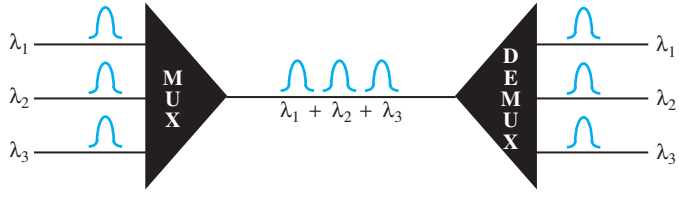
6.1.2 Wavelength-Division Multiplexing

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

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

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

Figure 6.10 *Wavelength-division multiplexing*

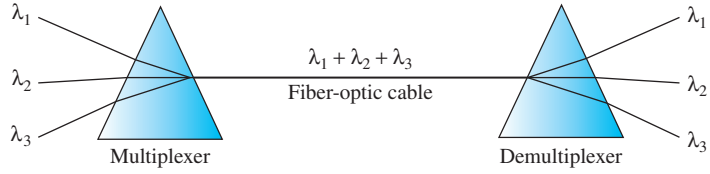


WDM is an analog multiplexing technique to combine optical signals.

Although WDM technology is very complex, the basic idea is very simple. We want to combine multiple light sources into one single light at the multiplexer and do the reverse at the demultiplexer. The combining and splitting of light sources are easily handled by a prism. Recall from basic physics that a prism bends a beam of light based on the angle of incidence and the frequency. Using this technique, a multiplexer can be

made to combine several input beams of light, each containing a narrow band of frequencies, into one output beam of a wider band of frequencies. A demultiplexer can also be made to reverse the process. Figure 6.11 shows the concept.

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



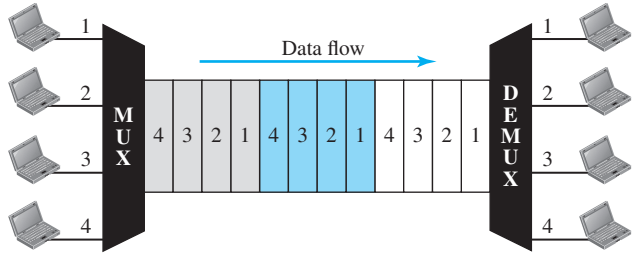
One application of WDM is the SONET network, in which multiple optical fiber lines are multiplexed and demultiplexed. We discuss SONET in Chapter 14.

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

6.1.3 Time-Division Multiplexing

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

Figure 6.12 *TDM*



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

We also need to remember that TDM is, in principle, a digital multiplexing technique. Digital data from different sources are combined into one timeshared link. However, this

does not mean that the sources cannot produce analog data; analog data can be sampled, changed to digital data, and then multiplexed by using TDM.

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

We can divide TDM into two different schemes: synchronous and statistical. We first discuss **synchronous TDM** and then show how **statistical TDM** differs.

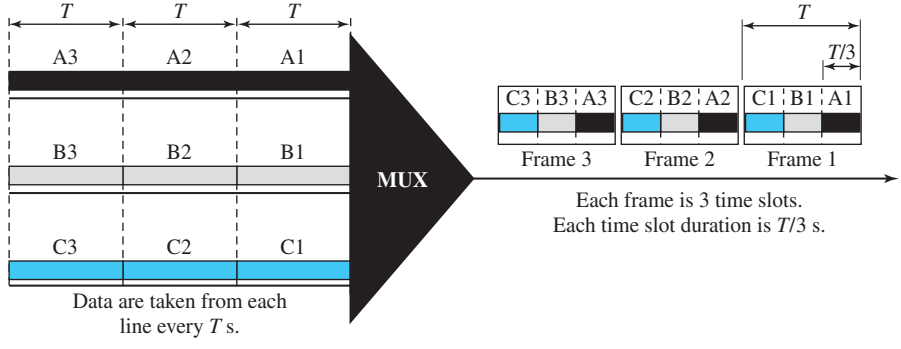
Synchronous TDM

In synchronous TDM, each input connection has an allotment in the output even if it is not sending data.

Time Slots and Frames

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

Figure 6.13 *Synchronous time-division multiplexing*



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

The data rate of the output link must be n times the data rate of a connection to guarantee the flow of data. In Figure 6.13, the data rate of the link is 3 times the data rate of a connection; likewise, the duration of a unit on a connection is 3 times that of

the time slot (duration of a unit on the link). In the figure we represent the data prior to multiplexing as 3 times the size of the data after multiplexing. This is just to convey the idea that each unit is 3 times longer in duration before multiplexing than after.

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

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

Example 6.5

In Figure 6.13, the data rate for each input connection is 1 kbps. If 1 bit at a time is multiplexed (a unit is 1 bit), what is the duration of

1. each input slot,
2. each output slot, and
3. each frame?

Solution

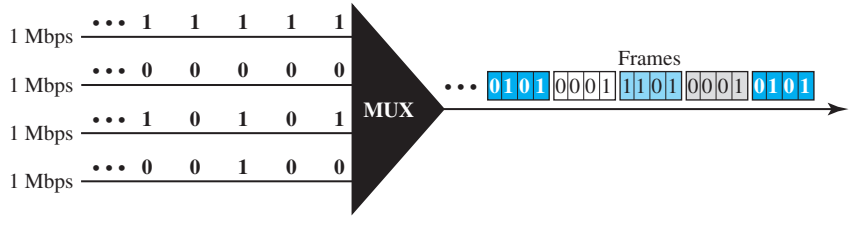
We can answer the questions as follows:

1. The data rate of each input connection is 1 kbps. This means that the bit duration is 1/1000 s or 1 ms. The duration of the input time slot is 1 ms (same as bit duration).
2. The duration of each output time slot is one-third of the input time slot. This means that the duration of the output time slot is 1/3 ms.
3. Each frame carries three output time slots. So the duration of a frame is $3 \times 1/3$ ms, or 1 ms. The duration of a frame is the same as the duration of an input unit.

Example 6.6

Figure 6.14 shows synchronous TDM with a data stream for each input and one data stream for the output. The unit of data is 1 bit. Find (1) the input bit duration, (2) the output bit duration, (3) the output bit rate, and (4) the output frame rate.

Figure 6.14 *Example 6.6*



Solution

We can answer the questions as follows:

1. The input bit duration is the inverse of the bit rate: 1/1 Mbps = 1 μs.
2. The output bit duration is one-fourth of the input bit duration, or 1/4 μs.

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

Example 6.7

Four 1-kbps connections are multiplexed together. A unit is 1 bit. Find (1) the duration of 1 bit before multiplexing, (2) the transmission rate of the link, (3) the duration of a time slot, and (4) the duration of a frame.

Solution

We can answer the questions as follows:

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

Interleaving

TDM can be visualized as two fast-rotating switches, one on the multiplexing side and the other on the demultiplexing side. The switches are synchronized and rotate at the same speed, but in opposite directions. On the multiplexing side, as the switch opens in front of a connection, that connection has the opportunity to send a unit onto the path. This process is called **interleaving.** On the demultiplexing side, as the switch opens in front of a connection, that connection has the opportunity to receive a unit from the path.

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

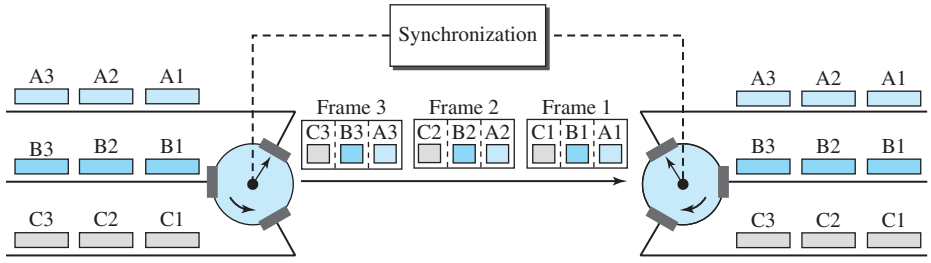
Example 6.8

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

Solution

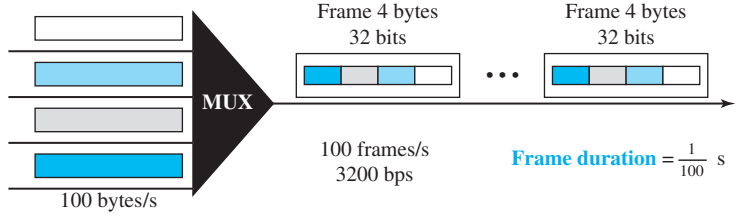
The multiplexer is shown in Figure 6.16. Each frame carries 1 byte from each channel; the size of each frame, therefore, is 4 bytes, or 32 bits. Because each channel is sending 100 bytes/s and a frame carries 1 byte from each channel, the frame rate must be 100 frames per second. The

Figure 6.15 *Interleaving*



duration of a frame is therefore 1/100 s. The link is carrying 100 frames per second, and since each frame contains 32 bits, the bit rate is 100×32, or 3200 bps. This is actually 4 times the bit rate of each channel, which is $100 \times 8 = 800$ bps.

Figure 6.16 *Example 6.8*



Example 6.9

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

Solution

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

Empty Slots

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

The first output frame has three slots filled, the second frame has two slots filled, and the third frame has three slots filled. No frame is full. We learn in the next section

Figure 6.17 *Example 6.9*

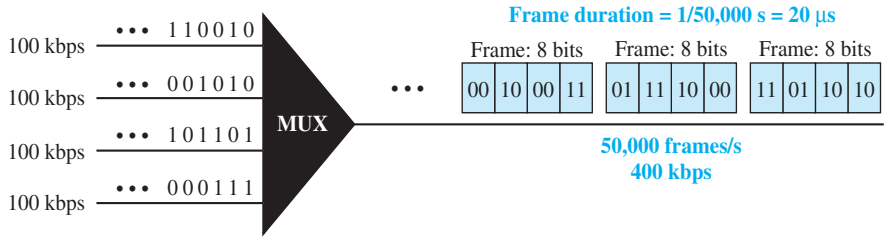
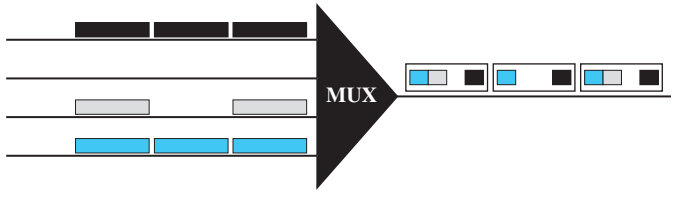


Figure 6.18 *Empty slots*



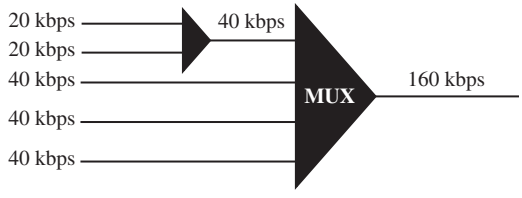
that statistical TDM can improve the efficiency by removing the empty slots from the frame.

Data Rate Management

One problem with TDM is how to handle a disparity in the input data rates. In all our discussion so far, we assumed that the data rates of all input lines were the same. However, if data rates are not the same, three strategies, or a combination of them, can be used. We call these three strategies **multilevel multiplexing, multiple-slot allocation,** and **pulse stuffing.**

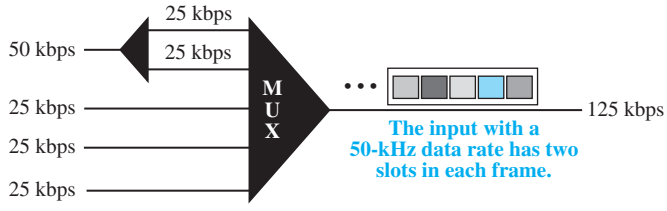
Multilevel Multiplexing Multilevel multiplexing is a technique used when the data rate of an input line is a multiple of others. For example, in Figure 6.19, we have two inputs of 20 kbps and three inputs of 40 kbps. The first two input lines can be multiplexed together to provide a data rate equal to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

Figure 6.19 *Multilevel multiplexing*



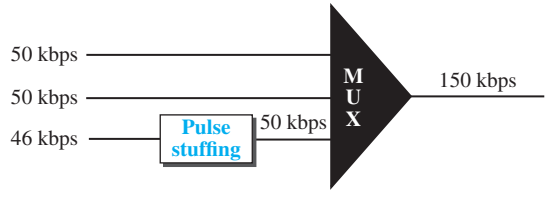
Multiple-Slot Allocation Sometimes it is more efficient to allot more than one slot in a frame to a single input line. For example, we might have an input line that has a data rate that is a multiple of another input. In Figure 6.20, the input line with a 50-kbps data rate can be given two slots in the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

Figure 6.20 *Multiple-slot multiplexing*



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

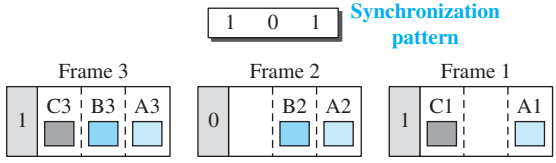
Figure 6.21 *Pulse stuffing*



Frame Synchronizing

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

Figure 6.22 *Framing bits*



Example 6.10

We have four sources, each creating 250 characters per second. If the interleaved unit is a character and 1 synchronizing bit is added to each frame, find (1) the data rate of each source, (2) the duration of each character in each source, (3) the frame rate, (4) the duration of each frame, (5) the number of bits in each frame, and (6) the data rate of the link.

Solution

We can answer the questions as follows:

1. The data rate of each source is $250 \times 8 = 2000$ bps = 2 kbps.
2. Each source sends 250 characters per second; therefore, the duration of a character is 1/250 s, or 4 ms.
3. Each frame has one character from each source, which means the link needs to send 250 frames per second to keep the transmission rate of each source.
4. The duration of each frame is 1/250 s, or 4 ms. Note that the duration of each frame is the same as the duration of each character coming from each source.
5. Each frame carries 4 characters and 1 extra synchronizing bit. This means that each frame is $4 \times 8 + 1 = 33$ bits.
6. The link sends 250 frames per second, and each frame contains 33 bits. This means that the data rate of the link is 250×33, or 8250 bps. Note that the bit rate of the link is greater than the combined bit rates of the four channels. If we add the bit rates of four channels, we get 8000 bps. Because 250 frames are traveling per second and each contains 1 extra bit for synchronizing, we need to add 250 to the sum to get 8250 bps.

Example 6.11

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

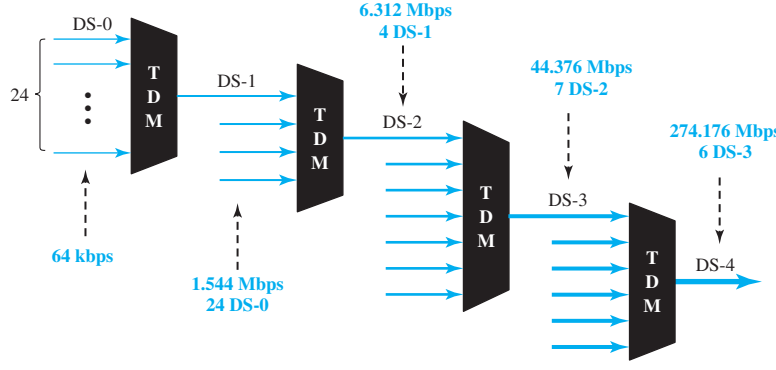
Solution

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

Digital Signal Service

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

Figure 6.23 *Digital hierarchy*



- **DS-0** is a single digital channel of 64 kbps.
- **DS-1** is a 1.544-Mbps service; 1.544 Mbps is 24 times 64 kbps plus 8 kbps of overhead. It can be used as a single service for 1.544-Mbps transmissions, or it can be used to multiplex 24 DS-0 channels or to carry any other combination desired by the user that can fit within its 1.544-Mbps capacity.
- **DS-2** is a 6.312-Mbps service; 6.312 Mbps is 96 times 64 kbps plus 168 kbps of overhead. It can be used as a single service for 6.312-Mbps transmissions; or it can be used to multiplex 4 DS-1 channels, 96 DS-0 channels, or a combination of these service types.
- **DS-3** is a 44.376-Mbps service; 44.376 Mbps is 672 times 64 kbps plus 1.368 Mbps of overhead. It can be used as a single service for 44.376-Mbps transmissions; or it can be used to multiplex 7 DS-2 channels, 28 DS-1 channels, 672 DS-0 channels, or a combination of these service types.
- **DS-4** is a 274.176-Mbps service; 274.176 is 4032 times 64 kbps plus 16.128 Mbps of overhead. It can be used to multiplex 6 DS-3 channels, 42 DS-2 channels, 168 DS-1 channels, 4032 DS-0 channels, or a combination of these service types.

T Lines

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

Table 6.1 *DS and T line rates*

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

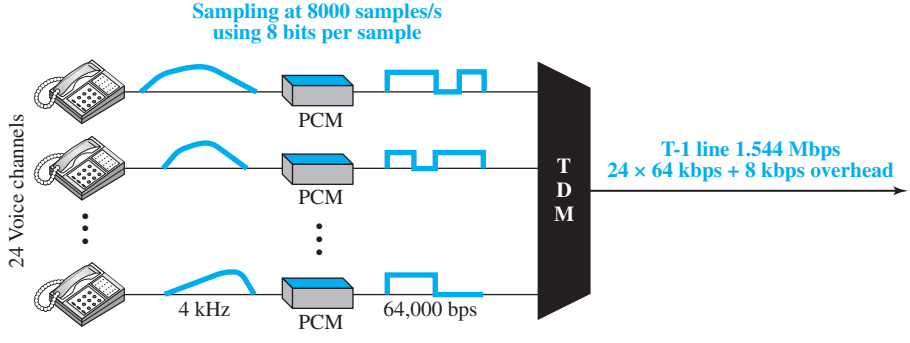
The T-1 line is used to implement DS-1; T-2 is used to implement DS-2; and so on. As you can see from Table 6.1, DS-0 is not actually offered as a service, but it has been defined as a basis for reference purposes.

T Lines for Analog Transmission

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

The possibility of using T lines as analog carriers opened up a new generation of services for the telephone companies. Earlier, when an organization wanted 24 separate telephone lines, it needed to run 24 twisted-pair cables from the company to the central exchange. (Remember those old movies showing a busy executive with 10 telephones lined up on his desk? Or the old office telephones with a big fat cable running from them? Those cables contained a bundle of separate lines.) Today, that same organization can combine the 24 lines into one T-1 line and run only the T-1 line to the exchange. Figure 6.24 shows how 24 voice channels can be multiplexed onto one T-1 line. (Refer to Chapter 4 for PCM encoding.)

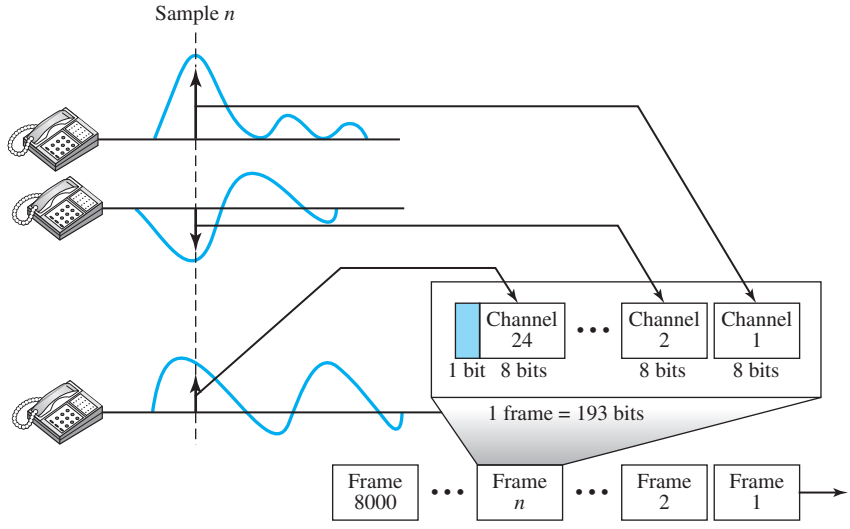
Figure 6.24 *T-1 line for multiplexing telephone lines*



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

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

Figure 6.25 *T-1 frame structure*



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

E Lines

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

Table 6.2 *E line rates*

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

More Synchronous TDM Applications

Some second-generation cellular telephone companies use synchronous TDM. For example, the digital version of cellular telephony divides the available bandwidth into 30-kHz bands. For each band, TDM is applied so that six users can share the band. This means that each 30-kHz band is now made of six time slots, and the digitized voice

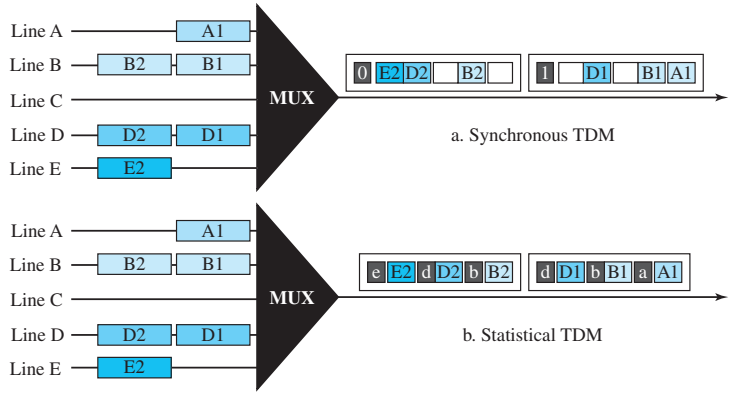
signals of the users are inserted in the slots. Using TDM, the number of telephone users in each area is now 6 times greater. We discuss second-generation cellular telephony in Chapter 16.

Statistical Time-Division Multiplexing

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

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

Figure 6.26 *TDM slot comparison*



Addressing

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

lines with $n = \log_2 N$. For example, for eight different output lines, we need a 3-bit address.

Slot Size

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

No Synchronization Bit

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

Bandwidth

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

6.2 SPREAD SPECTRUM

Multiplexing combines signals from several sources to achieve bandwidth efficiency; the available bandwidth of a link is divided between the sources. In **spread spectrum (SS),** we also combine signals from different sources to fit into a larger bandwidth, but our goals are somewhat different. Spread spectrum is designed to be used in wireless applications (LANs and WANs). In these types of applications, we have some concerns that outweigh bandwidth efficiency. In wireless applications, all stations use air (or a vacuum) as the medium for communication. Stations must be able to share this medium without interception by an eavesdropper and without being subject to jamming from a malicious intruder (in military operations, for example).

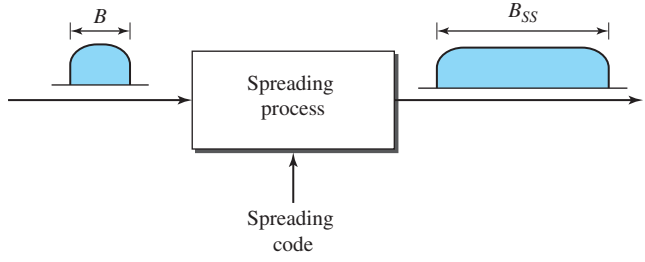
To achieve these goals, spread spectrum techniques add redundancy; they spread the original spectrum needed for each station. If the required bandwidth for each station is B, spread spectrum expands it to B_{ss}, such that $B_{ss} >> B$. The expanded bandwidth allows the source to wrap its message in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

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

1. The bandwidth allocated to each station needs to be, by far, larger than what is needed. This allows redundancy.

2. The expanding of the original bandwidth B to the bandwidth B_{ss} must be done by a process that is independent of the original signal. In other words, the spreading process occurs after the signal is created by the source.

Figure 6.27 *Spread spectrum*



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

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

6.2.1 Frequency Hopping Spread Spectrum

The **frequency hopping spread spectrum (FHSS)** technique uses M different carrier frequencies that are modulated by the source signal. At one moment, the signal modulates one carrier frequency; at the next moment, the signal modulates another carrier frequency. Although the modulation is done using one carrier frequency at a time, M frequencies are used in the long run. The bandwidth occupied by a source after spreading is $B_{FHSS} >> B$.

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

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

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

Figure 6.28 *Frequency hopping spread spectrum (FHSS)*

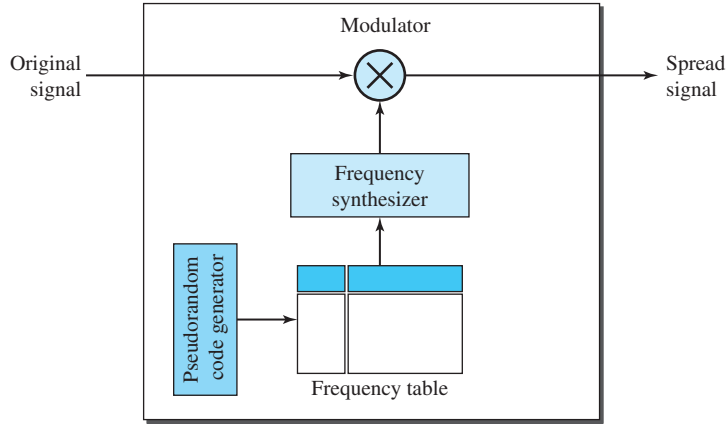
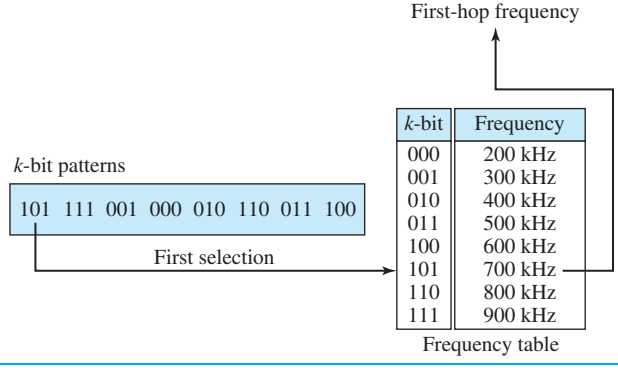


Figure 6.29 *Frequency selection in FHSS*



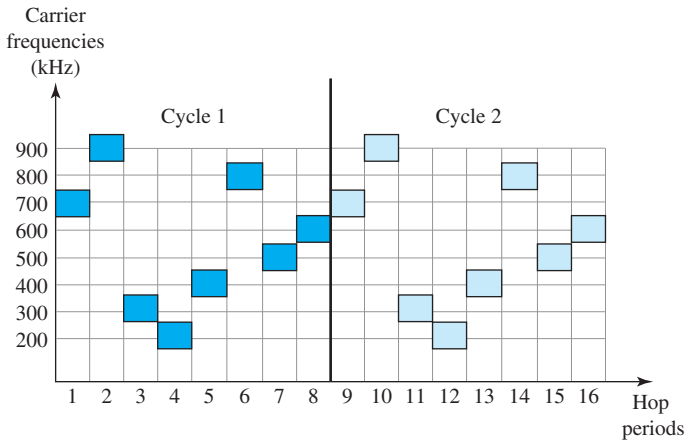
hops around from carrier to carrier. We assume the required bandwidth of the original signal is 100 kHz.

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

Bandwidth Sharing

If the number of hopping frequencies is M, we can multiplex M channels into one by using the same B_{ss} bandwidth. This is possible because a station uses just one frequency in each hopping period; $M - 1$ other frequencies can be used by $M - 1$ other stations. In

Figure 6.30 *FHSS cycles*



other words, M different stations can use the same B_{ss} if an appropriate modulation technique such as multiple FSK (MFSK) is used. FHSS is similar to FDM, as shown in Figure 6.31.

Figure 6.31 *Bandwidth sharing*

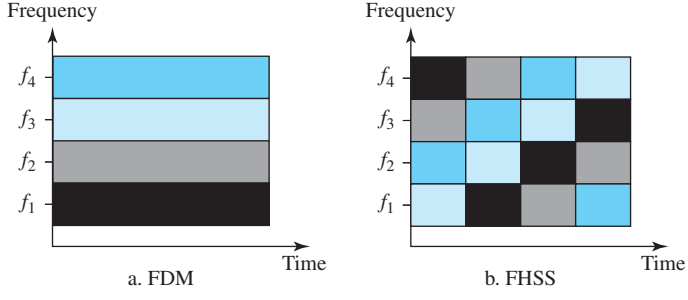
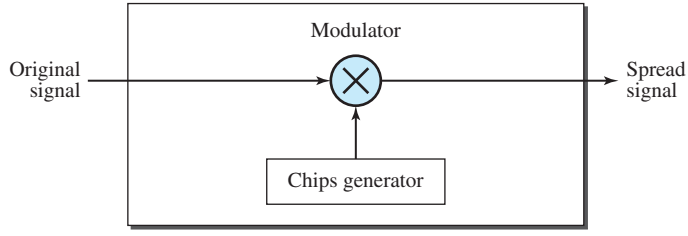


Figure 6.31 shows an example of four channels using FDM and four channels using FHSS. In FDM, each station uses $1/M$ of the bandwidth, but the allocation is fixed; in FHSS, each station uses $1/M$ of the bandwidth, but the allocation changes hop to hop.

6.2.2 Direct Sequence Spread Spectrum

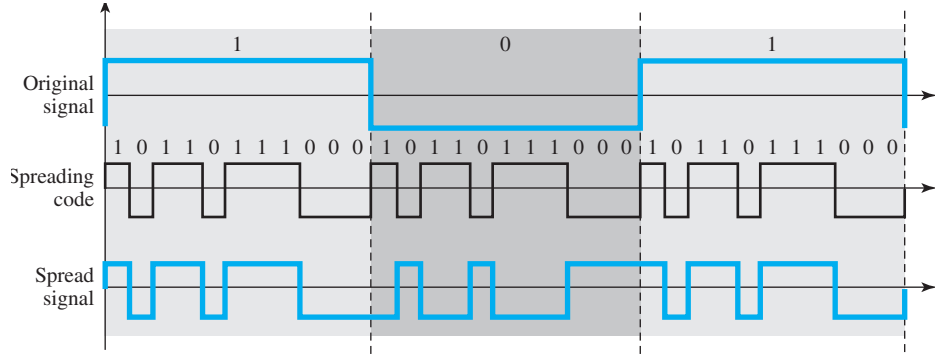
The **direct sequence spread spectrum (DSSS)** technique also expands the bandwidth of the original signal, but the process is different. In DSSS, we replace each data bit with n bits using a spreading code. In other words, each bit is assigned a code of n bits, called ***chips,*** where the chip rate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

Figure 6.32 *DSSS*



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

Figure 6.33 *DSSS example*



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

Bandwidth Sharing

Can we share a bandwidth in DSSS as we did in FHSS? The answer is no and yes. If we use a spreading code that spreads signals (from different stations) that cannot be combined and separated, we cannot share a bandwidth. For example, as we will see in Chapter 15, some wireless LANs use DSSS and the spread bandwidth cannot be shared. However, if we use a special type of sequence code that allows the combining and separating of spread signals, we can share the bandwidth. As we will see in

Chapter 16, a special spreading code allows us to use DSSS in cellular telephony and share a bandwidth among several users.

6.3 END-CHAPTER MATERIALS

6.3.1 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

Multiplexing is discussed in [Pea92]. [Cou01] gives excellent coverage of TDM and FDM. More advanced materials can be found in [Ber96]. Multiplexing is discussed in [Sta04]. A good coverage of spread spectrum can be found in [Cou01] and [Sta04].

6.3.2 Key Terms

analog hierarchy
Barker sequence
channel
chip
demultiplexer (DEMUX)
dense WDM (DWDM)
digital signal (DS) service
direct sequence spread spectrum (DSSS)
E line
framing bit
frequency hopping spread spectrum (FHSS)
frequency-division multiplexing (FDM)
group
guard band
hopping period
interleaving
jumbo group
link
master group
multilevel multiplexing
multiple-slot allocation
multiplexer (MUX)
multiplexing
pseudorandom code generator
pseudorandom noise (PN)
pulse stuffing
spread spectrum (SS)
statistical TDM
supergroup
synchronous TDM
T line
time-division multiplexing (TDM)
wavelength-division multiplexing (WDM)

6.3.3 Summary

Bandwidth utilization is the use of available bandwidth to achieve specific goals. Efficiency can be achieved by using multiplexing; privacy and antijamming can be achieved by using spreading.

Multiplexing is the set of techniques that allow the simultaneous transmission of multiple signals across a single data link. In a multiplexed system, n lines share the bandwidth of one link. The word *link* refers to the physical path. The word *channel* refers to the portion of a link that carries a transmission. There are three basic multiplexing techniques: frequency-division multiplexing, wavelength-division multiplexing, and time-division multiplexing. The first two are techniques designed for analog signals, the third, for digital signals. Frequency-division multiplexing (FDM) is an analog

technique that can be applied when the bandwidth of a link (in hertz) is greater than the combined bandwidths of the signals to be transmitted. Wavelength-division multiplexing (WDM) is designed to use the high bandwidth capability of fiber-optic cable. WDM is an analog multiplexing technique to combine optical signals. Time-division multiplexing (TDM) is a digital process that allows several connections to share the high bandwidth of a link. TDM is a digital multiplexing technique for combining several low-rate channels into one high-rate one. We can divide TDM into two different schemes: synchronous or statistical. In synchronous TDM, each input connection has an allotment in the output even if it is not sending data. In statistical TDM, slots are dynamically allocated to improve bandwidth efficiency.

In spread spectrum (SS), we combine signals from different sources to fit into a larger bandwidth. Spread spectrum is designed to be used in wireless applications in which stations must be able to share the medium without interception by an eavesdropper and without being subject to jamming from a malicious intruder. The frequency hopping spread spectrum (FHSS) technique uses M different carrier frequencies that are modulated by the source signal. At one moment, the signal modulates one carrier frequency; at the next moment, the signal modulates another carrier frequency. The direct sequence spread spectrum (DSSS) technique expands the bandwidth of a signal by replacing each data bit with *n* bits using a spreading code. In other words, each bit is assigned a code of *n* bits, called chips.

6.4 PRACTICE SET

6.4.1 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.

6.4.2 Questions

Q6-1. Describe the goals of multiplexing.

Q6-2. List three main multiplexing techniques mentioned in this chapter.

Q6-3. Distinguish between a link and a channel in multiplexing.

Q6-4. Which of the three multiplexing techniques is (are) used to combine analog signals? Which of the three multiplexing techniques is (are) used to combine digital signals?

Q6-5. Define the analog hierarchy used by telephone companies and list different levels of the hierarchy.

Q6-6. Define the digital hierarchy used by telephone companies and list different levels of the hierarchy.

Q6-7. Which of the three multiplexing techniques is common for fiber-optic links? Explain the reason.

Q6-8. Distinguish between multilevel TDM, multiple-slot TDM, and pulse-stuffed TDM.

Q6-9. Distinguish between synchronous and statistical TDM.

Q6-10. Define spread spectrum and its goal. List the two spread spectrum techniques discussed in this chapter.

Q6-11. Define FHSS and explain how it achieves bandwidth spreading.

Q6-12. Define DSSS and explain how it achieves bandwidth spreading.

6.4.3 Problems

P6-1. Assume that a voice channel occupies a bandwidth of 4 kHz. We need to multiplex 10 voice channels with guard bands of 500 Hz using FDM. Calculate the required bandwidth.

P6-2. We need to transmit 100 digitized voice channels using a passband channel of 20 KHz. What should be the ratio of bits/Hz if we use no guard band?

P6-3. In the analog hierarchy of Figure 6.9, find the overhead (extra bandwidth for guard band or control) in each hierarchy level (group, supergroup, master group, and jumbo group).

P6-4. We need to use synchronous TDM and combine 20 digital sources, each of 100 Kbps. Each output slot carries 1 bit from each digital source, but one extra bit is added to each frame for synchronization. Answer the following questions:

- **a.** What is the size of an output frame in bits?
- **b.** What is the output frame rate?
- **c.** What is the duration of an output frame?
- **d.** What is the output data rate?
- **e.** What is the efficiency of the system (ratio of useful bits to the total bits)?

P6-5. Repeat Problem 6-4 if each output slot carries 2 bits from each source.

P6-6. We have 14 sources, each creating 500 8-bit characters per second. Since only some of these sources are active at any moment, we use statistical TDM to combine these sources using character interleaving. Each frame carries 6 slots at a time, but we need to add 4-bit addresses to each slot. Answer the following questions:

- **a.** What is the size of an output frame in bits?
- **b.** What is the output frame rate?
- **c.** What is the duration of an output frame?
- **d.** What is the output data rate?

P6-7. Ten sources, six with a bit rate of 200 kbps and four with a bit rate of 400 kbps, are to be combined using multilevel TDM with no synchronizing bits. Answer the following questions about the final stage of the multiplexing:

- **a.** What is the size of a frame in bits?
- **b.** What is the frame rate?
- **c.** What is the duration of a frame?
- **d.** What is the data rate?

P6-8. Four channels, two with a bit rate of 200 kbps and two with a bit rate of 150 kbps, are to be multiplexed using multiple-slot TDM with no synchronization bits. Answer the following questions:

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

b. What is the frame rate?

c. What is the duration of a frame?

d. What is the data rate?

P6-9. Two channels, one with a bit rate of 190 kbps and another with a bit rate of 180 kbps, are to be multiplexed using pulse-stuffing TDM with no synchronization bits. Answer the following questions:

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

b. What is the frame rate?

c. What is the duration of a frame?

d. What is the data rate?

P6-10. Answer the following questions about a T-1 line:

a. What is the duration of a frame?

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

P6-11. Show the contents of the five output frames for a synchronous TDM multiplexer that combines four sources sending the following characters. Note that the characters are sent in the same order that they are typed. The third source is silent.

a. Source 1 message: HELLO

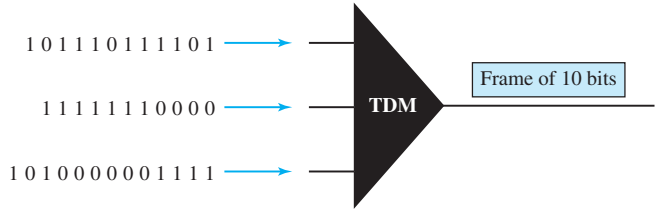
b. Source 2 message: HI

c. Source 3 message:

d. Source 4 message: BYE

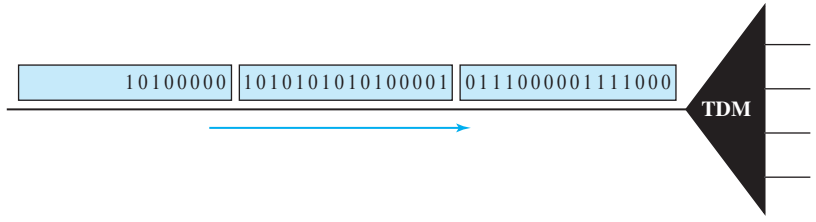
P6-12. Figure 6.34 shows a multiplexer in a synchronous TDM system. Each output slot is only 10 bits long (3 bits taken from each input plus 1 framing bit). What is the output stream? The bits arrive at the multiplexer as shown by the arrows.

Figure 6.34 *Problem P6-12*



P6-13. Figure 6.35 shows a demultiplexer in a synchronous TDM. If the input slot is 16 bits long (no framing bits), what is the bit stream in each output? The bits arrive at the demultiplexer as shown by the arrows.

Figure 6.35 *Problem P6-13*



P6-14. Answer the following questions about the digital hierarchy in Figure 6.23:

a. What is the overhead (number of extra bits) in the DS-1 service?

b. What is the overhead (number of extra bits) in the DS-2 service?

c. What is the overhead (number of extra bits) in the DS-3 service?

d. What is the overhead (number of extra bits) in the DS-4 service?

P6-15. What is the minimum number of bits in a PN sequence if we use FHSS with a channel bandwidth of $B = 4$ KHz and $B_{ss} = 100$ KHz?

P6-16. An FHSS system uses a 4-bit PN sequence. If the bit rate of the PN is 64 bits per second, answer the following questions:

a. What is the total number of possible channels?

b. What is the time needed to finish a complete cycle of PN?

P6-17. A pseudorandom number generator uses the following formula to create a random series:

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

In which N_i defines the current random number and N_{i+1} defines the next random number. The term *mod* means the value of the remainder when dividing $(5 + 7N_i)$ by 17. Show the sequence created by this generator to be used for spread spectrum.

P6-18. We have a digital medium with a data rate of 10 Mbps. How many 64-kbps voice channels can be carried by this medium if we use DSSS with the Barker sequence?

6.5 SIMULATION EXPERIMENTS

6.5.1 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.

CHAPTER 7

Transmission Media

We discussed many issues related to the physical layer in Chapters 3 through 6. In this chapter, we discuss transmission media. We definitely need transmission media to conduct signals from the source to the destination. However, the media can be wired or wireless.

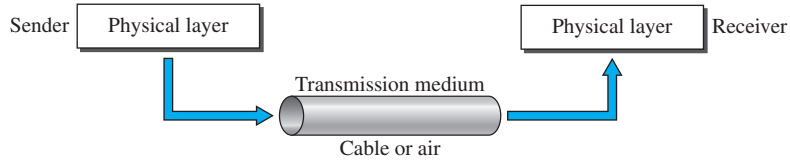
This chapter is divided into three sections:

- The first section introduces the transmission media and defines its position in the Internet model. It shows that we can classify transmission media into two broad categories: guided and unguided media.
- The second section discusses guided media. The first part describes twisted-pair cables and their characteristics and applications. The second part describes coaxial cables and their characteristics and applications. Finally, the third part describes fiber-optic cables and their characteristics and applications.
- The third section discusses unguided media. The first part describes radio waves and their characteristics and applications. The second part describes microwaves and their characteristics and applications. Finally, the third part describes infrared waves and their characteristics and applications.

7.1 INTRODUCTION

Transmission media are actually located below the physical layer and are directly controlled by the physical layer. We could say that transmission media belong to layer zero. Figure 7.1 shows the position of transmission media in relation to the physical layer.

Figure 7.1 *Transmission medium and physical layer*



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

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

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

Extending the range of the human voice became possible when the telephone was invented in 1869. Telephone communication at that time also needed a metallic medium to carry the electric signals that were the result of a conversion from the human voice. The communication was, however, unreliable due to the poor quality of the wires. The lines were often noisy and the technology was unsophisticated.

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

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

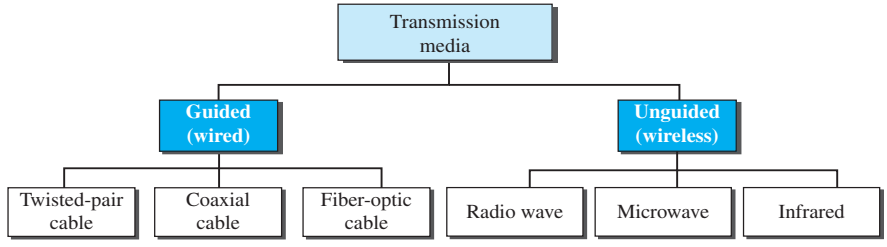
As discussed in Chapter 3, computers and other telecommunication devices use signals to represent data. These signals are transmitted from one device to another in the form of electromagnetic energy, which is propagated through transmission media.

Electromagnetic energy, a combination of electric and magnetic fields vibrating in relation to each other, includes power, radio waves, infrared light, visible light, ultraviolet

light, and X, gamma, and cosmic rays. Each of these constitutes a portion of the **electromagnetic spectrum.** Not all portions of the spectrum are currently usable for telecommunications, however. The media to harness those that are usable are also limited to a few types.

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

Figure 7.2 *Classes of transmission media*



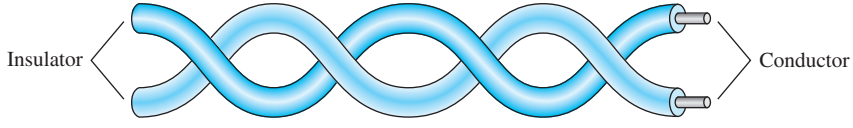
7.2 GUIDED MEDIA

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

7.2.1 Twisted-Pair Cable

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

Figure 7.3 *Twisted-pair cable*



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

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

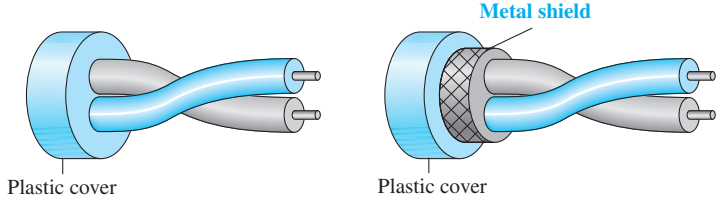
If the two wires are parallel, the effect of these unwanted signals is not the same in both wires because they are at different locations relative to the noise or crosstalk sources (e.g., one is closer and the other is farther). This results in a difference at the receiver.

By twisting the pairs, a balance is maintained. For example, suppose in one twist, one wire is closer to the noise source and the other is farther; in the next twist, the reverse is true. Twisting makes it probable that both wires are equally affected by external influences (noise or crosstalk). This means that the receiver, which calculates the difference between the two, receives no unwanted signals. The unwanted signals are mostly canceled out. From the above discussion, it is clear that the number of twists per unit of length (e.g., inch) has some effect on the quality of the cable.

Unshielded Versus Shielded Twisted-Pair Cable

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

Figure 7.4 *UTP and STP cables*



Categories

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

Table 7.1 *Categories of unshielded twisted-pair cables*

Category	*Specification*	*Data Rate (Mbps)*	*Use*
1	Unshielded twisted-pair used in telephone	< 0.1	Telephone
2	Unshielded twisted-pair originally used in T lines	2	T-1 lines
3	Improved CAT 2 used in LANs	10	LANs
4	Improved CAT 3 used in Token Ring networks	20	LANs
5	Cable wire is normally 24 AWG with a jacket and outside sheath	100	LANs

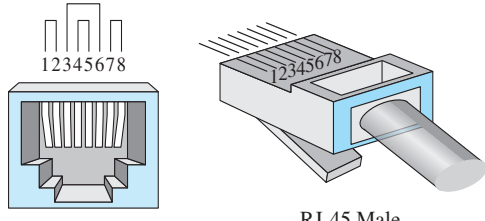
Table 7.1 *Categories of unshielded twisted-pair cables (continued)*

Category	*Specification*	*Data Rate (Mbps)*	*Use*
5E	An extension to category 5 that includes extra features to minimize the crosstalk and electromagnetic interference	125	LANs
6	A new category with matched components coming from the same manufacturer. The cable must be tested at a 200-Mbps data rate.	200	LANs
7	Sometimes called *SSTP (shielded screen twisted-pair)*. Each pair is individually wrapped in a helical metallic foil followed by a metallic foil shield in addition to the outside sheath. The shield decreases the effect of crosstalk and increases the data rate.	600	LANs

Connectors

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

Figure 7.5 *UTP connector*



RJ-45 Female RJ-45 Male

Performance

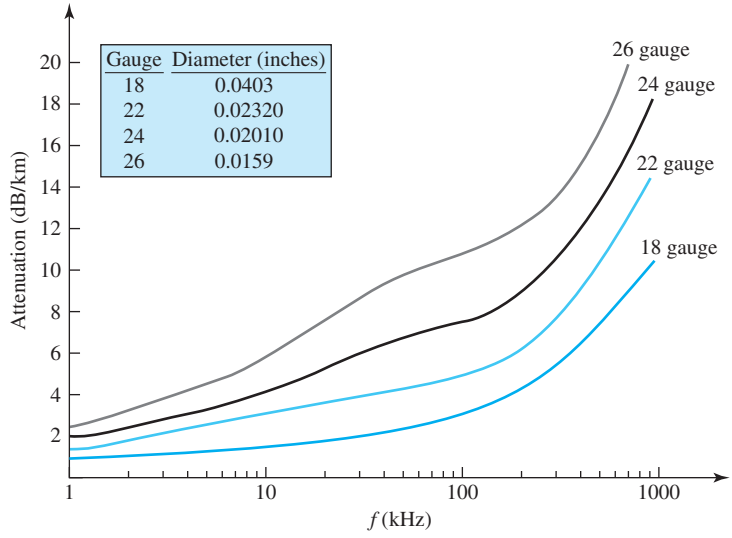
One way to measure the performance of twisted-pair cable is to compare attenuation versus frequency and distance. A twisted-pair cable can pass a wide range of frequencies. However, Figure 7.6 shows that with increasing frequency, the attenuation, measured in decibels per kilometer (dB/km), sharply increases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

Applications

Twisted-pair cables are used in telephone lines to provide voice and data channels. The local loop—the line that connects subscribers to the central telephone office—commonly consists of unshielded twisted-pair cables. We discuss telephone networks in Chapter 14.

The DSL lines that are used by the telephone companies to provide high-data-rate connections also use the high-bandwidth capability of unshielded twisted-pair cables. We discuss DSL technology in Chapter 14.

Figure 7.6 *UTP performance*

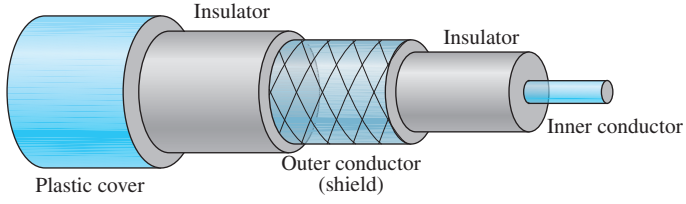


Local-area networks, such as 10Base-T and 100Base-T, also use twisted-pair cables. We discuss these networks in Chapter 13.

7.2.2 Coaxial Cable

Coaxial cable (or *coax*) carries signals of higher frequency ranges than those in twisted-pair cable, in part because the two media are constructed quite differently. Instead of having two wires, coax has a central core conductor of solid or stranded wire (usually copper) enclosed in an insulating sheath, which is, in turn, encased in an outer conductor of metal foil, braid, or a combination of the two. The outer metallic wrapping serves both as a shield against noise and as the second conductor, which completes the circuit. This outer conductor is also enclosed in an insulating sheath, and the whole cable is protected by a plastic cover (see Figure 7.7).

Figure 7.7 *Coaxial cable*



Coaxial Cable Standards

Coaxial cables are categorized by their **Radio Government (RG)** ratings. Each RG number denotes a unique set of physical specifications, including the wire gauge of the

inner conductor, the thickness and type of the inner insulator, the construction of the shield, and the size and type of the outer casing. Each cable defined by an RG rating is adapted for a specialized function, as shown in Table 7.2.

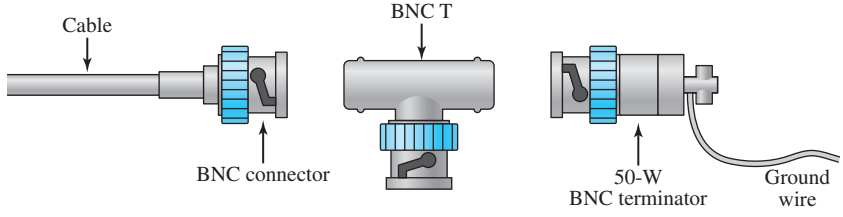
Table 7.2 *Categories of coaxial cables*

Category	*Impedance*	*Use*
RG-59	75 Ω	Cable TV
RG-58	50 Ω	Thin Ethernet
RG-11	50 Ω	Thick Ethernet

Coaxial Cable Connectors

To connect coaxial cable to devices, we need coaxial connectors. The most common type of connector used today is the **Bayonet Neill-Concelman (BNC)** connector. Figure 7.8 shows three popular types of these connectors: the BNC connector, the BNC T connector, and the BNC terminator.

Figure 7.8 *BNC connectors*



The BNC connector is used to connect the end of the cable to a device, such as a TV set. The BNC T connector is used in Ethernet networks (see Chapter 13) to branch out to a connection to a computer or other device. The BNC terminator is used at the end of the cable to prevent the reflection of the signal.

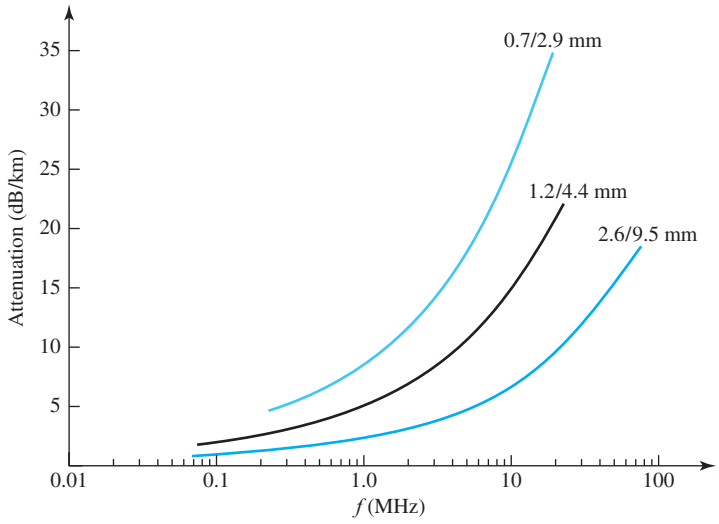
Performance

As we did with twisted-pair cable, we can measure the performance of a coaxial cable. We notice in Figure 7.9 that the attenuation is much higher in coaxial cable than in twisted-pair cable. In other words, although coaxial cable has a much higher bandwidth, the signal weakens rapidly and requires the frequent use of repeaters.

Applications

Coaxial cable was widely used in analog telephone networks where a single coaxial network could carry 10,000 voice signals. Later it was used in digital telephone networks where a single coaxial cable could carry digital data up to 600 Mbps. However, coaxial cable in telephone networks has largely been replaced today with fiber-optic cable.

Figure 7.9 *Coaxial cable performance*



Cable TV networks (see Chapter 14) also use coaxial cables. In the traditional cable TV network, the entire network used coaxial cable. Later, however, cable TV providers replaced most of the media with fiber-optic cable; hybrid networks use coaxial cable only at the network boundaries, near the consumer premises. Cable TV uses RG-59 coaxial cable.

Another common application of coaxial cable is in traditional Ethernet LANs (see Chapter 13). Because of its high bandwidth, and consequently high data rate, coaxial cable was chosen for digital transmission in early Ethernet LANs. The 10Base-2, or Thin Ethernet, uses RG-58 coaxial cable with BNC connectors to transmit data at 10 Mbps with a range of 185 m. The 10Base5, or Thick Ethernet, uses RG-11 (thick coaxial cable) to transmit 10 Mbps with a range of 5000 m. Thick Ethernet has specialized connectors.

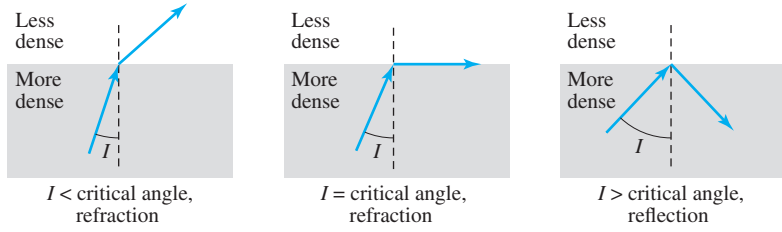
7.2.3 Fiber-Optic Cable

A fiber-optic cable is made of glass or plastic and transmits signals in the form of light. To understand optical fiber, we first need to explore several aspects of the nature of light.

Light travels in a straight line as long as it is moving through a single uniform substance. If a ray of light traveling through one substance suddenly enters another substance (of a different density), the ray changes direction. Figure 7.10 shows how a ray of light changes direction when going from a more dense to a less dense substance.

As the figure shows, if the **angle of incidence** I (the angle the ray makes with the line perpendicular to the interface between the two substances) is less than the **critical angle,** the ray **refracts** and moves closer to the surface. If the angle of incidence is equal to the critical angle, the light bends along the interface. If the angle is greater than the critical angle, the ray **reflects** (makes a turn) and travels again in the denser

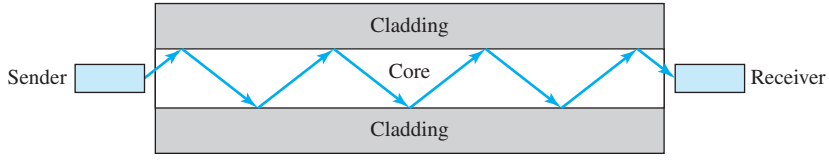
Figure 7.10 *Bending of light ray*



substance. Note that the critical angle is a property of the substance, and its value differs from one substance to another.

Optical fibers use reflection to guide light through a channel. A glass or plastic **core** is surrounded by a **cladding** of less dense glass or plastic. The difference in density of the two materials must be such that a beam of light moving through the core is reflected off the cladding instead of being refracted into it. See Figure 7.11.

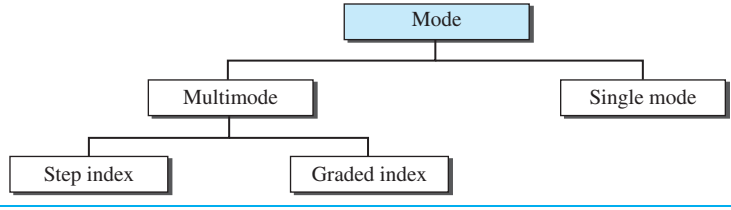
Figure 7.11 *Optical fiber*



Propagation Modes

Current technology supports two modes (multimode and single mode) for propagating light along optical channels, each requiring fiber with different physical characteristics. Multimode can be implemented in two forms: step-index or graded-index (see Figure 7.12).

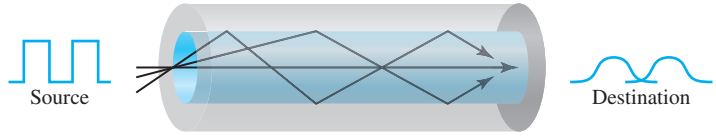
Figure 7.12 *Propagation modes*



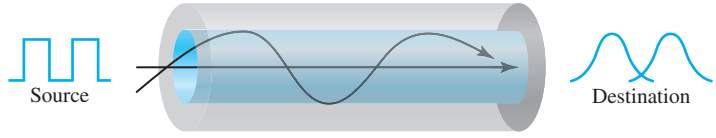
Multimode

Multimode is so named because multiple beams from a light source move through the core in different paths. How these beams move within the cable depends on the structure of the core, as shown in Figure 7.13.

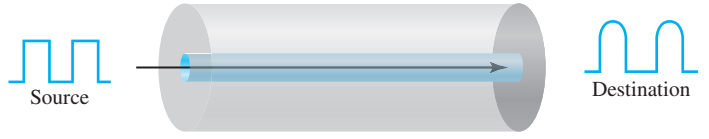
Figure 7.13 *Modes*



a. Multimode, step index



b. Multimode, graded index



c. Single mode

In **multimode step-index fiber,** the density of the core remains constant from the center to the edges. A beam of light moves through this constant density in a straight line until it reaches the interface of the core and the cladding. At the interface, there is an abrupt change due to a lower density; this alters the angle of the beam's motion. The term *step-index* refers to the suddenness of this change, which contributes to the distortion of the signal as it passes through the fiber.

A second type of fiber, called **multimode graded-index fiber,** decreases this distortion of the signal through the cable. The word *index* here refers to the index of refraction. As we saw above, the index of refraction is related to density. A graded-index fiber, therefore, is one with varying densities. Density is highest at the center of the core and decreases gradually to its lowest at the edge. Figure 7.13 shows the impact of this variable density on the propagation of light beams.

Single-Mode

Single-mode uses step-index fiber and a highly focused source of light that limits beams to a small range of angles, all close to the horizontal. The **single-mode fiber** itself is manufactured with a much smaller diameter than that of multimode fiber, and with substantially lower density (index of refraction). The decrease in density results in a critical angle that is close enough to 90° to make the propagation of beams almost horizontal. In this case, propagation of different beams is almost identical, and delays are negligible. All the beams arrive at the destination "together" and can be recombined with little distortion to the signal (see Figure 7.13).

Fiber Sizes

Optical fibers are defined by the ratio of the diameter of their core to the diameter of their cladding, both expressed in micrometers. The common sizes are shown in Table 7.3. Note that the last size listed is for single-mode only.

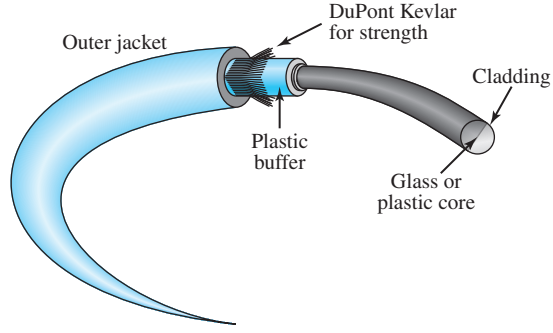
Table 7.3 *Fiber types*

Type	*Core (μm)*	*Cladding (μm)*	*Mode*
50/125	50.0	125	Multimode, graded index
62.5/125	62.5	125	Multimode, graded index
100/125	100.0	125	Multimode, graded index
7/125	7.0	125	Single mode

Cable Composition

Figure 7.14 shows the composition of a typical fiber-optic cable. The outer jacket is

Figure 7.14 *Fiber construction*



made of either PVC or Teflon. Inside the jacket are Kevlar strands to strengthen the cable. Kevlar is a strong material used in the fabrication of bulletproof vests. Below the Kevlar is another plastic coating to cushion the fiber. The fiber is at the center of the cable, and it consists of cladding and core.

Fiber-Optic Cable Connectors

There are three types of connectors for fiber-optic cables, as shown in Figure 7.15. The **subscriber channel (SC) connector** is used for cable TV. It uses a push/pull locking system. The **straight-tip (ST) connector** is used for connecting cable to networking devices. It uses a bayonet locking system and is more reliable than SC. **MT-RJ** is a connector that is the same size as RJ45.

Performance

The plot of attenuation versus wavelength in Figure 7.16 shows a very interesting phenomenon in fiber-optic cable. Attenuation is flatter than in the case of twisted-pair cable and coaxial cable. The performance is such that we need fewer (actually one-tenth as many) repeaters when we use fiber-optic cable.

Figure 7.15 *Fiber-optic cable connectors*

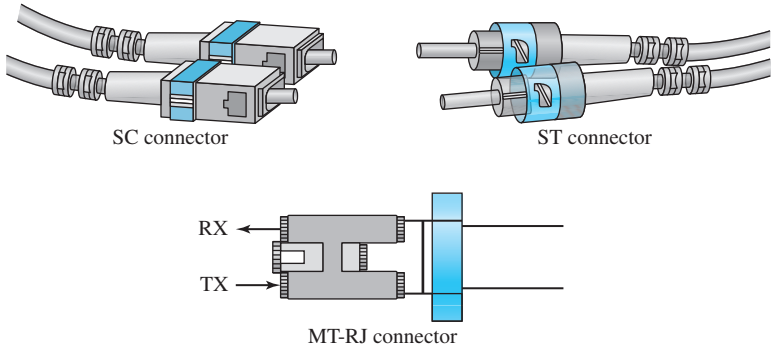
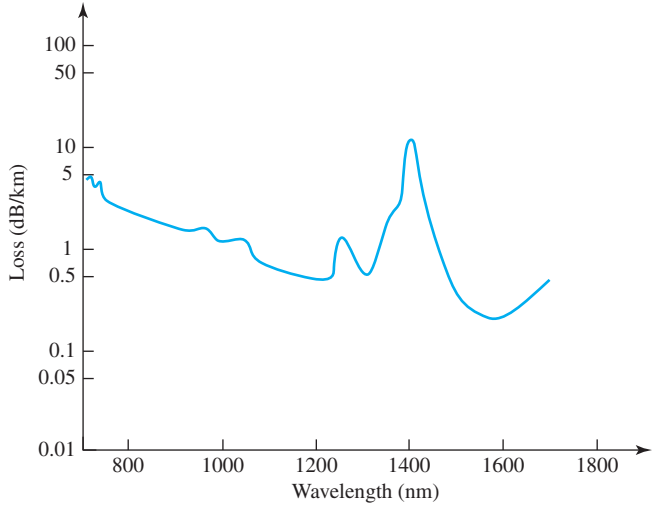


Figure 7.16 *Optical fiber performance*



Applications

Fiber-optic cable is often found in backbone networks because its wide bandwidth is cost-effective. Today, with wavelength-division multiplexing (WDM), we can transfer data at a rate of 1600 Gbps. The SONET network that we discuss in Chapter 14 provides such a backbone.

Some cable TV companies use a combination of optical fiber and coaxial cable, thus creating a hybrid network. Optical fiber provides the backbone structure while coaxial cable provides the connection to the user premises. This is a cost-effective configuration since the narrow bandwidth requirement at the user end does not justify the use of optical fiber.

Local-area networks such as 100Base-FX network (Fast Ethernet) and 1000Base-X also use fiber-optic cable.

Advantages and Disadvantages of Optical Fiber

Advantages

Fiber-optic cable has several advantages over metallic cable (twisted-pair or coaxial).

- **Higher bandwidth.** Fiber-optic cable can support dramatically higher bandwidths (and hence data rates) than either twisted-pair or coaxial cable. Currently, data rates and bandwidth utilization over fiber-optic cable are limited not by the medium but by the signal generation and reception technology available.
- **Less signal attenuation.** Fiber-optic transmission distance is significantly greater than that of other guided media. A signal can run for 50 km without requiring regeneration. We need repeaters every 5 km for coaxial or twisted-pair cable.
- **Immunity to electromagnetic interference.** Electromagnetic noise cannot affect fiber-optic cables.
- **Resistance to corrosive materials.** Glass is more resistant to corrosive materials than copper.
- **Light weight.** Fiber-optic cables are much lighter than copper cables.
- **Greater immunity to tapping.** Fiber-optic cables are more immune to tapping than copper cables. Copper cables create antenna effects that can easily be tapped.

Disadvantages

There are some disadvantages in the use of optical fiber.

- **Installation and maintenance.** Fiber-optic cable is a relatively new technology. Its installation and maintenance require expertise that is not yet available everywhere.
- **Unidirectional light propagation.** Propagation of light is unidirectional. If we need bidirectional communication, two fibers are needed.
- **Cost.** The cable and the interfaces are relatively more expensive than those of other guided media. If the demand for bandwidth is not high, often the use of optical fiber cannot be justified.

7.3 UNGUIDED MEDIA: WIRELESS

Unguided medium transport electromagnetic waves without using a physical conductor. This type of communication is often referred to as ***wireless communication*****.** Signals are normally broadcast through free space and thus are available to anyone who has a device capable of receiving them.

Figure 7.17 shows the part of the electromagnetic spectrum, ranging from 3 kHz to 900 THz, used for wireless communication.

Unguided signals can travel from the source to the destination in several ways: ground propagation, sky propagation, and line-of-sight propagation, as shown in Figure 7.18.

Figure 7.17 *Electromagnetic spectrum for wireless communication*

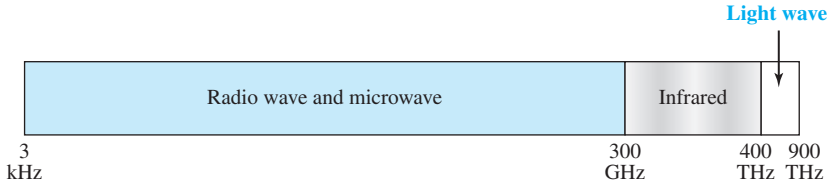
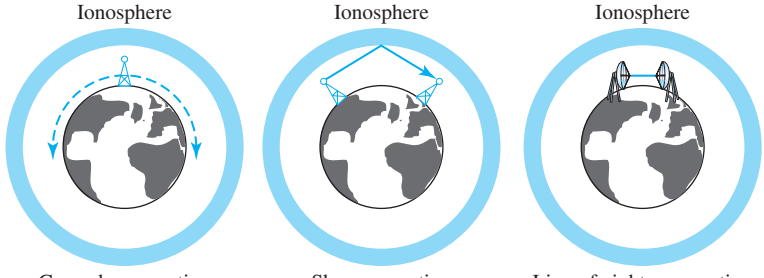


Figure 7.18 *Propagation methods*



In **ground propagation,** radio waves travel through the lowest portion of the atmosphere, hugging the earth. These low-frequency signals emanate in all directions from the transmitting antenna and follow the curvature of the planet. Distance depends on the amount of power in the signal: The greater the power, the greater the distance. In **sky propagation,** higher-frequency radio waves radiate upward into the ionosphere (the layer of atmosphere where particles exist as ions) where they are reflected back to earth. This type of transmission allows for greater distances with lower output power. In **line-of-sight propagation,** very high-frequency signals are transmitted in straight lines directly from antenna to antenna. Antennas must be directional, facing each other, and either tall enough or close enough together not to be affected by the curvature of the earth. Line-of-sight propagation is tricky because radio transmissions cannot be completely focused.

The section of the electromagnetic spectrum defined as radio waves and microwaves is divided into eight ranges, called *bands,* each regulated by government authorities. These bands are rated from *very low frequency* (VLF) to *extremely high frequency* (EHF). Table 7.4 lists these bands, their ranges, propagation methods, and some applications.

Table 7.4 *Bands*

Band	*Range*	*Propagation*	*Application*
very low frequency (VLF)	3–30 kHz	Ground	Long-range radio navigation
low frequency (LF)	30–300 kHz	Ground	Radio beacons and navigational locators

Table 7.4 *Bands (continued)*

Band	*Range*	*Propagation*	*Application*
middle frequency (MF)	300 kHz–3 MHz	Sky	AM radio
high frequency (HF)	3–30 MHz	Sky	Citizens band (CB), ship/aircraft
very high frequency (VHF)	30–300 MHz	Sky and line-of-sight	VHF TV, FM radio
ultrahigh frequency (UHF)	300 MHz–3 GHz	Line-of-sight	UHF TV, cellular phones, paging, satellite
superhigh frequency (SF)	3–30 GHz	Line-of-sight	Satellite
extremely high frequency (EHF)	30–300 GHz	Line-of-sight	Radar, satellite

We can divide wireless transmission into three broad groups: radio waves, microwaves, and infrared waves.

7.3.1 Radio Waves

Although there is no clear-cut demarcation between radio waves and microwaves, electromagnetic waves ranging in frequencies between 3 kHz and 1 GHz are normally called **radio waves;** waves ranging in frequencies between 1 and 300 GHz are called **microwaves.** However, the behavior of the waves, rather than the frequencies, is a better criterion for classification.

Radio waves, for the most part, are omnidirectional. When an antenna transmits radio waves, they are propagated in all directions. This means that the sending and receiving antennas do not have to be aligned. A sending antenna sends waves that can be received by any receiving antenna. The omnidirectional property has a disadvantage, too. The radio waves transmitted by one antenna are susceptible to interference by another antenna that may send signals using the same frequency or band.

Radio waves, particularly those waves that propagate in the sky mode, can travel long distances. This makes radio waves a good candidate for long-distance broadcasting such as AM radio.

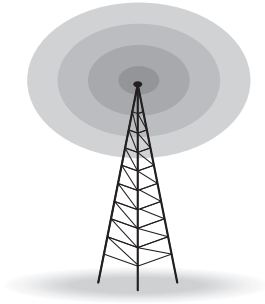
Radio waves, particularly those of low and medium frequencies, can penetrate walls. This characteristic can be both an advantage and a disadvantage. It is an advantage because, for example, an AM radio can receive signals inside a building. It is a disadvantage because we cannot isolate a communication to just inside or outside a building. The radio wave band is relatively narrow, just under 1 GHz, compared to the microwave band. When this band is divided into subbands, the subbands are also narrow, leading to a low data rate for digital communications.

Almost the entire band is regulated by authorities (e.g., the FCC in the United States). Using any part of the band requires permission from the authorities.

Omnidirectional Antenna

Radio waves use **omnidirectional antennas** that send out signals in all directions. Based on the wavelength, strength, and the purpose of transmission, we can have several types of antennas. Figure 7.19 shows an omnidirectional antenna.

Figure 7.19 *Omnidirectional antenna*



Applications

The omnidirectional characteristics of radio waves make them useful for multicasting, in which there is one sender but many receivers. AM and FM radio, television, maritime radio, cordless phones, and paging are examples of multicasting.

Radio waves are used for multicast communications, such as radio and television, and paging systems.

7.3.2 Microwaves

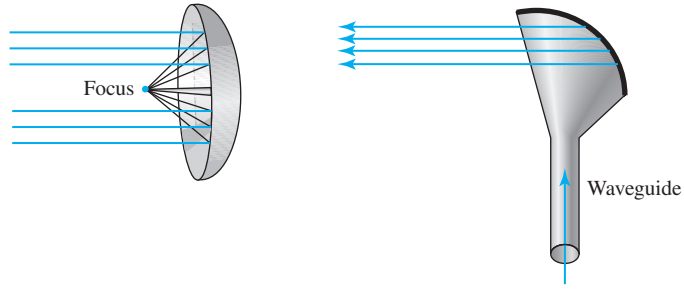
Electromagnetic waves having frequencies between 1 and 300 GHz are called microwaves. Microwaves are unidirectional. When an antenna transmits microwaves, they can be narrowly focused. This means that the sending and receiving antennas need to be aligned. The unidirectional property has an obvious advantage. A pair of antennas can be aligned without interfering with another pair of aligned antennas. The following describes some characteristics of microwave propagation:

- Microwave propagation is line-of-sight. Since the towers with the mounted antennas need to be in direct sight of each other, towers that are far apart need to be very tall. The curvature of the earth as well as other blocking obstacles do not allow two short towers to communicate by using microwaves. Repeaters are often needed for long-distance communication.
- Very high-frequency microwaves cannot penetrate walls. This characteristic can be a disadvantage if receivers are inside buildings.
- The microwave band is relatively wide, almost 299 GHz. Therefore wider subbands can be assigned, and a high data rate is possible.
- Use of certain portions of the band requires permission from authorities.

Unidirectional Antenna

Microwaves need **unidirectional antennas** that send out signals in one direction. Two types of antennas are used for microwave communications: the parabolic dish and the horn (see Figure 7.20).

Figure 7.20 *Unidirectional antennas*



a. Parabolic dish antenna b. Horn antenna

A **parabolic dish antenna** is based on the geometry of a parabola: Every line parallel to the line of symmetry (line of sight) reflects off the curve at angles such that all the lines intersect in a common point called the focus. The parabolic dish works as a funnel, catching a wide range of waves and directing them to a common point. In this way, more of the signal is recovered than would be possible with a single-point receiver.

Outgoing transmissions are broadcast through a horn aimed at the dish. The microwaves hit the dish and are deflected outward in a reversal of the receipt path.

A **horn antenna** looks like a gigantic scoop. Outgoing transmissions are broadcast up a stem (resembling a handle) and deflected outward in a series of narrow parallel beams by the curved head. Received transmissions are collected by the scooped shape of the horn, in a manner similar to the parabolic dish, and are deflected down into the stem.

Applications

Microwaves, due to their unidirectional properties, are very useful when unicast (one-to-one) communication is needed between the sender and the receiver. They are used in cellular phones (Chapter 16), satellite networks (Chapter 16), and wireless LANs (Chapter 15).

Microwaves are used for unicast communication such as cellular telephones, satellite networks, and wireless LANs.

7.3.3 Infrared

Infrared waves, with frequencies from 300 GHz to 400 THz (wavelengths from 1 mm to 770 nm), can be used for short-range communication. Infrared waves, having high frequencies, cannot penetrate walls. This advantageous characteristic prevents interference between one system and another; a short-range communication system in one room cannot be affected by another system in the next room. When we use our infrared remote control, we do not interfere with the use of the remote by our neighbors. However, this same characteristic makes infrared signals useless for long-range communication. In addition, we cannot use infrared waves outside a building because the sun's rays contain infrared waves that can interfere with the communication.

Applications

The infrared band, almost 400 THz, has an excellent potential for data transmission. Such a wide bandwidth can be used to transmit digital data with a very high data rate. The *Infrared Data Association* (IrDA), an association for sponsoring the use of infrared waves, has established standards for using these signals for communication between devices such as keyboards, mice, PCs, and printers. For example, some manufacturers provide a special port called the **IrDA port** that allows a wireless keyboard to commnicate with a PC. The standard originally defined a data rate of 75 kbps for a distance up to 8 m. The recent standard defines a data rate of 4 Mbps.

Infrared signals defined by IrDA transmit through line of sight; the IrDA port on the keyboard needs to point to the PC for transmission to occur.

Infrared signals can be used for short-range communication in a closed area using line-of-sight propagation

7.4 END-CHAPTER MATERIALS

7.4.1 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

Transmission media is discussed in [GW04], [Sta04], and [Tan03]. [SSS05] gives full coverage of transmission media.

7.4.2 Key Terms

angle of incidence
Bayonet Neill-Concelman (BNC)
cladding
coaxial cable
core
critical angle
electromagnetic spectrum
fiber-optic cable
gauge
ground propagation
guided media
horn antenna
infrared wave
IrDA port
line-of-sight propagation
microwave
MT-RJ
multimode graded-index fiber
multimode step-index fiber
omnidirectional antenna
optical fiber
parabolic dish antenna
Radio Government (RG) rating
radio wave
reflection
refraction
RJ45
shielded twisted-pair (STP)
single-mode fiber
sky propagation
straight-tip (ST) connector
subscriber channel (SC) connector
transmission medium
twisted-pair cable
unguided medium
unidirectional antenna
unshielded twisted-pair (UTP)
wireless communication

7.4.3 Summary

Transmission media are actually located below the physical layer and are directly controlled by the physical layer. We could say that transmission media belong to layer zero.

A guided medium provides a physical conduit from one device to another. Twisted-pair cable consists of two insulated copper wires twisted together. Twisted-pair cable is used for voice and data communications. Coaxial cable consists of a central conductor and a shield. Coaxial cable is used in cable TV networks and traditional Ethernet LANs. Fiber-optic cables are composed of a glass or plastic inner core surrounded by cladding, all encased in an outside jacket. Fiber-optic transmission is becoming increasingly popular due to its noise resistance, low attenuation, and high-bandwidth capabilities. Fiber-optic cable is used in backbone networks, cable TV networks, and Fast Ethernet networks.

Unguided media (free space) transport electromagnetic waves without the use of a physical conductor. Wireless data are transmitted through ground propagation, sky propagation, and line-of-sight propagation.Wireless waves can be classified as radio waves, microwaves, or infrared waves. Radio waves are omnidirectional; microwaves are unidirectional. Microwaves are used for cellular phone, satellite, and wireless LAN communications. Infrared waves are used for short-range communications such as those between a PC and a peripheral device. They can also be used for indoor LANs.

7.5 PRACTICE SET

7.5.1 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.

7.5.2 Questions

Q7-1. What is the position of the transmission media in the OSI or the Internet model?

Q7-2. Name the two major categories of transmission media.

Q7-3. How do guided media differ from unguided media?

Q7-4. What are the three major classes of guided media?

Q7-5. What is the function of the twisting in twisted-pair cable?

Q7-6. What is refraction? What is reflection?

Q7-7. What is the purpose of cladding in an optical fiber?

Q7-8. Name the advantages of optical fiber over twisted-pair and coaxial cable.

Q7-9. How does sky propagation differ from line-of-sight propagation?

Q7-10. What is the difference between omnidirectional waves and unidirectional waves?

7.5.3 Problems

P7-1. Using Figure 7.6, tabulate the attenuation (in dB) of a 18-gauge UTP for the indicated frequencies and distances.

Table 7.5 *Attenuation for 18-gauge UTP*

Distance	*dB at 1 KHz*	*dB at 10 KHz*	*dB at 100 KHz*
1 Km			
10 Km			
15 Km			
20 Km			

P7-2. Use the results of Problem P7-1 to infer that the bandwidth of a UTP cable decreases with an increase in distance.

P7-3. If the power at the beginning of a 1 Km 18-gauge UTP is 200 mw, what is the power at the end for frequencies 1 KHz, 10 KHz, and 100 KHz? Use the results of Problem P7-1.

P7-4. Using Figure 7.9, tabulate the attenuation (in dB) of a 2.6/9.5 mm coaxial cable for the indicated frequencies and distances.

Table 7.6 *Attenuation for 2.6/9.5 mm coaxial cable*

Distance	*dB at 1 KHz*	*dB at 10 KHz*	*dB at 100 KHz*
1 Km			
10 Km			
15 Km			
20 Km			

P7-5. Use the results of Problem P7-4 to infer that the bandwidth of a coaxial cable decreases with the increase in distance.

P7-6. If the power at the beginning of a 1 Km 2.6/9.5 mm coaxial cable is 200 mw, what is the power at the end for frequencies 1 KHz, 10 KHz, and 100 KHz? Use the results of Problem P7-4.

P7-7. Calculate the bandwidth of the light for the following wavelength ranges (assume a propagation speed of 2×10^8 m):

a. 1000 to 1200 nm **b.** 1000 to 1400 nm

P7-8. The horizontal axes in Figures 7.6 and 7.9 represent frequencies. The horizontal axis in Figure 7.16 represents wavelength. Can you explain the reason? If the propagation speed in an optical fiber is 2×10^8 m, can you change the units in the horizontal axis to frequency? Should the vertical-axis units be changed too? Should the curve be changed too?

P7-9. Using Figure 7.16, tabulate the attenuation (in dB) of an optical fiber for the indicated wavelength and distances.

Table 7.7 *Attenuation for optical fiber*

Distance	*dB at 800 nm*	*dB at 1000 nm*	*dB at 1200 nm*
1 Km			
10 Km			
15 Km			
20 Km			

P7-10. A light signal is travelling through a fiber. What is the delay in the signal if the length of the fiber-optic cable is 10 m, 100 m, and 1 Km (assume a propagation speed of 2×10^8 m)?

P7-11. A beam of light moves from one medium to another medium with less density. The critical angle is 60°. Do we have refraction or reflection for each of the following incident angles? Show the bending of the light ray in each case.

a. 40° **b.** 60° **c.** 80°

CHAPTER 8

Switching

Switching is a topic that can be discussed at several layers. We have switching at the physical layer, at the data-link layer, at the network layer, and even logically at the application layer (message switching). We have decided to discuss the general idea behind switching in this chapter, the last chapter related to the physical layer. We particularly discuss circuit-switching, which occurs at the physical layer. We introduce the idea of packet-switching, which occurs at the data-link and network layers, but we postpone the details of these topics until the appropriate chapters. Finally, we talk about the physical structures of the switches and routers.

This chapter is divided into four sections:

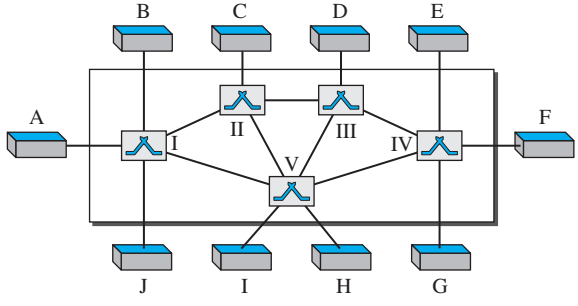
- The first section introduces switching. It mentions three methods of switching: circuit switching, packet switching, and message switching. The section then defines the switching methods that can occur in some layers of the Internet model.
- The second section discusses circuit-switched networks. It first defines three phases in these types of networks. It then describes the efficiency of these networks. The section also discusses the delay in circuit-switched networks.
- The third section briefly discusses packet-switched networks. It first describes datagram networks, listing their characteristics and advantages. The section then describes virtual circuit networks, explaining their features and operations. We will discuss packet-switched networks in more detail in Chapter 18.
- The last section discusses the structure of a switch. It first describes the structure of a circuit switch. It then explains the structure of a packet switch.

8.1 INTRODUCTION

A network is a set of connected devices. Whenever we have multiple devices, we have the problem of how to connect them to make one-to-one communication possible. One solution is to make a point-to-point connection between each pair of devices (a mesh topology) or between a central device and every other device (a star topology). These methods, however, are impractical and wasteful when applied to very large networks. The number and length of the links require too much infrastructure to be cost-efficient, and the majority of those links would be idle most of the time. Other topologies employing multipoint connections, such as a bus, are ruled out because the distances between devices and the total number of devices increase beyond the capacities of the media and equipment.

A better solution is **switching.** A switched network consists of a series of interlinked nodes, called ***switches*****.** Switches are devices capable of creating temporary connections between two or more devices linked to the switch. In a switched network, some of these nodes are connected to the end systems (computers or telephones, for example). Others are used only for routing. Figure 8.1 shows a switched network.

Figure 8.1 *Switched network*

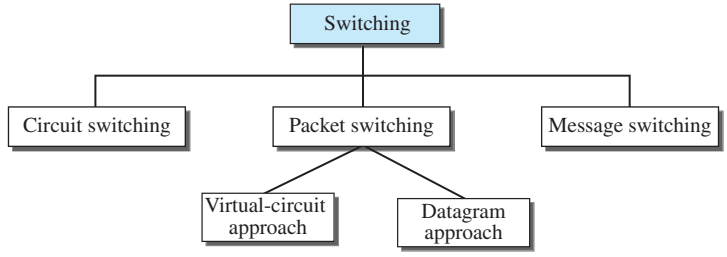


The **end systems** (communicating devices) are labeled A, B, C, D, and so on, and the switches are labeled I, II, III, IV, and V. Each switch is connected to multiple links.

8.1.1 Three Methods of Switching

Traditionally, three methods of switching have been discussed: **circuit switching**, **packet switching**, and **message switching**. The first two are commonly used today. The third has been phased out in general communications but still has networking applications. Packet switching can further be divided into two subcategories—virtual-circuit approach and datagram approach—as shown in Figure 8.2. In this chapter, we discuss only circuit switching and packet switching; message switching is more conceptual than practical.

Figure 8.2 *Taxonomy of switched networks*



8.1.2 Switching and TCP/IP Layers

Switching can happen at several layers of the TCP/IP protocol suite.

Switching at Physical Layer

At the physical layer, we can have only circuit switching. There are no packets exchanged at the physical layer. The switches at the physical layer allow signals to travel in one path or another.

Switching at Data-Link Layer

At the data-link layer, we can have packet switching. However, the term *packet* in this case means *frames* or *cells*. Packet switching at the data-link layer is normally done using a virtual-circuit approach.

Switching at Network Layer

At the network layer, we can have packet switching. In this case, either a virtual-circuit approach or a datagram approach can be used. Currently the Internet uses a datagram approach, as we see in Chapter 18, but the tendency is to move to a virtual-circuit approach.

Switching at Application Layer

At the application layer, we can have only message switching. The communication at the application layer occurs by exchanging messages. Conceptually, we can say that communication using e-mail is a kind of message-switched communication, but we do not see any network that actually can be called a message-switched network.

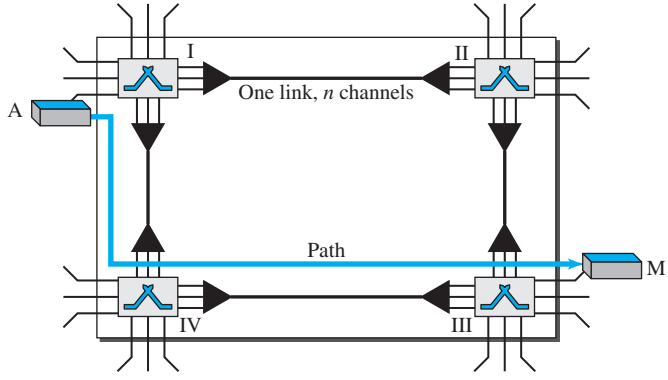
8.2 CIRCUIT-SWITCHED NETWORKS

A **circuit-switched network** consists of a set of switches connected by physical links. A connection between two stations is a dedicated path made of one or more links. However, each connection uses only one dedicated channel on each link. Each link is normally divided into *n* channels by using FDM or TDM, as discussed in Chapter 6.

A circuit-switched network is made of a set of switches connected by physical links, in which each link is divided into *n* channels.

Figure 8.3 shows a trivial circuit-switched network with four switches and four links. Each link is divided into *n* (*n* is 3 in the figure) channels by using FDM or TDM.

Figure 8.3 *A trivial circuit-switched network*



We have explicitly shown the multiplexing symbols to emphasize the division of the link into channels even though multiplexing can be implicitly included in the switch fabric.

The end systems, such as computers or telephones, are directly connected to a switch. We have shown only two end systems for simplicity. When end system A needs to communicate with end system M, system A needs to request a connection to M that must be accepted by all switches as well as by M itself. This is called the **setup phase;** a circuit (channel) is reserved on each link, and the combination of circuits or channels defines the dedicated path. After the dedicated path made of connected circuits (channels) is established, the **data-transfer phase** can take place. After all data have been transferred, the circuits are torn down.

We need to emphasize several points here:

- Circuit switching takes place at the physical layer.
- Before starting communication, the stations must make a reservation for the resources to be used during the communication. These resources, such as channels (bandwidth in FDM and time slots in TDM), switch buffers, switch processing time, and switch input/output ports, must remain dedicated during the entire duration of data transfer until the **teardown phase.**
- Data transferred between the two stations are not packetized (physical layer transfer of the signal). The data are a continuous flow sent by the source station and received by the destination station, although there may be periods of silence.

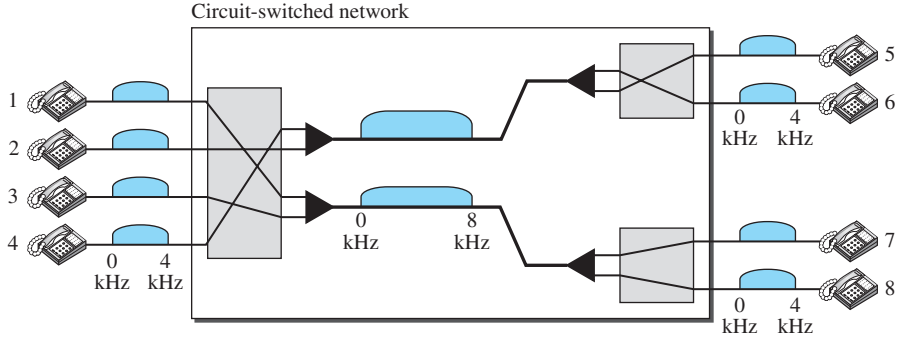
- There is no addressing involved during data transfer. The switches route the data based on their occupied band (FDM) or time slot (TDM). Of course, there is end-to-end addressing used during the setup phase, as we will see shortly.

In circuit switching, the resources need to be reserved during the setup phase; the resources remain dedicated for the entire duration of data transfer until the teardown phase.

Example 8.1

As a trivial example, let us use a circuit-switched network to connect eight telephones in a small area. Communication is through 4-kHz voice channels. We assume that each link uses FDM to connect a maximum of two voice channels. The bandwidth of each link is then 8 kHz. Figure 8.4 shows the situation. Telephone 1 is connected to telephone 7; 2 to 5; 3 to 8; and 4 to 6. Of course the situation may change when new connections are made. The switch controls the connections.

Figure 8.4 *Circuit-switched network used in Example 8.1*



Example 8.2

As another example, consider a circuit-switched network that connects computers in two remote offices of a private company. The offices are connected using a T-1 line leased from a communication service provider. There are two 4×8 (4 inputs and 8 outputs) switches in this network. For each switch, four output ports are folded into the input ports to allow communication between computers in the same office. Four other output ports allow communication between the two offices. Figure 8.5 shows the situation.

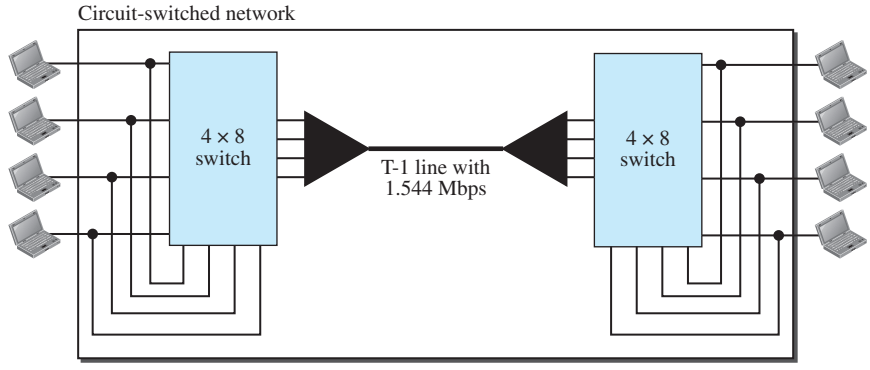
8.2.1 Three Phases

The actual communication in a circuit-switched network requires three phases: connection setup, data transfer, and connection teardown.

Setup Phase

Before the two parties (or multiple parties in a conference call) can communicate, a dedicated circuit (combination of channels in links) needs to be established. The end systems are normally connected through dedicated lines to the switches, so connection setup

Figure 8.5 *Circuit-switched network used in Example 8.2*



means creating dedicated channels between the switches. For example, in Figure 8.3, when system A needs to connect to system M, it sends a setup request that includes the address of system M, to switch I. Switch I finds a channel between itself and switch IV that can be dedicated for this purpose. Switch I then sends the request to switch IV, which finds a dedicated channel between itself and switch III. Switch III informs system M of system A's intention at this time.

In the next step to making a connection, an acknowledgment from system M needs to be sent in the opposite direction to system A. Only after system A receives this acknowledgment is the connection established.

Note that end-to-end addressing is required for creating a connection between the two end systems. These can be, for example, the addresses of the computers assigned by the administrator in a TDM network, or telephone numbers in an FDM network.

Data-Transfer Phase

After the establishment of the dedicated circuit (channels), the two parties can transfer data.

Teardown Phase

When one of the parties needs to disconnect, a signal is sent to each switch to release the resources.

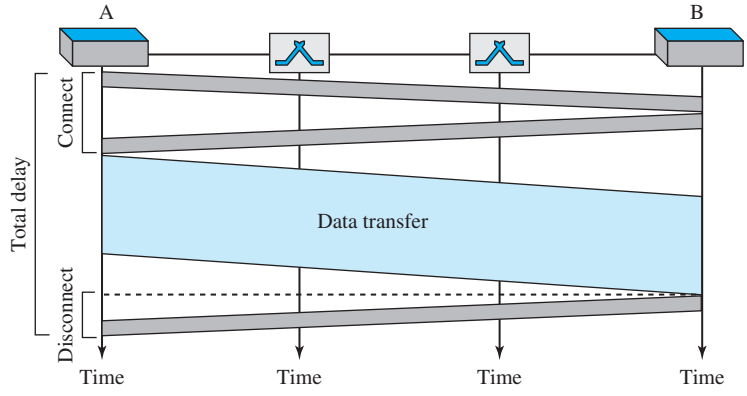
8.2.2 Efficiency

It can be argued that circuit-switched networks are not as efficient as the other two types of networks because resources are allocated during the entire duration of the connection. These resources are unavailable to other connections. In a telephone network, people normally terminate the communication when they have finished their conversation. However, in computer networks, a computer can be connected to another computer even if there is no activity for a long time. In this case, allowing resources to be dedicated means that other connections are deprived.

8.2.3 Delay

Although a circuit-switched network normally has low efficiency, the delay in this type of network is minimal. During data transfer the data are not delayed at each switch; the resources are allocated for the duration of the connection. Figure 8.6 shows the idea of delay in a circuit-switched network when only two switches are involved.

Figure 8.6 *Delay in a circuit-switched network*



As Figure 8.6 shows, there is no waiting time at each switch. The total delay is due to the time needed to create the connection, transfer data, and disconnect the circuit. The delay caused by the setup is the sum of four parts: the propagation time of the source computer request (slope of the first gray box), the request signal transfer time (height of the first gray box), the propagation time of the acknowledgment from the destination computer (slope of the second gray box), and the signal transfer time of the acknowledgment (height of the second gray box). The delay due to data transfer is the sum of two parts: the propagation time (slope of the colored box) and data transfer time (height of the colored box), which can be very long. The third box shows the time needed to tear down the circuit. We have shown the case in which the receiver requests disconnection, which creates the maximum delay.

8.3 PACKET SWITCHING

In data communications, we need to send messages from one end system to another. If the message is going to pass through a **packet-switched network**, it needs to be divided into packets of fixed or variable size. The size of the packet is determined by the network and the governing protocol.

In packet switching, there is no resource allocation for a packet. This means that there is no reserved bandwidth on the links, and there is no scheduled processing time for each packet. Resources are allocated on demand. The allocation is done on a first-come, first-served basis. When a switch receives a packet, no matter what the source or destination is, the packet must wait if there are other packets being processed. As with

other systems in our daily life, this lack of reservation may create delay. For example, if we do not have a reservation at a restaurant, we might have to wait.

In a packet-switched network, there is no resource reservation; resources are allocated on demand.

We can have two types of packet-switched networks: datagram networks and virtual-circuit networks.

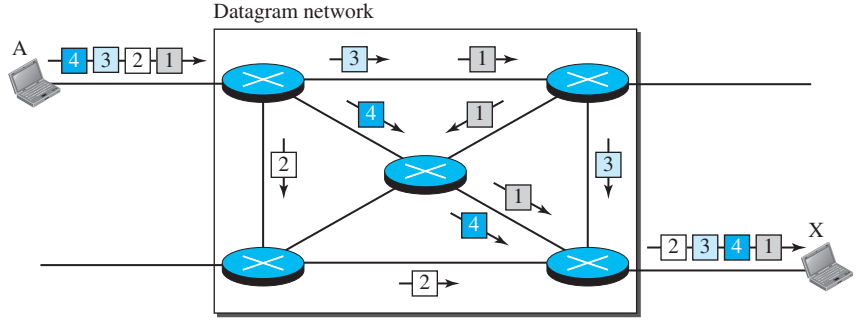
8.3.1 Datagram Networks

In a **datagram network**, each packet is treated independently of all others. Even if a packet is part of a multipacket transmission, the network treats it as though it existed alone. Packets in this approach are referred to as ***datagrams***.

Datagram switching is normally done at the network layer. We briefly discuss datagram networks here as a comparison with circuit-switched and virtual-circuit-switched networks. In Chapter 18 of this text, we go into greater detail.

Figure 8.7 shows how the datagram approach is used to deliver four packets from station A to station X. The switches in a datagram network are traditionally referred to as routers. That is why we use a different symbol for the switches in the figure.

Figure 8.7 *A datagram network with four switches (routers)*



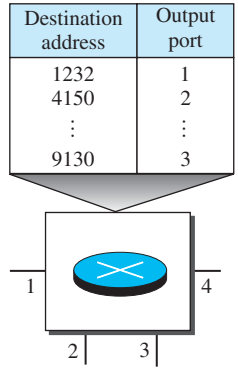
In this example, all four packets (or datagrams) belong to the same message, but may travel different paths to reach their destination. This is so because the links may be involved in carrying packets from other sources and do not have the necessary bandwidth available to carry all the packets from A to X. This approach can cause the datagrams of a transmission to arrive at their destination out of order with different delays between the packets. Packets may also be lost or dropped because of a lack of resources. In most protocols, it is the responsibility of an upper-layer protocol to reorder the datagrams or ask for lost datagrams before passing them on to the application.

The datagram networks are sometimes referred to as *connectionless networks*. The term *connectionless* here means that the switch (packet switch) does not keep information about the connection state. There are no setup or teardown phases. Each packet is treated the same by a switch regardless of its source or destination.

Routing Table

If there are no setup or teardown phases, how are the packets routed to their destinations in a datagram network? In this type of network, each switch (or packet switch) has a routing table which is based on the destination address. The routing tables are dynamic and are updated periodically. The destination addresses and the corresponding forwarding output ports are recorded in the tables. This is different from the table of a circuit-switched network (discussed later) in which each entry is created when the setup phase is completed and deleted when the teardown phase is over. Figure 8.8 shows the routing table for a switch.

Figure 8.8 *Routing table in a datagram network*



A switch in a datagram network uses a routing table that is based on the destination address.

Destination Address

Every packet in a datagram network carries a header that contains, among other information, the destination address of the packet. When the switch receives the packet, this destination address is examined; the routing table is consulted to find the corresponding port through which the packet should be forwarded. This address, unlike the address in a virtual-circuit network, remains the same during the entire journey of the packet.

The destination address in the header of a packet in a datagram network remains the same during the entire journey of the packet.

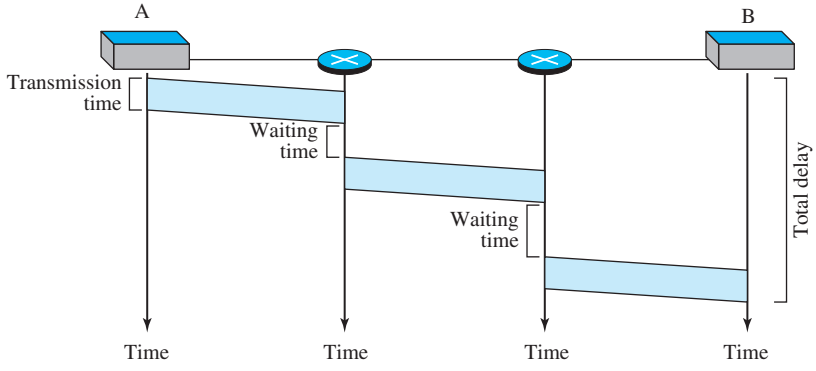
Efficiency

The efficiency of a datagram network is better than that of a circuit-switched network; resources are allocated only when there are packets to be transferred. If a source sends a packet and there is a delay of a few minutes before another packet can be sent, the resources can be reallocated during these minutes for other packets from other sources.

Delay

There may be greater delay in a datagram network than in a virtual-circuit network. Although there are no setup and teardown phases, each packet may experience a wait at a switch before it is forwarded. In addition, since not all packets in a message necessarily travel through the same switches, the delay is not uniform for the packets of a message. Figure 8.9 gives an example of delay in a datagram network for one packet.

Figure 8.9 *Delay in a datagram network*



The packet travels through two switches. There are three transmission times ($3T$), three propagation delays (slopes 3τ of the lines), and two waiting times ($w_1 + w_2$). We ignore the processing time in each switch. The total delay is

$$\textbf{Total delay} = 3T + 3\tau + w_1 + w_2$$

8.3.2 Virtual-Circuit Networks

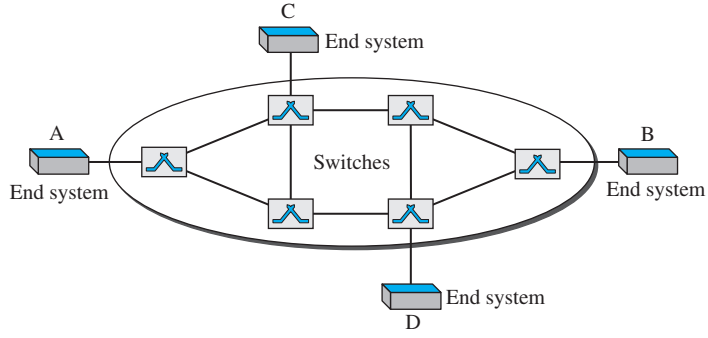
A **virtual-circuit network** is a cross between a circuit-switched network and a datagram network. It has some characteristics of both.

1. As in a circuit-switched network, there are setup and teardown phases in addition to the data transfer phase.
2. Resources can be allocated during the setup phase, as in a circuit-switched network, or on demand, as in a datagram network.
3. As in a datagram network, data are packetized and each packet carries an address in the header. However, the address in the header has local jurisdiction (it defines what the next switch should be and the channel on which the packet is being carried), not end-to-end jurisdiction. The reader may ask how the intermediate switches know where to send the packet if there is no final destination address carried by a packet. The answer will be clear when we discuss virtual-circuit identifiers in the next section.
4. As in a circuit-switched network, all packets follow the same path established during the connection.

5. A virtual-circuit network is normally implemented in the data-link layer, while a circuit-switched network is implemented in the physical layer and a datagram network in the network layer. But this may change in the future.

Figure 8.10 is an example of a virtual-circuit network. The network has switches that allow traffic from sources to destinations. A source or destination can be a computer, packet switch, bridge, or any other device that connects other networks.

Figure 8.10 *Virtual-circuit network*



Addressing

In a virtual-circuit network, two types of addressing are involved: global and local (virtual-circuit identifier).

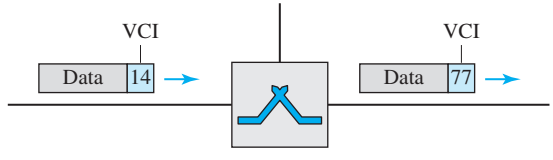
Global Addressing

A source or a destination needs to have a global address—an address that can be unique in the scope of the network or internationally if the network is part of an international network. However, we will see that a global address in virtual-circuit networks is used only to create a virtual-circuit identifier, as discussed next.

Virtual-Circuit Identifier

The identifier that is actually used for data transfer is called the ***virtual-circuit identifier*** **(VCI)** or the ***label.*** A VCI, unlike a global address, is a small number that has only switch scope; it is used by a frame between two switches. When a frame arrives at a switch, it has a VCI; when it leaves, it has a different VCI. Figure 8.11 shows how the VCI in a data frame changes from one switch to another. Note that a VCI does not need to be a large number since each switch can use its own unique set of VCIs.

Figure 8.11 *Virtual-circuit identifier*



Three Phases

As in a circuit-switched network, a source and destination need to go through three phases in a virtual-circuit network: setup, data transfer, and teardown. In the setup phase, the source and destination use their global addresses to help switches make table entries for the connection. In the teardown phase, the source and destination inform the switches to delete the corresponding entry. Data transfer occurs between these two phases. We first discuss the data-transfer phase, which is more straightforward; we then talk about the setup and teardown phases.

Data-Transfer Phase

To transfer a frame from a source to its destination, all switches need to have a table entry for this virtual circuit. The table, in its simplest form, has four columns. This means that the switch holds four pieces of information for each virtual circuit that is already set up. We show later how the switches make their table entries, but for the moment we assume that each switch has a table with entries for all active virtual circuits. Figure 8.12 shows such a switch and its corresponding table.

Figure 8.12 *Switch and tables in a virtual-circuit network*

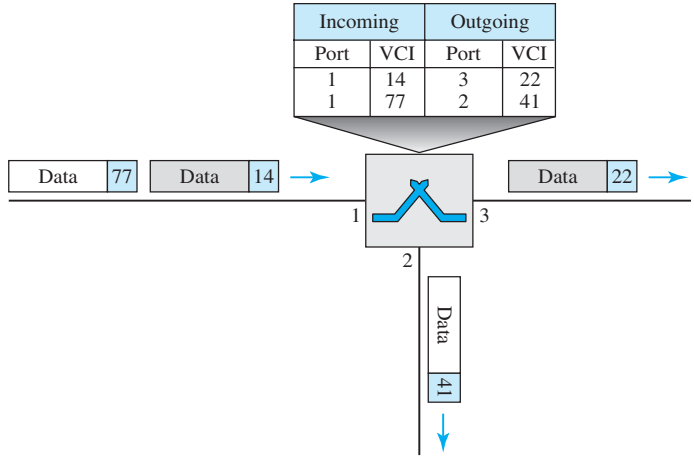


Figure 8.12 shows a frame arriving at port 1 with a VCI of 14. When the frame arrives, the switch looks in its table to find port 1 and a VCI of 14. When it is found, the switch knows to change the VCI to 22 and send out the frame from port 3.

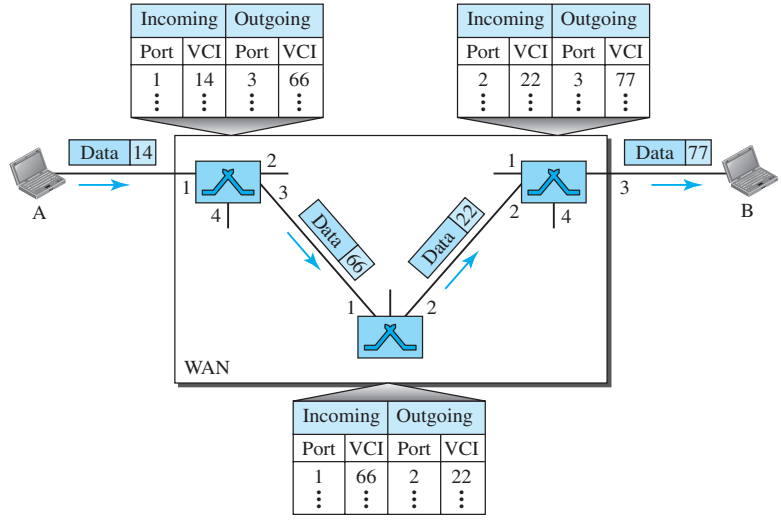
Figure 8.13 shows how a frame from source A reaches destination B and how its VCI changes during the trip. Each switch changes the VCI and routes the frame.

The data-transfer phase is active until the source sends all its frames to the destination. The procedure at the switch is the same for each frame of a message. The process creates a virtual circuit, not a real circuit, between the source and destination.

Setup Phase

In the setup phase, a switch creates an entry for a virtual circuit. For example, suppose source A needs to create a virtual circuit to B. Two steps are required: the setup request and the acknowledgment.

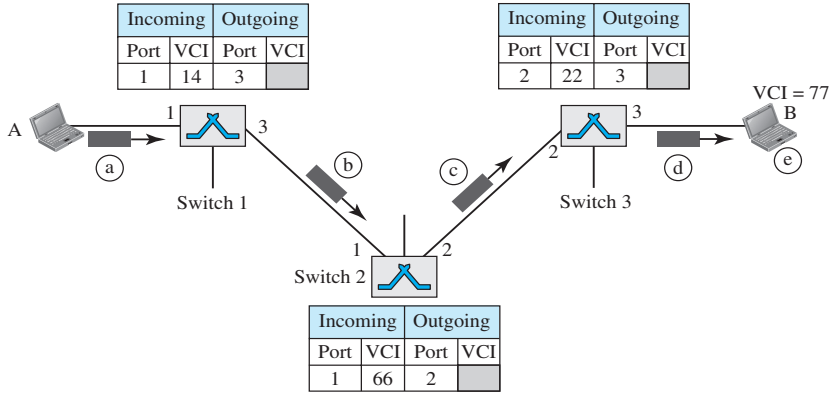
Figure 8.13 *Source-to-destination data transfer in a virtual-circuit network*



Setup Request

A setup request frame is sent from the source to the destination. Figure 8.14 shows the process.

Figure 8.14 *Setup request in a virtual-circuit network*



a. Source A sends a setup frame to switch 1.

b. Switch 1 receives the setup request frame. It knows that a frame going from A to B goes out through port 3. How the switch has obtained this information is a point covered in future chapters. The switch, in the setup phase, acts as a packet switch; it has a routing table which is different from the switching table. For the moment, assume that it knows the output port. The switch creates an entry in its table for this virtual circuit, but it is only able to fill three of the four columns. The switch assigns the incoming port (1) and chooses an available incoming VCI (14) and the

outgoing port (3). It does not yet know the outgoing VCI, which will be found during the acknowledgment step. The switch then forwards the frame through port 3 to switch 2.

c. Switch 2 receives the setup request frame. The same events happen here as at switch 1; three columns of the table are completed: in this case, incoming port (1), incoming VCI (66), and outgoing port (2).

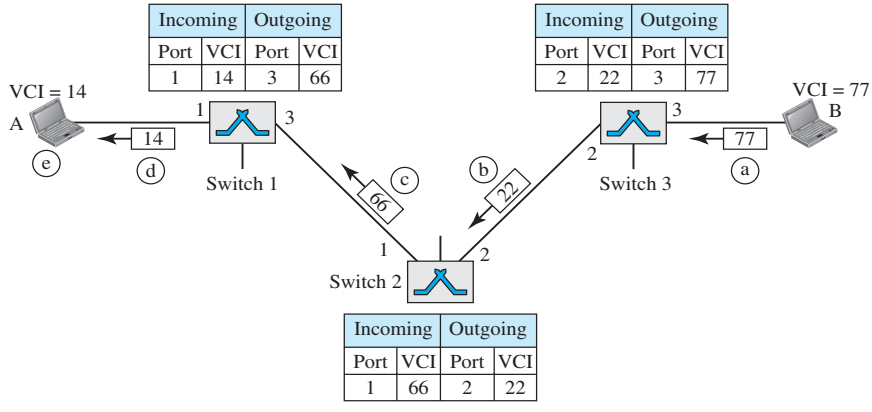
d. Switch 3 receives the setup request frame. Again, three columns are completed: incoming port (2), incoming VCI (22), and outgoing port (3).

e. Destination B receives the setup frame, and if it is ready to receive frames from A, it assigns a VCI to the incoming frames that come from A, in this case 77. This VCI lets the destination know that the frames come from A, and not other sources.

Acknowledgment

A special frame, called the *acknowledgment frame,* completes the entries in the switching tables. Figure 8.15 shows the process.

Figure 8.15 *Setup acknowledgment in a virtual-circuit network*



a. The destination sends an acknowledgment to switch 3. The acknowledgment carries the global source and destination addresses so the switch knows which entry in the table is to be completed. The frame also carries VCI 77, chosen by the destination as the incoming VCI for frames from A. Switch 3 uses this VCI to complete the outgoing VCI column for this entry. Note that 77 is the incoming VCI for destination B, but the outgoing VCI for switch 3.

b. Switch 3 sends an acknowledgment to switch 2 that contains its incoming VCI in the table, chosen in the previous step. Switch 2 uses this as the outgoing VCI in the table.

c. Switch 2 sends an acknowledgment to switch 1 that contains its incoming VCI in the table, chosen in the previous step. Switch 1 uses this as the outgoing VCI in the table.

d. Finally switch 1 sends an acknowledgment to source A that contains its incoming VCI in the table, chosen in the previous step.

e. The source uses this as the outgoing VCI for the data frames to be sent to destination B.

Teardown Phase

In this phase, source A, after sending all frames to B, sends a special frame called a *teardown request*. Destination B responds with a teardown confirmation frame. All switches delete the corresponding entry from their tables.

Efficiency

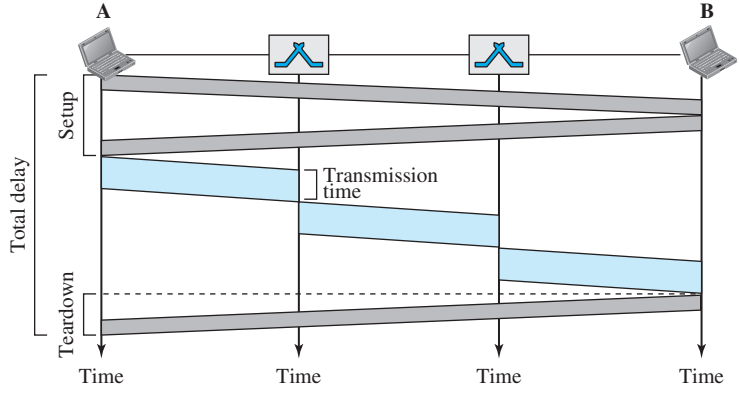
As we said before, resource reservation in a virtual-circuit network can be made during the setup or can be on demand during the data-transfer phase. In the first case, the delay for each packet is the same; in the second case, each packet may encounter different delays. There is one big advantage in a virtual-circuit network even if resource allocation is on demand. The source can check the availability of the resources, without actually reserving it. Consider a family that wants to dine at a restaurant. Although the restaurant may not accept reservations (allocation of the tables is on demand), the family can call and find out the waiting time. This can save the family time and effort.

In virtual-circuit switching, all packets belonging to the same source and destination travel the same path, but the packets may arrive at the destination with different delays if resource allocation is on demand.

Delay in Virtual-Circuit Networks

In a virtual-circuit network, there is a one-time delay for setup and a one-time delay for teardown. If resources are allocated during the setup phase, there is no wait time for individual packets. Figure 8.16 shows the delay for a packet traveling through two switches in a virtual-circuit network.

Figure 8.16 *Delay in a virtual-circuit network*



The packet is traveling through two switches (routers). There are three transmission times ($3T$), three propagation times (3τ), data transfer depicted by the sloping lines, a setup delay (which includes transmission and propagation in two directions),

and a teardown delay (which includes transmission and propagation in one direction). We ignore the processing time in each switch. The total delay time is

Total delay = 3*T* + 3τ + setup delay + teardown delay

Circuit-Switched Technology in WANs

As we will see in Chapter 14, virtual-circuit networks are used in switched WANs such as ATM networks. The data-link layer of these technologies is well suited to the virtual-circuit technology.

Switching at the data-link layer in a switched WAN is normally implemented by using virtual-circuit techniques.

8.4 STRUCTURE OF A SWITCH

We use switches in circuit-switched and packet-switched networks. In this section, we discuss the structures of the switches used in each type of network.

8.4.1 Structure of Circuit Switches

Circuit switching today can use either of two technologies: the space-division switch or the time-division switch.

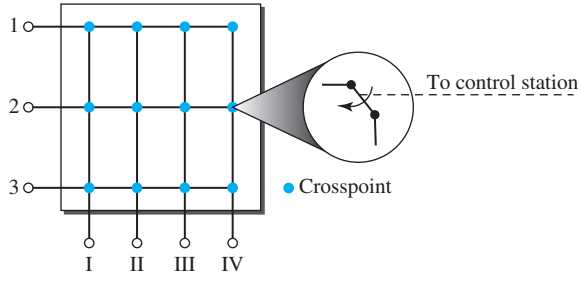
Space-Division Switch

In **space-division switching,** the paths in the circuit are separated from one another spatially. This technology was originally designed for use in analog networks but is used currently in both analog and digital networks. It has evolved through a long history of many designs.

Crossbar Switch

A **crossbar switch** connects *n* inputs to *m* outputs in a grid, using electronic microswitches (transistors) at each **crosspoint** (see Figure 8.17). The major limitation of this design is the number of crosspoints required. To connect *n* inputs to *m* outputs using a

Figure 8.17 *Crossbar switch with three inputs and four outputs*

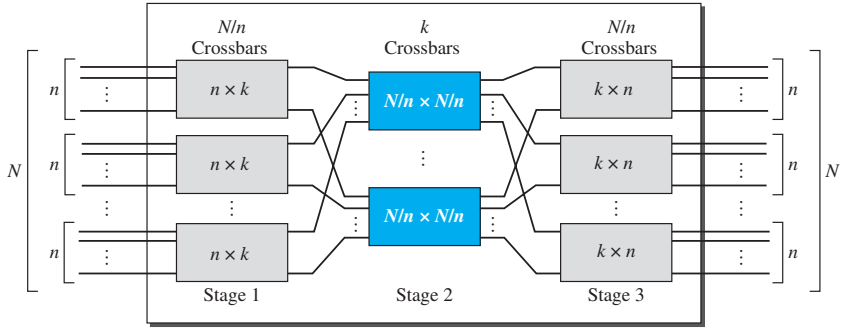


crossbar switch requires $n \times m$ crosspoints. For example, to connect 1000 inputs to 1000 outputs requires a switch with 1,000,000 crosspoints. A crossbar switch [?] with this number of crosspoints is impractical. Such a switch is also inefficient because statistics show that, in practice, fewer than 25 percent of the crosspoints are in use at any given time. The rest are idle.

Multistage Switch

The solution to the limitations of the crossbar switch is the **multistage switch,** which combines crossbar switches in several (normally three) stages, as shown in Figure 8.18. In a single crossbar switch, only one row or column (one path) is active for any connection. So we need $N \times N$ crosspoints. If we can allow multiple paths inside the switch, we can decrease the number of crosspoints. Each crosspoint in the middle stage can be accessed by multiple crosspoints in the first or third stage.

Figure 8.18 *Multistage switch*



To design a three-stage switch, we follow these steps:

1. We divide the N input lines into groups, each of n lines. For each group, we use one crossbar of size $n \times k$, where k is the number of crossbars in the middle stage. In other words, the first stage has N/n crossbars of $n \times k$ crosspoints.
2. We use k crossbars, each of size $(N/n) \times (N/n)$ in the middle stage.
3. We use N/n crossbars, each of size $k \times n$ at the third stage.

We can calculate the total number of crosspoints as follows:

$$\frac{N}{n}(n \times k) + k\left(\frac{N}{n} \times \frac{N}{n}\right) + \frac{N}{n}(k \times n) = 2kN + k\left(\frac{N}{n}\right)^2$$

In a three-stage switch, the total number of crosspoints is

$$2kN + k\left(\frac{N}{n}\right)^2$$

which is much smaller than the number of crosspoints in a single-stage switch (N^2).

Example 8.3

Design a three-stage, 200 × 200 switch ($N = 200$) with $k = 4$ and $n = 20$.

Solution

In the first stage we have N/n or 10 crossbars, each of size 20 × 4. In the second stage, we have 4 crossbars, each of size 10 × 10. In the third stage, we have 10 crossbars, each of size 4 × 20. The total number of crosspoints is $2kN + k(N/n)^2$, or 2000 crosspoints. This is 5 percent of the number of crosspoints in a single-stage switch (200 × 200 = 40,000).

The multistage switch in Example 8.3 has one drawback—**blocking** during periods of heavy traffic. The whole idea of multistage switching is to share the crosspoints in the middle-stage crossbars. Sharing can cause a lack of availability if the resources are limited and all users want a connection at the same time. *Blocking* refers to times when one input cannot be connected to an output because there is no path available between them—all the possible intermediate switches are occupied.

In a single-stage switch, blocking does not occur because every combination of input and output has its own crosspoint; there is always a path. (Cases in which two inputs are trying to contact the same output do not count. That path is not blocked; the output is merely busy.) In the multistage switch described in Example 8.3, however, only four of the first 20 inputs can use the switch at a time, only four of the second 20 inputs can use the switch at a time, and so on. The small number of crossbars at the middle stage creates blocking.

In large systems, such as those having 10,000 inputs and outputs, the number of stages can be increased to cut down on the number of crosspoints required. As the number of stages increases, however, possible blocking increases as well. Many people have experienced blocking on public telephone systems in the wake of a natural disaster when the calls being made to check on or reassure relatives far outnumber the regular load of the system.

Clos investigated the condition of nonblocking in multistage switches and came up with the following formula. In a nonblocking switch, the number of middle-stage switches must be at least $2n - 1$. In other words, we need to have $k \geq 2n - 1$.

Note that the number of crosspoints is still smaller than that in a single-stage switch. Now we need to minimize the number of crosspoints with a fixed N by using the Clos criteria. We can take the derivative of the equation with respect to n (the only variable) and find the value of n that makes the result zero. This n must be equal to or greater than $(N/2)^{1/2}$. In this case, the total number of crosspoints is greater than or equal to $4N\,[(2N)^{1/2} - 1]$. In other words, the minimum number of crosspoints according to the Clos criteria is proportional to $N^{3/2}$.

According to Clos criterion: $\boldsymbol{n = (N/2)^{1/2}}$ ***and*** $\boldsymbol{k \geq 2n - 1}$
Total number of crosspoints $\boldsymbol{\geq 4N\,[(2N)^{1/2} - 1]}$

Example 8.4

Redesign the previous three-stage, 200 × 200 switch, using the Clos criteria with a minimum number of crosspoints.

Solution

We let $n = (200/2)^{1/2}$, or $n = 10$. We calculate $k = 2n - 1 = 19$. In the first stage, we have 200/10, or 20, crossbars, each with 10 × 19 crosspoints. In the second stage, we have 19 crossbars,

each with 10 × 10 crosspoints. In the third stage, we have 20 crossbars each with 19 × 10 crosspoints. The total number of crosspoints is 20(10 × 19) + 19(10 × 10) + 20(19 × 10) = 9500. If we use a single-stage switch, we need 200 × 200 = 40,000 crosspoints. The number of crosspoints in this three-stage switch is 24 percent that of a single-stage switch. More points are needed than in Example 8.3 (5 percent). The extra crosspoints are needed to prevent blocking.

A multistage switch that uses the Clos criteria and a minimum number of crosspoints still requires a huge number of crosspoints. For example, to have a 100,000 input/output switch, we need something close to 200 million crosspoints (instead of 10 billion). This means that if a telephone company needs to provide a switch to connect 100,000 telephones in a city, it needs 200 million crosspoints. The number can be reduced if we accept blocking. Today, telephone companies use time-division switching or a combination of space- and time-division switches, as we will see shortly.

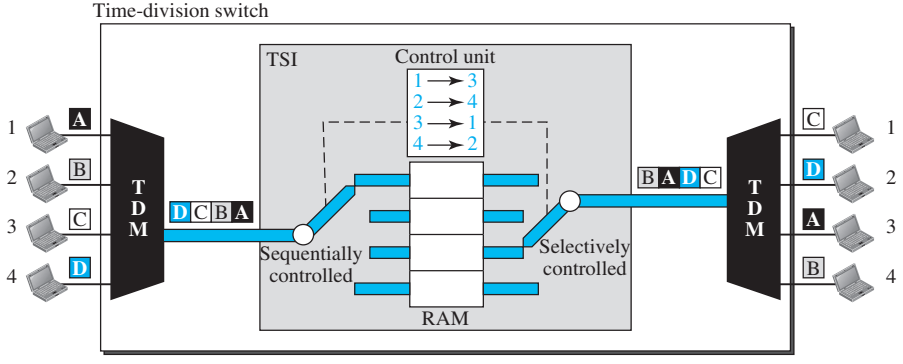
Time-Division Switch

Time-division switching uses time-division multiplexing (TDM) inside a switch. The most popular technology is called the **time-slot interchange (TSI).**

Time-Slot Interchange

Figure 8.19 shows a system connecting four input lines to four output lines. Imagine that each input line wants to send data to an output line according to the following pattern: (1 → 3), (2 → 4), (3 → 1), and (4 → 2), in which the arrow means "to."

Figure 8.19 *Time-slot interchange*



The figure combines a TDM multiplexer, a TDM demultiplexer, and a TSI consisting of random access memory (RAM) with several memory locations. The size of each location is the same as the size of a single time slot. The number of locations is the same as the number of inputs (in most cases, the numbers of inputs and outputs are equal). The RAM fills up with incoming data from time slots in the order received. Slots are then sent out in an order based on the decisions of a control unit.

Time- and Space-Division Switch Combinations

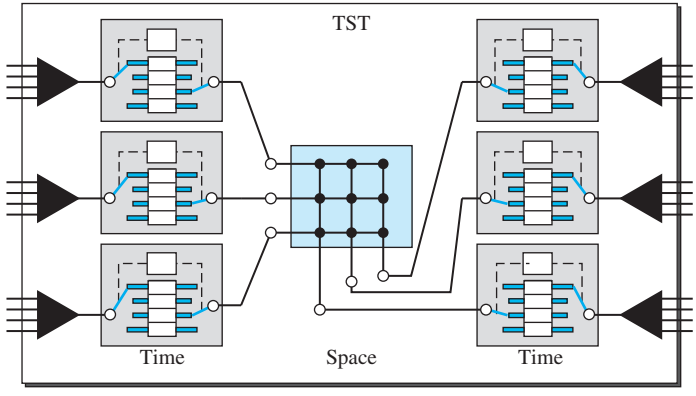
When we compare space-division and time-division switching, some interesting facts emerge. The advantage of space-division switching is that it is instantaneous. Its disadvantage is the number of crosspoints required to make space-division switching acceptable in terms of blocking.

The advantage of time-division switching is that it needs no crosspoints. Its disadvantage, in the case of TSI, is that processing each connection creates delays. Each time slot must be stored by the RAM, then retrieved and passed on.

In a third option, we combine space-division and time-division technologies to take advantage of the best of both. Combining the two results in switches that are optimized both physically (the number of crosspoints) and temporally (the amount of delay). Multistage switches of this sort can be designed as **time-space-time (TST) switches.**

Figure 8.20 shows a simple TST switch that consists of two time stages and one space stage and has 12 inputs and 12 outputs. Instead of one time-division switch, it divides the inputs into three groups (of four inputs each) and directs them to three time-slot interchanges. The result is that the average delay is one-third of what would result from using one time-slot interchange to handle all 12 inputs.

Figure 8.20 *Time-space-time switch*

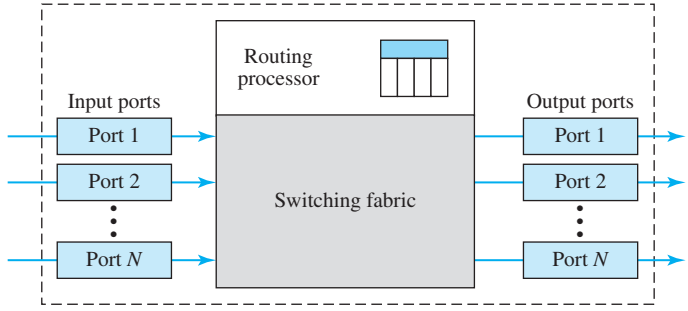


The last stage is a mirror image of the first stage. The middle stage is a space-division switch (crossbar) that connects the TSI groups to allow connectivity between all possible input and output pairs (e.g., to connect input 3 of the first group to output 7 of the second group).

8.4.2 Structure of Packet Switches

A switch used in a packet-switched network has a different structure from a switch used in a circuit-switched network.We can say that a packet switch has four components: **input ports, output ports,** the **routing processor,** and the **switching fabric,** as shown in Figure 8.21.

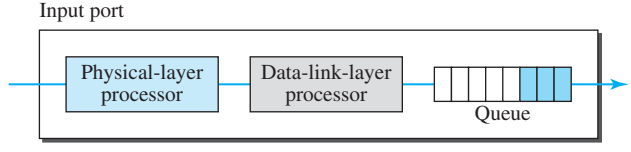
Figure 8.21 *Packet switch components*



Input Ports

An input port performs the physical and data-link functions of the packet switch. The bits are constructed from the received signal. The packet is decapsulated from the frame. Errors are detected and corrected. The packet is now ready to be routed by the network layer. In addition to a physical-layer processor and a data-link processor, the input port has buffers (queues) to hold the packet before it is directed to the switching fabric. Figure 8.22 shows a schematic diagram of an input port.

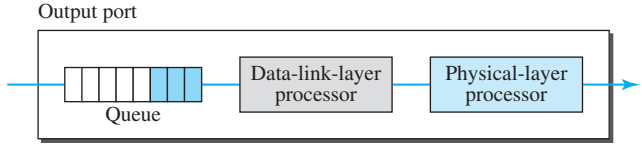
Figure 8.22 *Input port*



Output Port

The output port performs the same functions as the input port, but in the reverse order. First the outgoing packets are queued, then the packet is encapsulated in a frame, and finally the physical-layer functions are applied to the frame to create the signal to be sent on the line. Figure 8.23 shows a schematic diagram of an output port.

Figure 8.23 *Output port*



Routing Processor

The routing processor performs the functions of the network layer. The destination address is used to find the address of the next hop and, at the same time, the output port number from which the packet is sent out. This activity is sometimes referred to as **table lookup** because the routing processor searches the routing table. In the newer packet switches, this function of the routing processor is being moved to the input ports to facilitate and expedite the process.

Switching Fabrics

The most difficult task in a packet switch is to move the packet from the input queue to the output queue. The speed with which this is done affects the size of the input/output queue and the overall delay in packet delivery. In the past, when a packet switch was actually a dedicated computer, the memory of the computer or a bus was used as the switching fabric. The input port stored the packet in memory; the output port retrieved the packet from memory. Today, packet switches are specialized mechanisms that use a variety of switching fabrics. We briefly discuss some of these fabrics here.

Crossbar Switch

The simplest type of switching fabric is the crossbar switch, discussed in the previous section.

Banyan Switch

A more realistic approach than the crossbar switch is the **banyan switch** (named after the banyan tree). A banyan switch is a multistage switch with microswitches at each stage that route the packets based on the output port represented as a binary string. For n inputs and n outputs, we have $\log_2 n$ stages with $n/2$ microswitches at each stage. The first stage routes the packet based on the high-order bit of the binary string. The second stage routes the packet based on the second high-order bit, and so on. Figure 8.24 shows a banyan switch with eight inputs and eight outputs. The number of stages is $\log_2(8) = 3$.

Figure 8.24 *A banyan switch*

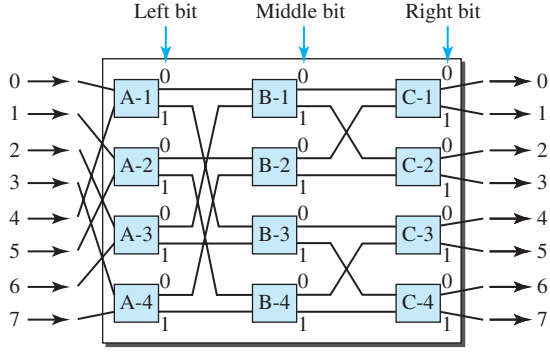
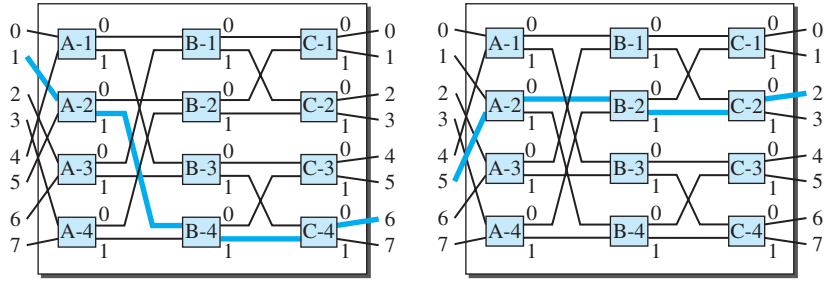


Figure 8.25 shows the operation. In part a, a packet has arrived at input port 1 and must go to output port 6 (110 in binary). The first microswitch (A-2) routes the packet

Figure 8.25 *Examples of routing in a banyan switch*



a. Input 1 sending a cell to output 6 (110)

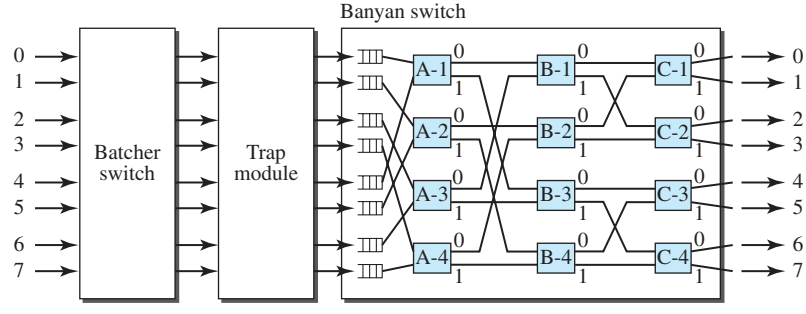
b. Input 5 sending a cell to output 2 (010)

based on the first bit (1), the second microswitch (B-4) routes the packet based on the second bit (1), and the third microswitch (C-4) routes the packet based on the third bit (0). In part b, a packet has arrived at input port 5 and must go to output port 2 (010 in binary). The first microswitch (A-2) routes the packet based on the first bit (0), the second microswitch (B-2) routes the packet based on the second bit (1), and the third microswitch (C-2) routes the packet based on the third bit (0).

Batcher-Banyan Switch The problem with the banyan switch is the possibility of internal collision even when two packets are not heading for the same output port. We can solve this problem by sorting the arriving packets based on their destination port.

K. E. Batcher designed a switch that comes before the banyan switch and sorts the incoming packets according to their final destinations. The combination is called the **Batcher-banyan switch.** The sorting switch uses hardware merging techniques, but we do not discuss the details here. Normally, another hardware module called a **trap** is added between the Batcher switch and the banyan switch (see Figure 8.26) The trap module prevents duplicate packets (the packets with the same output destination) from passing to the banyan switch simultaneously. Only one packet for each destination is allowed at each tick; if there is more than one, they wait for the next tick.

Figure 8.26 *Batcher-banyan switch*



8.5 END-CHAPTER MATERIALS

8.5.1 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

Switching is discussed in [Sta04] and [GW04]. Circuit-switching is fully discussed in [BEL01].

8.5.2 Key terms

banyan switch
Batcher-banyan switch
blocking
circuit switching
circuit-switched network
crossbar switch
crosspoint
data-transfer phase
datagram
datagram network
end system
input port
message switching
multistage switch
output port
packet switching
packet-switched network
routing processor
setup phase
space-division switching
switch
switching
switching fabric
table lookup
teardown phase
time-division switching
time-slot interchange (TSI)
time-space-time (TST) switch
trap
virtual-circuit identifier (VCI)
virtual-circuit network

8.5.3 Summary

A switched network consists of a series of interlinked nodes, called *switches*. Traditionally, three methods of switching have been important: circuit switching, packet switching, and message switching.

We can divide today's networks into three broad categories: circuit-switched networks, packet-switched networks, and message-switched networks. Packet-switched networks can also be divided into two subcategories: virtual-circuit networks and datagram networks. A circuit-switched network is made of a set of switches connected by physical links, in which each link is divided into n channels. Circuit switching takes place at the physical layer. In circuit switching, the resources need to be reserved during the setup phase; the resources remain dedicated for the entire duration of the data-transfer phase until the teardown phase.

In packet switching, there is no resource allocation for a packet. This means that there is no reserved bandwidth on the links, and there is no scheduled processing time for each packet. Resources are allocated on demand. In a datagram network, each packet is treated independently of all others. Packets in this approach are referred to as datagrams. There are no setup or teardown phases. A virtual-circuit network is a cross between a

circuit-switched network and a datagram network. It has some characteristics of both. Circuit switching uses either of two technologies: the space-division switch or the time-division switch. A switch in a packet-switched network has a different structure from a switch used in a circuit-switched network. We can say that a packet switch has four types of components: input ports, output ports, a routing processor, and switching fabric.

8.6 PRACTICE SET

8.6.1 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.

8.6.2 Questions

Q8-1. Describe the need for switching and define a switch.

Q8-2. List the three traditional switching methods. Which are the most common today?

Q8-3. What are the two approaches to packet switching?

Q8-4. Compare and contrast a circuit-switched network and a packet-switched network.

Q8-5. What is the role of the address field in a packet traveling through a datagram network?

Q8-6. What is the role of the address field in a packet traveling through a virtual-circuit network?

Q8-7. Compare space-division and time-division switches.

Q8-8. What is TSI and what is its role in time-division switching?

Q8-9. Compare and contrast the two major categories of circuit switches.

Q8-10. List four major components of a packet switch and their functions.

8.6.3 Problems

P8-1. A path in a digital circuit-switched network has a data rate of 1 Mbps. The exchange of 1000 bits is required for the setup and teardown phases. The distance between two parties is 5000 km. Answer the following questions if the propagation speed is 2×10^8 m:

- **a.** What is the total delay if 1000 bits of data are exchanged during the data-transfer phase?
- **b.** What is the total delay if 100,000 bits of data are exchanged during the data-transfer phase?
- **c.** What is the total delay if 1,000,000 bits of data are exchanged during the data-transfer phase?
- **d.** Find the delay per 1000 bits of data for each of the above cases and compare them. What can you infer?

P8-2. Five equal-size datagrams belonging to the same message leave for the destination one after another. However, they travel through different paths as shown in Table 8.1.

Table 8.1 *P8-2*

Datagram	*Path Length*	*Visited Switches*
1	3200 km	1, 3, 5
2	11,700 km	1, 2, 5
3	12,200 km	1, 2, 3, 5
4	10,200 km	1, 4, 5
5	10,700 km	1, 4, 3, 5

We assume that the delay for each switch (including waiting and processing) is 3, 10, 20, 7, and 20 ms respectively. Assuming that the propagation speed is 2×10^8 m, find the order the datagrams arrive at the destination and the delay for each. Ignore any other delays in transmission.

P8-3. Transmission of information in any network involves end-to-end addressing and sometimes local addressing (such as VCI). Table 8.2 shows the types of networks and the addressing mechanism used in each of them.

Table 8.2 *P8-3*

Network	*Setup*	*Data Transfer*	*Teardown*
Circuit-switched	End-to-end		End-to-end
Datagram		End-to-end	
Virtual-circuit	End-to-end	Local	End-to-end

Answer the following questions:

a. Why does a circuit-switched network need end-to-end addressing during the setup and teardown phases? Why are no addresses needed during the data transfer phase for this type of network?

b. Why does a datagram network need only end-to-end addressing during the data transfer phase, but no addressing during the setup and teardown phases?

c. Why does a virtual-circuit network need addresses during all three phases?

P8-4. We mentioned that two types of networks, datagram and virtual-circuit, need a routing or switching table to find the output port from which the information belonging to a destination should be sent out, but a circuit-switched network has no need for such a table. Give the reason for this difference.

P8-5. An entry in the switching table of a virtual-circuit network is normally created during the setup phase and deleted during the teardown phase. In other words, the entries in this type of network reflect the current connections, the activity in the network. In contrast, the entries in a routing table of a datagram network do not depend on the current connections; they show the configuration of the network and how any packet should be routed to a final destination. The entries may remain the same even if there is no activity in the network. The routing tables, however, are updated if there are changes in the network. Can you explain the reason for these two different characteristics? Can we say that

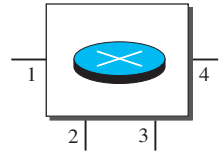
a virtual-circuit is a *connection-oriented* network and a datagram network is a *connectionless* network because of the above characteristics?

P8-6. The minimum number of columns in a datagram network is two; the minimum number of columns in a virtual-circuit network is four. Can you explain the reason? Is the difference related to the type of addresses carried in the packets of each network?

P8-7. Figure 8.27 shows a switch (router) in a datagram network.

Figure 8.27 *Problem P8-7*

Destination address	Output port
1233	3
1456	2
3255	1
4470	4
7176	2
8766	3
9144	2



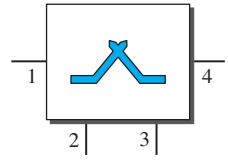
Find the output port for packets with the following destination addresses:

a. Packet 1: 7176
b. Packet 2: 1233
c. Packet 3: 8766
d. Packet 4: 9144

P8-8. Figure 8.28 shows a switch in a virtual-circuit network.

Figure 8.28 *Problem P8-8*

Incoming		Outgoing	
Port	VCI	Port	VCI
1	14	3	22
2	71	4	41
2	92	1	45
3	58	2	43
3	78	2	70
4	56	3	11



Find the output port and the output VCI for packets with the following input port and input VCI addresses:

a. Packet 1: 3, 78
b. Packet 2: 2, 92
c. Packet 3: 4, 56
d. Packet 4: 2, 71

P8-9. Answer the following questions:

a. Can a routing table in a datagram network have two entries with the same destination address? Explain.

b. Can a switching table in a virtual-circuit network have two entries with the same input port number? With the same output port number? With the same incoming VCIs? With the same outgoing VCIs? With the same incoming values (port, VCI)? With the same outgoing values (port, VCI)?

P8-10. It is obvious that a router or a switch needs to search to find information in the corresponding table. The searching in a routing table for a datagram network is based on the destination address; the searching in a switching table in a virtual-circuit network is based on the combination of incoming port and incoming VCI. Explain the reason and define how these tables must be ordered (sorted) based on these values.

P8-11. Consider an $n \times k$ crossbar switch with n inputs and k outputs.

- **a.** Can we say that the switch acts as a multiplexer if $n > k$?
- **b.** Can we say that the switch acts as a demultiplexer if $n < k$?

P8-12. We need a three-stage space-division switch with $N = 100$. We use 10 crossbars at the first and third stages and 4 crossbars at the middle stage.

- **a.** Draw the configuration diagram.
- **b.** Calculate the total number of crosspoints.
- **c.** Find the possible number of simultaneous connections.
- **d.** Find the possible number of simultaneous connections if we use a single crossbar (100×100).
- **e.** Find the blocking factor, the ratio of the number of connections in part c and in part d.

P8-13. Repeat Problem 8-12 if we use 6 crossbars at the middle stage.

P8-14. Redesign the configuration of Problem 8-12 using the Clos criteria.

P8-15. We need to have a space-division switch with 1000 inputs and outputs. What is the total number of crosspoints in each of the following cases?

- **a.** Using a single crossbar.
- **b.** Using a multi-stage switch based on the Clos criteria.

P8-16. We need a three-stage time-space-time switch with $N = 100$. We use 10 TSIs at the first and third stages and 4 crossbars at the middle stage.

- **a.** Draw the configuration diagram.
- **b.** Calculate the total number of crosspoints.
- **c.** Calculate the total number of memory locations we need for the TSIs.

8.7 SIMULATION EXPERIMENTS

8.7.1 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.

PART III

Data-Link Layer

In the third part of the book, we discuss the data-link layer in nine chapters. General topics are covered in Chapters 9 to 12. Wired networks are covered in Chapters 13 and 14. Wireless networks are covered in Chapters 15 and 16. Finally, we show how to connect LANs in Chapter 17.

CHAPTER 9

Introduction to Data-Link Layer

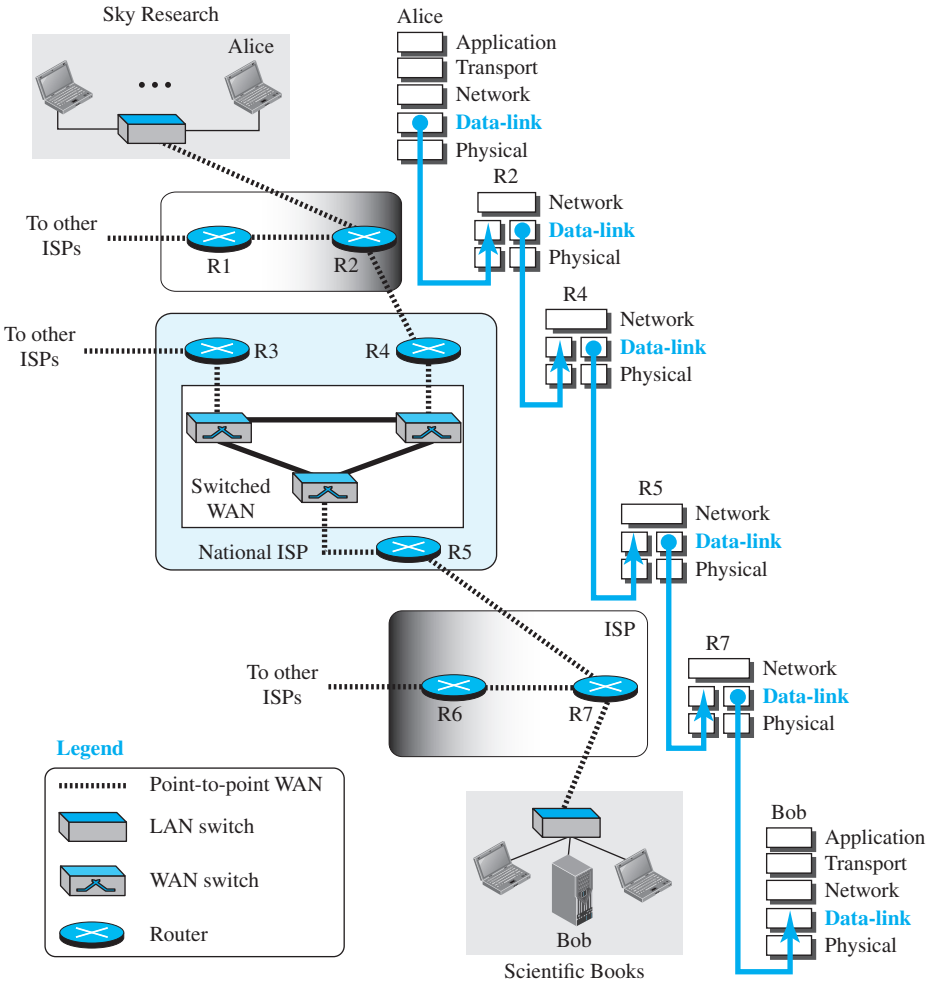
The TCP/IP protocol suite does not define any protocol in the data-link layer or physical layer. These two layers are territories of networks that when connected make up the Internet. These networks, wired or wireless, provide services to the upper three layers of the TCP/IP suite. This may give us a clue that there are several standard protocols in the market today. For this reason, we discuss the data-link layer in several chapters. This chapter is an introduction that gives the general idea and common issues in the data-link layer that relate to all networks.

- The first section introduces the data-link layer. It starts with defining the concept of links and nodes. The section then lists and briefly describes the services provided by the data-link layer. It next defines two categories of links: point-to-point and broadcast links. The section finally defines two sublayers at the data-link layer that will be elaborated on in the next few chapters.
- The second section discusses link-layer addressing. It first explains the rationale behind the existence of an addressing mechanism at the data-link layer. It then describes three types of link-layer addresses to be found in some link-layer protocols. The section discusses the Address Resolution Protocol (ARP), which maps the addresses at the network layer to addresses at the data-link layer. This protocol helps a packet at the network layer find the link-layer address of the next node for delivery of the frame that encapsulates the packet. To show how the network layer helps us to find the data-link-layer addresses, a long example is included in this section that shows what happens at each node when a packet is travelling through the Internet.

9.1 INTRODUCTION

The Internet is a combination of networks glued together by connecting devices (routers or switches). If a packet is to travel from a host to another host, it needs to pass through these networks. Figure 9.1 shows the same scenario we discussed in Chapter 3, but we are now interested in communication at the data-link layer. Communication at the data-link layer is made up of five separate logical connections between the data-link layers in the path.

Figure 9.1 *Communication at the data-link layer*



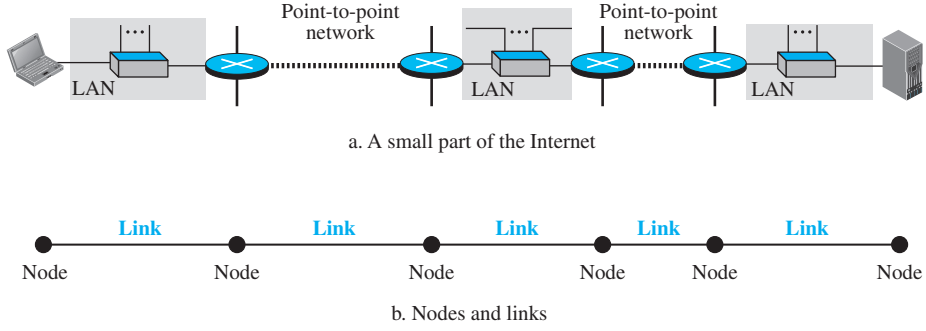
The data-link layer at Alice's computer communicates with the data-link layer at router R2. The data-link layer at router R2 communicates with the data-link layer at router R4,

and so on. Finally, the data-link layer at router R7 communicates with the data-link layer at Bob's computer. Only one data-link layer is involved at the source or the destination, but two data-link layers are involved at each router. The reason is that Alice's and Bob's computers are each connected to a single network, but each router takes input from one network and sends output to another network. Note that although switches are also involved in the data-link-layer communication, for simplicity we have not shown them in the figure.

9.1.1 Nodes and Links

Communication at the data-link layer is node-to-node. A data unit from one point in the Internet needs to pass through many networks (LANs and WANs) to reach another point. Theses LANs and WANs are connected by routers. It is customary to refer to the two end hosts and the routers as ***nodes*** and the networks in between as ***links*.** Figure 9.2 is a simple representation of links and nodes when the path of the data unit is only six nodes.

Figure 9.2 *Nodes and Links*



The first node is the source host; the last node is the destination host. The other four nodes are four routers. The first, the third, and the fifth links represent the three LANs; the second and the fourth links represent the two WANs.

9.1.2 Services

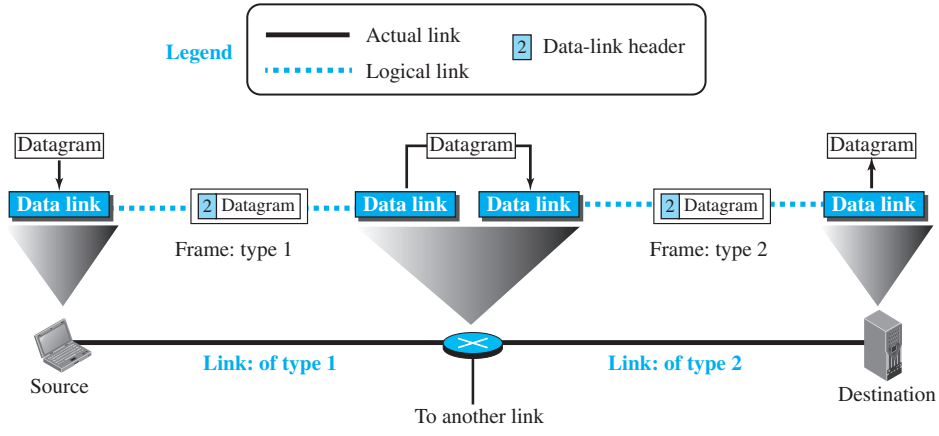
The data-link layer is located between the physical and the network layers. The data-link layer provides services to the network layer; it receives services from the physical layer. Let us discuss services provided by the data-link layer.

The duty scope of the data-link layer is node-to-node. When a packet is travelling in the Internet, the data-link layer of a node (host or router) is responsible for delivering a datagram to the next node in the path. For this purpose, the data-link layer of the sending node needs to encapsulate the datagram received from the network in a frame, and the data-link layer of the receiving node needs to decapsulate the datagram from the frame. In other words, the data-link layer of the source host needs only to

encapsulate, the data-link layer of the destination host needs to decapsulate, but each intermediate node needs to both encapsulate and decapsulate. One may ask why we need encapsulation and decapsulation at each intermediate node. The reason is that each link may be using a different protocol with a different frame format. Even if one link and the next are using the same protocol, encapsulation and decapsulation are needed because the link-layer addresses are normally different. An analogy may help in this case. Assume a person needs to travel from her home to her friend's home in another city. The traveller can use three transportation tools. She can take a taxi to go to the train station in her own city, then travel on the train from her own city to the city where her friend lives, and finally reach her friend's home using another taxi. Here we have a source node, a destination node, and two intermediate nodes. The traveller needs to get into the taxi at the source node, get out of the taxi and get into the train at the first intermediate node (train station in the city where she lives), get out of the train and get into another taxi at the second intermediate node (train station in the city where her friend lives), and finally get out of the taxi when she arrives at her destination. A kind of encapsulation occurs at the source node, encapsulation and decapsulation occur at the intermediate nodes, and decapsulation occurs at the destination node. Our traveller is the same, but she uses three transporting tools to reach the destination.

Figure 9.3 shows the encapsulation and decapsulation at the data-link layer. For simplicity, we have assumed that we have only one router between the source and destination. The datagram received by the data-link layer of the source host is encapsulated in a frame. The frame is logically transported from the source host to the router. The frame is decapsulated at the data-link layer of the router and encapsulated at another frame. The new frame is logically transported from the router to the destination host. Note that, although we have shown only two data-link layers at the router, the router actually has three data-link layers because it is connected to three physical links.

Figure 9.3 *A communication with only three nodes*



With the contents of the above figure in mind, we can list the services provided by a data-link layer as shown below.

Framing

Definitely, the first service provided by the data-link layer is **framing**. The data-link layer at each node needs to encapsulate the datagram (packet received from the network layer) in a **frame** before sending it to the next node. The node also needs to decapsulate the datagram from the frame received on the logical channel. Although we have shown only a header for a frame, we will see in future chapters that a frame may have both a header and a trailer. Different data-link layers have different formats for framing.

A packet at the data-link layer is normally called a *frame*.

Flow Control

Whenever we have a producer and a consumer, we need to think about flow control. If the producer produces items that cannot be consumed, accumulation of items occurs. The sending data-link layer at the end of a link is a producer of frames; the receiving data-link layer at the other end of a link is a consumer. If the rate of produced frames is higher than the rate of consumed frames, frames at the receiving end need to be buffered while waiting to be consumed (processed). Definitely, we cannot have an unlimited buffer size at the receiving side. We have two choices. The first choice is to let the receiving data-link layer drop the frames if its buffer is full. The second choice is to let the receiving data-link layer send a feedback to the sending data-link layer to ask it to stop or slow down. Different data-link-layer protocols use different strategies for flow control. Since flow control also occurs at the transport layer, with a higher degree of importance, we discuss this issue in Chapter 23 when we talk about the transport layer.

Error Control

At the sending node, a frame in a data-link layer needs to be changed to bits, transformed to electromagnetic signals, and transmitted through the transmission media. At the receiving node, electromagnetic signals are received, transformed to bits, and put together to create a frame. Since electromagnetic signals are susceptible to error, a frame is susceptible to error. The error needs first to be detected. After detection, it needs to be either corrected at the receiver node or discarded and retransmitted by the sending node. Since error detection and correction is an issue in every layer (node-to-node or host-to-host), we have dedicated all of Chapter 10 to this issue.

Congestion Control

Although a link may be congested with frames, which may result in frame loss, most data-link-layer protocols do not directly use a congestion control to alleviate congestion, although some wide-area networks do. In general, congestion control is considered an issue in the network layer or the transport layer because of its end-to-end nature. We will discuss congestion control in the network layer and the transport layer in later chapters.

9.1.3 Two Categories of Links

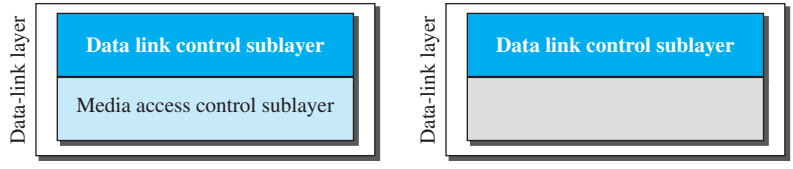
Although two nodes are physically connected by a transmission medium such as cable or air, we need to remember that the data-link layer controls how the medium is used. We can have a data-link layer that uses the whole capacity of the medium; we can also

have a data-link layer that uses only part of the capacity of the link. In other words, we can have a *point-to-point link* or a *broadcast link*. In a point-to-point link, the link is dedicated to the two devices; in a broadcast link, the link is shared between several pairs of devices. For example, when two friends use the traditional home phones to chat, they are using a point-to-point link; when the same two friends use their cellular phones, they are using a broadcast link (the air is shared among many cell phone users).

9.1.4 Two Sublayers

To better understand the functionality of and the services provided by the link layer, we can divide the data-link layer into two sublayers: **data link control (DLC)** and **media access control (MAC).** This is not unusual because, as we will see in later chapters, LAN protocols actually use the same strategy. The data link control sublayer deals with all issues common to both point-to-point and broadcast links; the media access control sublayer deals only with issues specific to broadcast links. In other words, we separate these two types of links at the data-link layer, as shown in Figure 9.4.

Figure 9.4 *Dividing the data-link layer into two sublayers*



a. Data-link layer of a broadcast link

b. Data-link layer of a point-to-point link

We discuss the DLC and MAC sublayers later, each in a separate chapter. In addition, we discuss the issue of error detection and correction, a duty of the data-link and other layers, also in a separate chapter.

9.2 LINK-LAYER ADDRESSING

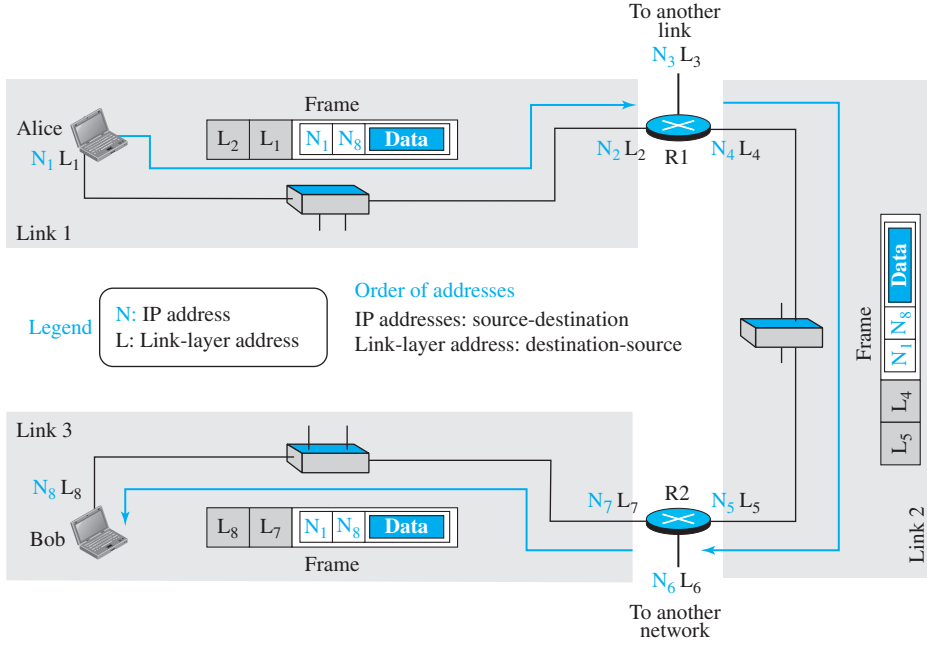
The next issue we need to discuss about the data-link layer is the link-layer addresses. In Chapter 18, we will discuss IP addresses as the identifiers at the network layer that define the exact points in the Internet where the source and destination hosts are connected. However, in a connectionless internetwork such as the Internet we cannot make a datagram reach its destination using only IP addresses. The reason is that each datagram in the Internet, from the same source host to the same destination host, may take a different path. The source and destination IP addresses define the two ends but cannot define which links the datagram should pass through.

We need to remember that the IP addresses in a datagram should not be changed. If the destination IP address in a datagram changes, the packet never reaches its destination; if the source IP address in a datagram changes, the destination host or a router can never communicate with the source if a response needs to be sent back or an error needs to be reported back to the source (see ICMP in Chapter 19).

The above discussion shows that we need another addressing mechanism in a connectionless internetwork: the link-layer addresses of the two nodes. A *link-layer address* is sometimes called a *link address*, sometimes a *physical address*, and sometimes a *MAC address*. We use these terms interchangeably in this book.

Since a link is controlled at the data-link layer, the addresses need to belong to the data-link layer. When a datagram passes from the network layer to the data-link layer, the datagram will be encapsulated in a frame and two data-link addresses are added to the frame header. These two addresses are changed every time the frame moves from one link to another. Figure 9.5 demonstrates the concept in a small internet.

Figure 9.5 *IP addresses and link-layer addresses in a small internet*



In the internet in Figure 9.5, we have three links and two routers. We also have shown only two hosts: Alice (source) and Bob (destination). For each host, we have shown two addresses, the IP addresses (N) and the link-layer addresses (L). Note that a router has as many pairs of addresses as the number of links the router is connected to. We have shown three frames, one in each link. Each frame carries the same datagram with the same source and destination addresses (**N1** and **N8**), but the link-layer addresses of the frame change from link to link. In link 1, the link-layer addresses are L_1 and L_2. In link 2, they are L_4 and L_5. In link 3, they are L_7 and L_8. Note that the IP addresses and the link-layer addresses are not in the same order. For IP addresses, the source address comes before the destination address; for link-layer addresses, the destination address comes before the source. The datagrams and

frames are designed in this way, and we follow the design. We may raise several questions:

- If the IP address of a router does not appear in any datagram sent from a source to a destination, why do we need to assign IP addresses to routers? The answer is that in some protocols a router may act as a sender or receiver of a datagram. For example, in routing protocols we will discuss in Chapters 20 and 21, a router is a sender or a receiver of a message. The communications in these protocols are between routers.
- Why do we need more than one IP address in a router, one for each interface? The answer is that an interface is a connection of a router to a link. We will see that an IP address defines a point in the Internet at which a device is connected. A router with *n* interfaces is connected to the Internet at *n* points. This is the situation of a house at the corner of a street with two gates; each gate has the address related to the corresponding street.
- How are the source and destination IP addresses in a packet determined? The answer is that the host should know its own IP address, which becomes the source IP address in the packet. As we will discuss in Chapter 26, the application layer uses the services of DNS to find the destination address of the packet and passes it to the network layer to be inserted in the packet.
- How are the source and destination link-layer addresses determined for each link? Again, each hop (router or host) should know its own link-layer address, as we discuss later in the chapter. The destination link-layer address is determined by using the Address Resolution Protocol, which we discuss shortly.
- What is the size of link-layer addresses? The answer is that it depends on the protocol used by the link. Although we have only one IP protocol for the whole Internet, we may be using different data-link protocols in different links. This means that we can define the size of the address when we discuss different link-layer protocols.

9.2.1 Three Types of addresses

Some link-layer protocols define three types of addresses: unicast, multicast, and broadcast.

Unicast Address

Each host or each interface of a router is assigned a unicast address. Unicasting means one-to-one communication. A frame with a unicast address destination is destined only for one entity in the link.

Example 9.1

As we will see in Chapter 13, the unicast link-layer addresses in the most common LAN, Ethernet, are 48 bits (six bytes) that are presented as 12 hexadecimal digits separated by colons; for example, the following is a link-layer address of a computer.

A3:34:45:11:92:F1

Multicast Address

Some link-layer protocols define multicast addresses. Multicasting means one-to-many communication. However, the jurisdiction is local (inside the link).

Example 9.2

As we will see in Chapter 13, the multicast link-layer addresses in the most common LAN, Ethernet, are 48 bits (six bytes) that are presented as 12 hexadecimal digits separated by colons. The second digit, however, needs to be an even number in hexadecimal. The following shows a multicast address:

A2:34:45:11:92:F1

Broadcast Address

Some link-layer protocols define a broadcast address. Broadcasting means one-to-all communication. A frame with a destination broadcast address is sent to all entities in the link.

Example 9.3

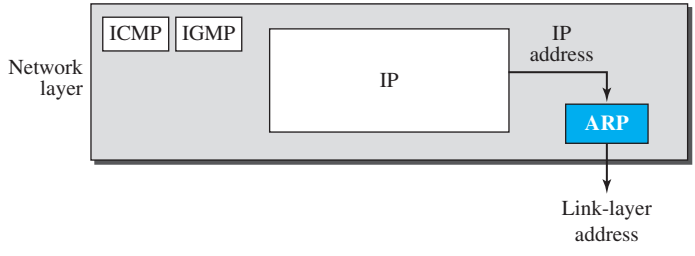
As we will see in Chapter 13, the broadcast link-layer addresses in the most common LAN, Ethernet, are 48 bits, all 1s, that are presented as 12 hexadecimal digits separated by colons. The following shows a broadcast address:

FF:FF:FF:FF:FF:FF

9.2.2 Address Resolution Protocol (ARP)

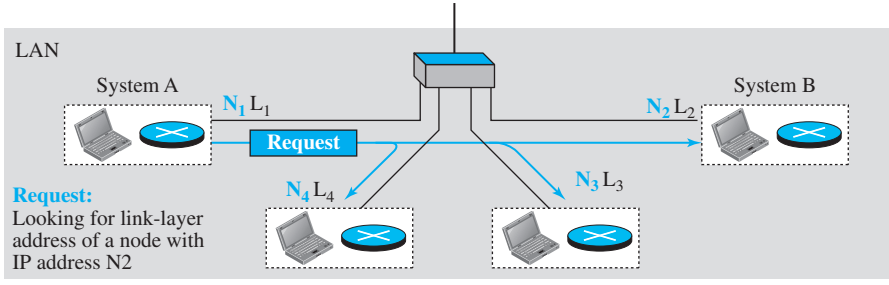
Anytime a node has an IP datagram to send to another node in a link, it has the IP address of the receiving node. The source host knows the IP address of the default router. Each router except the last one in the path gets the IP address of the next router by using its forwarding table. The last router knows the IP address of the destination host. However, the IP address of the next node is not helpful in moving a frame through a link; we need the link-layer address of the next node. This is the time when the **Address Resolution Protocol (ARP)** becomes helpful. The ARP protocol is one of the auxiliary protocols defined in the network layer, as shown in Figure 9.6. It belongs to the network layer, but we discuss it in this chapter because it maps an IP address to a logical-link address. ARP accepts an IP address from the IP protocol, maps the address to the corresponding link-layer address, and passes it to the data-link layer.

Figure 9.6 *Position of ARP in TCP/IP protocol suite*

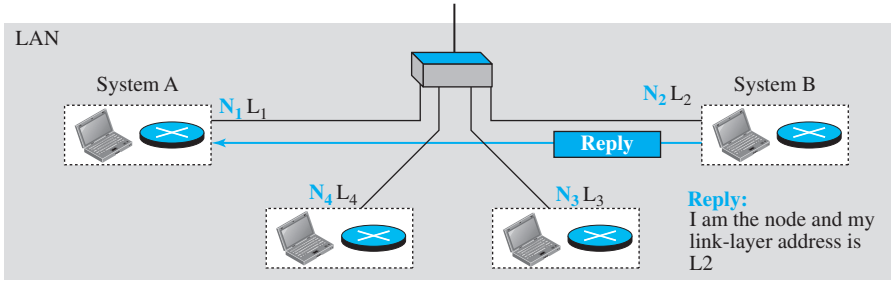


Anytime a host or a router needs to find the link-layer address of another host or router in its network, it sends an ARP request packet. The packet includes the link-layer and IP addresses of the sender and the IP address of the receiver. Because the sender does not know the link-layer address of the receiver, the query is broadcast over the link using the link-layer broadcast address, which we discuss for each protocol later (see Figure 9.7).

Figure 9.7 *ARP operation*



a. ARP request is broadcast



b. ARP reply is unicast

Every host or router on the network receives and processes the ARP request packet, but only the intended recipient recognizes its IP address and sends back an ARP response packet. The response packet contains the recipient's IP and link-layer addresses. The packet is unicast directly to the node that sent the request packet.

In Figure 9.7a, the system on the left (A) has a packet that needs to be delivered to another system (B) with IP address **N2**. System A needs to pass the packet to its data-link layer for the actual delivery, but it does not know the physical address of the recipient. It uses the services of ARP by asking the ARP protocol to send a broadcast ARP request packet to ask for the physical address of a system with an IP address of **N2**.

This packet is received by every system on the physical network, but only system B will answer it, as shown in Figure 9.7b. System B sends an ARP reply packet that includes its physical address. Now system A can send all the packets it has for this destination using the physical address it received.

Caching

A question that is often asked is this: If system A can broadcast a frame to find the link-layer address of system B, why can't system A send the datagram for system B using a broadcast frame? In other words, instead of sending one broadcast frame (ARP request), one unicast frame (ARP response), and another unicast frame (for sending the datagram), system A can encapsulate the datagram and send it to the network. System B receives it and keep it; other systems discard it.

To answer the question, we need to think about the efficiency. It is probable that system A has more than one datagram to send to system B in a short period of time. For example, if system B is supposed to receive a long e-mail or a long file, the data do not fit in one datagram.

Let us assume that there are 20 systems connected to the network (link): system A, system B, and 18 other systems. We also assume that system A has 10 datagrams to send to system B in one second.

a. Without using ARP, system A needs to send 10 broadcast frames. Each of the 18 other systems need to receive the frames, decapsulate the frames, remove the datagram and pass it to their network-layer to find out the datagrams do not belong to them.This means processing and discarding 180 broadcast frames.

b. Using ARP, system A needs to send only one broadcast frame. Each of the 18 other systems need to receive the frames, decapsulate the frames, remove the ARP message and pass the message to their ARP protocol to find that the frame must be discarded. This means processing and discarding only 18 (instead of 180) broadcast frames. After system B responds with its own data-link address, system A can store the link-layer address in its cache memory. The rest of the nine frames are only unicast. Since processing broadcast frames is expensive (time consuming), the first method is preferable.

Packet Format

Figure 9.8 shows the format of an ARP packet. The names of the fields are self-explanatory. The *hardware type* field defines the type of the link-layer protocol; Ethernet is given the type 1. The *protocol type* field defines the network-layer protocol: IPv4 protocol is $(0800)_{16}$. The source hardware and source protocol addresses are variable-length fields defining the link-layer and network-layer addresses of the sender. The destination hardware address and destination protocol address fields define the receiver link-layer and network-layer addresses. An ARP packet is encapsulated directly into a data-link frame. The frame needs to have a field to show that the payload belongs to the ARP and not to the network-layer datagram.

Example 9.4

A host with IP address **N1** and MAC address **L1** has a packet to send to another host with IP address **N2** and physical address **L2** (which is unknown to the first host). The two hosts are on the same network. Figure 9.9 shows the ARP request and response messages.

Figure 9.8 *ARP packet*

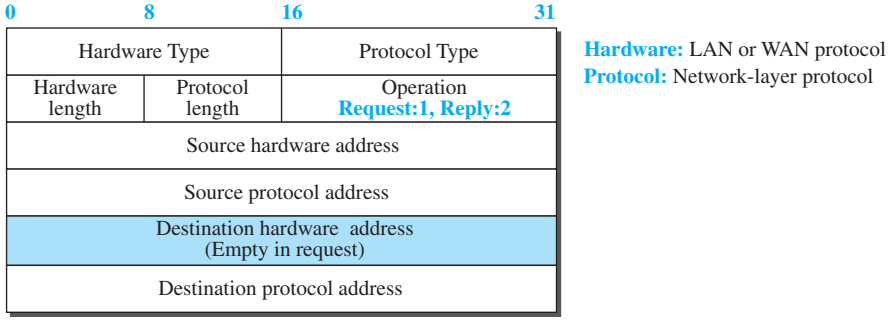
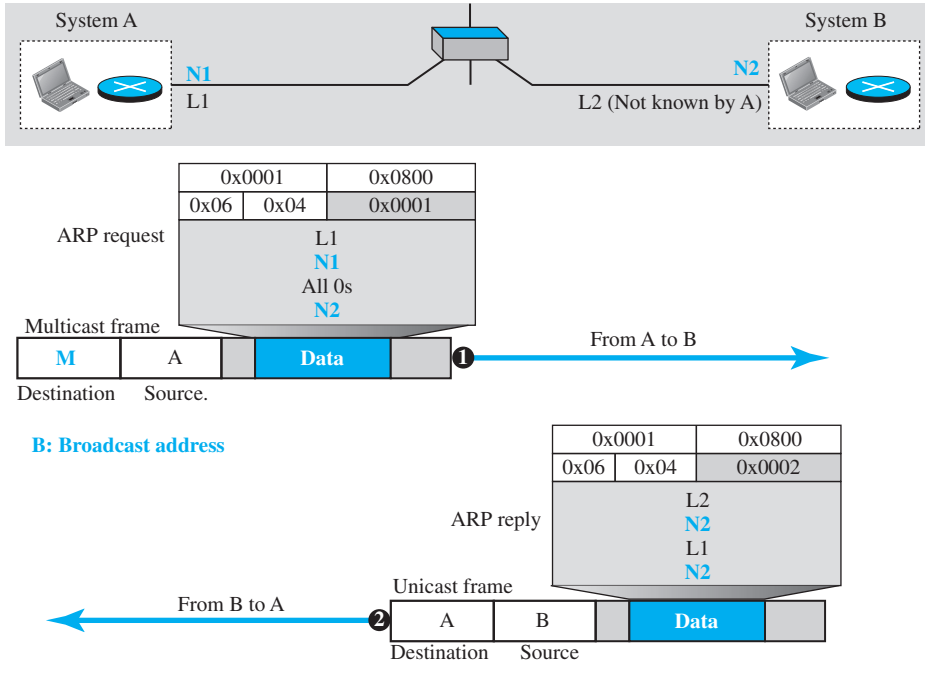


Figure 9.9 *Example 9.4*

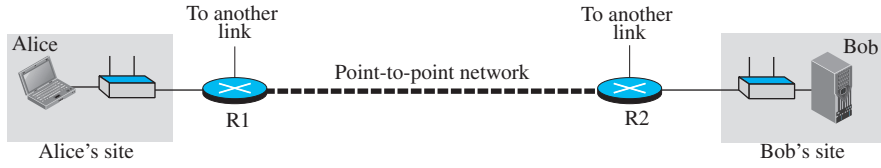


9.2.3 An Example of Communication

To show how communication is done at the data-link layer and how link-layer addresses are found, let us go through a simple example. Assume Alice needs to send a datagram to Bob, who is three nodes away in the Internet. How Alice finds the network-layer address of Bob is what we discover in Chapter 26 when we discuss DNS. For the moment, assume that Alice knows the network-layer (IP) address of Bob. In other words, Alice's host is given the data to be sent, the IP address of Bob, and the

IP address of Alice's host (each host needs to know its IP address). Figure 9.10 shows the part of the internet for our example.

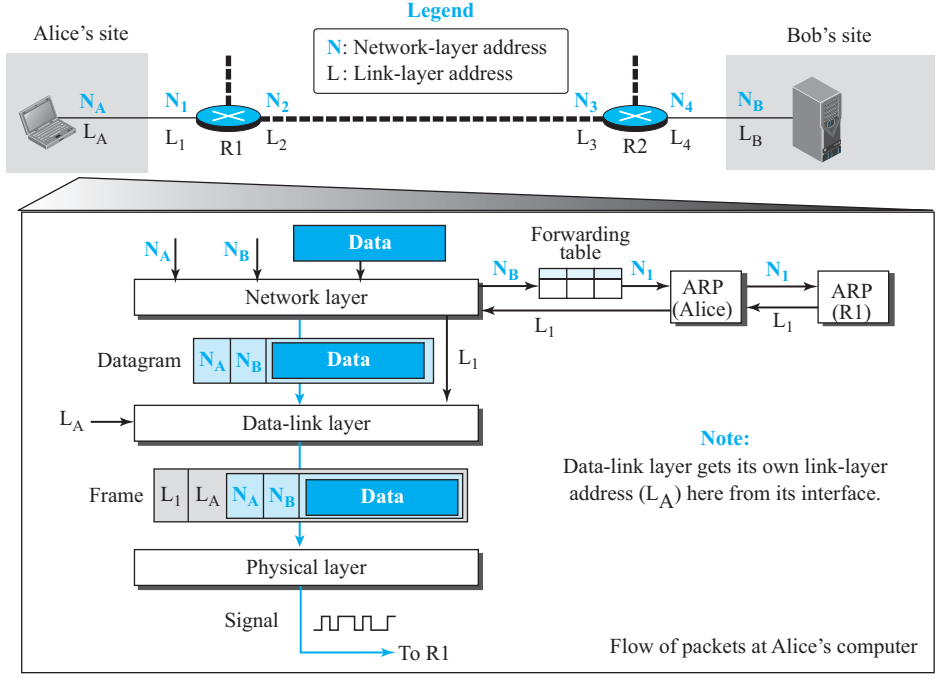
Figure 9.10 *The internet for our example*



Activities at Alice's Site

We will use symbolic addresses to make the figures more readable. Figure 9.11 shows what happens at Alice's site.

Figure 9.11 *Flow of packets at Alice's computer*



The network layer knows it's given N_A, N_B, and the packet, but it needs to find the link-layer address of the next node. The network layer consults its routing table and tries to find which router is next (the default router in this case) for the destination N_B. As we will discuss in Chapter 18, the routing table gives N_1, but the network layer

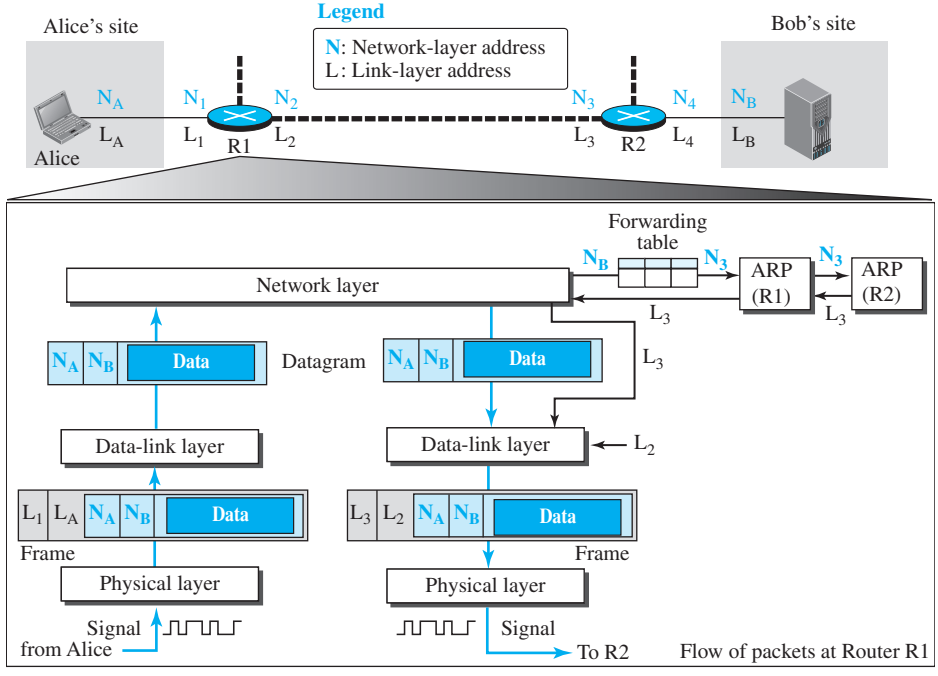
needs to find the link-layer address of router R1. It uses its ARP to find the link-layer address $\mathbf{L_1}$. The network layer can now pass the datagram with the link-layer address to the data-link layer.

The data-link layer knows its own link-layer address, $\mathbf{L_A}$. It creates the frame and passes it to the physical layer, where the address is converted to signals and sent through the media.

Activities at Router R1

Now let us see what happens at Router R1. Router R1, as we know, has only three lower layers. The packet received needs to go up through these three layers and come down. Figure 9.12 shows the activities.

Figure 9.12 *Flow of activities at router R1*



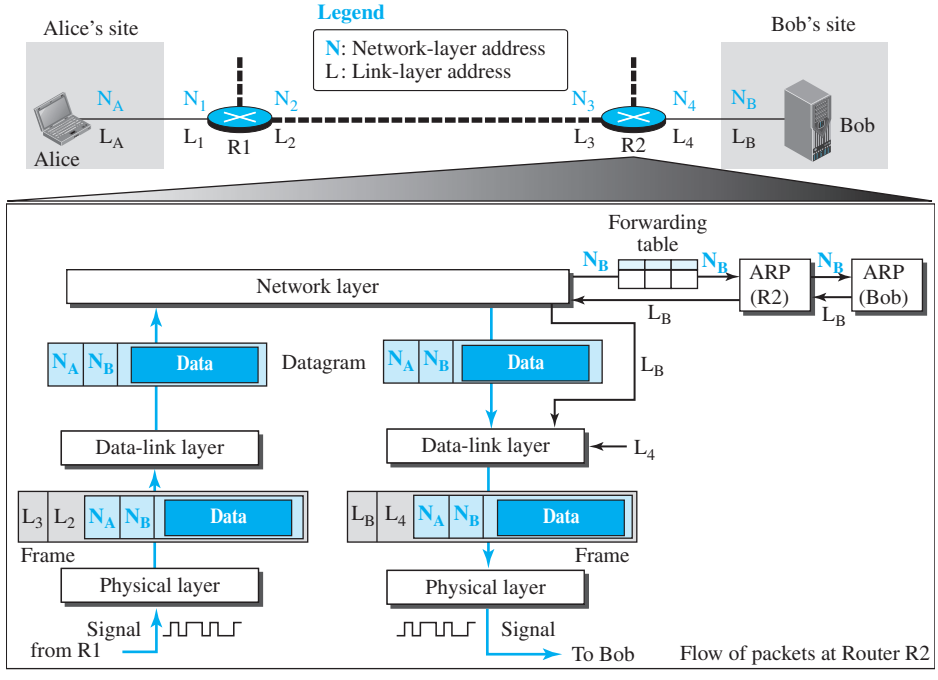
At arrival, the physical layer of the left link creates the frame and passes it to the data-link layer. The data-link layer decapsulates the datagram and passes it to the network layer. The network layer examines the network-layer address of the datagram and finds that the datagram needs to be delivered to the device with IP address $\mathbf{N_B}$. The network layer consults its routing table to find out which is the next node (router) in the path to $\mathbf{N_B}$. The forwarding table returns $\mathbf{N_3}$. The IP address of router R2 is in the same link with R1. The network layer now uses the ARP to find the link-layer address of this router, which comes up as $\mathbf{L_3}$. The network layer passes the datagram and $\mathbf{L_3}$ to the data-link layer belonging to the link at the right side. The link layer

encapsulates the datagram, adds **L3** and **L2** (its own link-layer address), and passes the frame to the physical layer. The physical layer encodes the bits to signals and sends them through the medium to R2.

Activities at Router R2

Activities at router R2 are almost the same as in R1, as shown in Figure 9.13.

Figure 9.13 *Activities at router R2.*



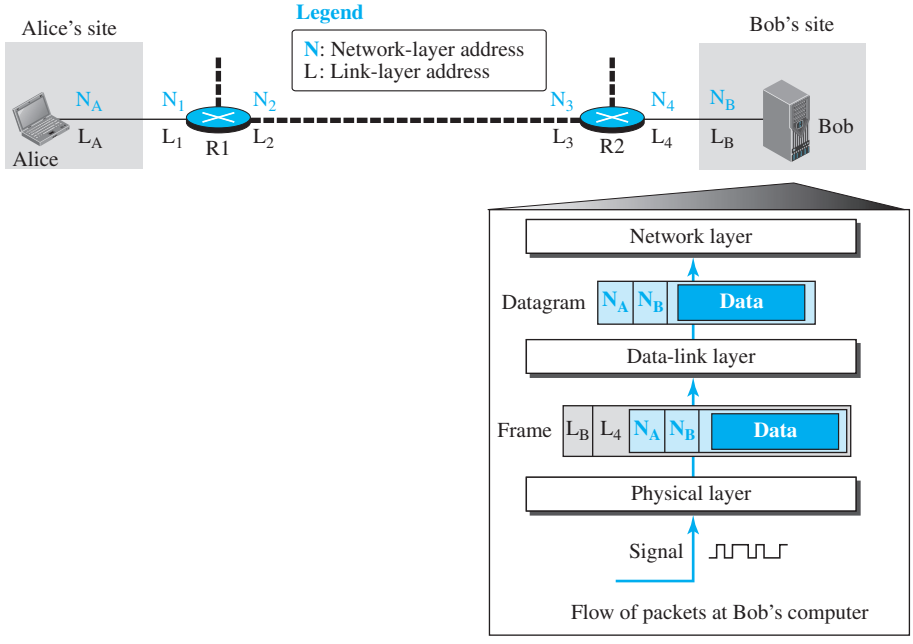
Activities at Bob's Site

Now let us see what happens at Bob's site. Figure 9.14 shows how the signals at Bob's site are changed to a message. At Bob's site there are no more addresses or mapping needed. The signal received from the link is changed to a frame. The frame is passed to the data-link layer, which decapsulates the datagram and passes it to the network layer. The network layer decapsulates the message and passes it to the transport layer.

Changes in Addresses

This example shows that the source and destination network-layer addresses, NA and NB, have not been changed during the whole journey. However, all four network-layer addresses of routers R1 and R2 (N1, N2, N3, and N4) are needed to transfer a datagram from Alice's computer to Bob's computer.

Figure 9.14 *Activities at Bob's site*



9.3 END-CHAPTER MATERIALS

9.3.1 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

Several books discuss link-layer issues. Among them we recommend [Ham 80], [Zar 02], [Ror 96], [Tan 03], [GW 04], [For 03], [KMK 04], [Sta 04], [Kes 02], [PD 03], [Kei 02], [Spu 00], [KCK 98], [Sau 98], [Izz 00], [Per 00], and [WV 00].

9.3.2 Key Terms

Address Resolution Protocol (ARP)
data link control (DLC)
frame
framing
links
media access control (MAC)
nodes

9.3.3 Summary

The Internet is made of many hosts, networks, and connecting devices such as routers. The hosts and connecting devices are referred to as *nodes;* the networks are referred to

as *links*. A path in the Internet from a source host to a destination host is a set of nodes and links through which a packet should travel.

The data-link layer is responsible for the creation and delivery of a frame to another node, along the link. It is responsible for packetizing (framing), flow control, error control, and congestion control along the link. Two data-link layers at the two ends of a link coordinate to deliver a frame from one node to the next.

As with any delivery between a source and destination in which there are many paths, we need two types of addressing. The end-to-end addressing defines the source and destination; the link-layer addressing defines the addresses of the nodes that the packet should pass through. To avoid including the link-layer addresses of all of these nodes in the frame, the Address Resolution Protocol (ARP) was devised to map an IP address to its corresponding link-layer address. When a packet is at one node ready to be sent to the next, the forwarding table finds the IP address of the next node and ARP find its link-layer address.

9.4 PRACTICE SET

9.4.1 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.

9.4.2 Questions

Q9-1. Distinguish between communication at the network layer and communication at the data-link layer.

Q9-2. Distinguish between a point-to-point link and a broadcast link.

Q9-3. Can two hosts in two different networks have the same link-layer address? Explain.

Q9-4. Is the size of the ARP packet fixed? Explain.

Q9-5. What is the size of an ARP packet when the protocol is IPv4 and the hardware is Ethernet?

Q9-6. Assume we have an isolated link (not connected to any other link) such as a private network in a company. Do we still need addresses in both the network layer and the data-link layer? Explain.

Q9-7. In Figure 9.9, why is the destination hardware address all 0s in the ARP request message?

Q9-8. In Figure 9.9, why is the destination hardware address of the frame from A to B a broadcast address?

Q9-9. In Figure 9.9, how does system A know what the link-layer address of system B is when it receives the ARP reply?

Q9-10. When we talkabout the broadcast address in a link, do we mean sending a message to all hosts and routers in the link or to all hosts and routers in the Internet? In other words, does a broadcast address have a local jurisdiction or a universal jurisdiction? Explain.

Q9-11. Why does a host or a router need to run the ARP program all of the time in the background?

Q9-12. Why does a router normally have more than one interface?

Q9-13. Why is it better not to change an end-to-end address from the source to the destination?

Q9-14. How many IP addresses and how many link-layer addresses should a router have when it is connected to five links?

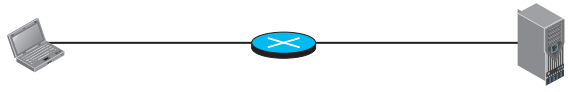
9.4.3 Problems

P9-1. Assume we have an internet (a private small internet) in which all hosts are connected in a mesh topology. Do we need routers in this internet? Explain.

P9-2. In the previous problem, do we need both network and data-link layers?

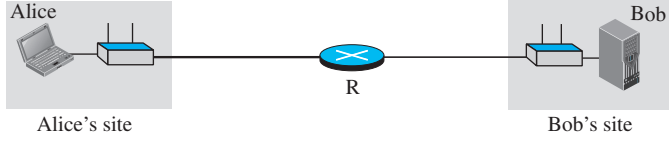
P9-3. Explain why we do not need the router in Figure 9.15.

Figure 9.15 *Problem 9-3*



P9-4. Explain why we may need a router in Figure 9.16.

Figure 9.16 *Problem 9-4*



P9-5. Is the current Internet using circuit-switching or packet-switching at the data-link layer? Explain.

P9-6. Assume Alice is travelling from 2020 Main Street in Los Angeles to 1432 American Boulevard in Chicago. If she is travelling by air from Los Angeles Airport to Chicago Airport,

a. find the end-to-end addresses in this scenario.

b. find the link-layer addresses in this scenario.

P9-7. In the previous problem, assume Alice cannot find a direct flight from the Los Angeles to the Chicago. If she needs to change flights in Denver,

a. find the end-to-end addresses in this scenario.

b. find the link-layer addresses in this scenario.

P9-8. When we send a letter using the services provided by the post office, do we use an end-to-end address? Does the post office necessarily use an end-to-end address to deliver the mail? Explain.

P9-9. In Figure 9.5, assume Link 2 is broken. How can Alice communicate with Bob?

P9-10. In Figure 9.5, show the process of frame change in routers R1 and R2.

P9-11. In Figure 9.7, assume system B is not running the ARP program. What would happen?

P9-12. In Figure 9.7, do you think that system A should first check its cache for mapping from N2 to L2 before even broadcasting the ARP request?

P9-13. Assume the network in Figure 9.7 does not support broadcasting. What do you suggest for sending the ARP request in this network?

P9-14. In Figures 9.11 to 9.13, both the forwarding table and ARP are doing a kind of mapping. Show the difference between them by listing the input and output of mapping for a forwarding table and ARP.

P9-15. Figure 9.7 shows a system as either a host or a router. What would be the actual entity (host or router) of system A and B in each of the following cases:

- **a.** If the link is the first one in the path?
- **b.** If the link is the middle one in the path?
- **c.** If the link is the last one in the path?
- **d.** If there is only one link in the path (local communication)?

CHAPTER 10

Error Detection and Correction

Networks must be able to transfer data from one device to another with acceptable accuracy. For most applications, a system must guarantee that the data received are identical to the data transmitted. Any time data are transmitted from one node to the next, they can become corrupted in passage. Many factors can alter one or more bits of a message. Some applications require a mechanism for detecting and correcting **errors.**

Some applications can tolerate a small level of error. For example, random errors in audio or video transmissions may be tolerable, but when we transfer text, we expect a very high level of accuracy.

At the data-link layer, if a frame is corrupted between the two nodes, it needs to be corrected before it continues its journey to other nodes. However, most link-layer protocols simply discard the frame and let the upper-layer protocols handle the retransmission of the frame. Some multimedia applications, however, try to correct the corrupted frame.

This chapter is divided into five sections.

- The first section introduces types of errors, the concept of redundancy, and distinguishes between error detection and correction.
- The second section discusses block coding. It shows how error can be detected using block coding and also introduces the concept of Hamming distance.
- The third section discusses cyclic codes. It discusses a subset of cyclic code, CRC, that is very common in the data-link layer. The section shows how CRC can be easily implemented in hardware and represented by polynomials.
- The fourth section discusses checksums. It shows how a checksum is calculated for a set of data words. It also gives some other approaches to traditional checksum.
- The fifth section discusses forward error correction. It shows how Hamming distance can also be used for this purpose. The section also describes cheaper methods to achieve the same goal, such as XORing of packets, interleaving chunks, or compounding high and low resolutions packets.

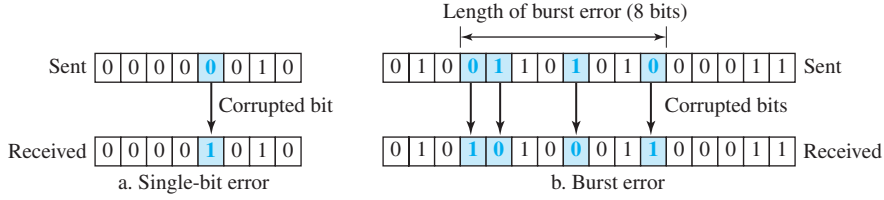
10.1 INTRODUCTION

Let us first discuss some issues related, directly or indirectly, to error detection and correction.

10.1.1 Types of Errors

Whenever bits flow from one point to another, they are subject to unpredictable changes because of **interference.** This interference can change the shape of the signal. The term ***single-bit error*** means that only 1 bit of a given data unit (such as a byte, character, or packet) is changed from 1 to 0 or from 0 to 1. The term ***burst error*** means that 2 or more bits in the data unit have changed from 1 to 0 or from 0 to 1. Figure 10.1 shows the effect of a single-bit and a burst error on a data unit.

Figure 10.1 *Single-bit and burst error*



A burst error is more likely to occur than a single-bit error because the duration of the noise signal is normally longer than the duration of 1 bit, which means that when noise affects data, it affects a set of bits. The number of bits affected depends on the data rate and duration of noise. For example, if we are sending data at 1 kbps, a noise of 1/100 second can affect 10 bits; if we are sending data at 1 Mbps, the same noise can affect 10,000 bits.

10.1.2 Redundancy

The central concept in detecting or correcting errors is **redundancy**. To be able to detect or correct errors, we need to send some extra bits with our data. These redundant bits are added by the sender and removed by the receiver. Their presence allows the receiver to detect or correct corrupted bits.

10.1.3 Detection versus Correction

The correction of errors is more difficult than the detection. In **error detection**, we are only looking to see if any error has occurred. The answer is a simple yes or no. We are not even interested in the number of corrupted bits. A single-bit error is the same for us as a burst error. In **error correction**, we need to know the exact number of bits that are corrupted and, more importantly, their location in the message. The number of errors and the size of the message are important factors. If we need to correct a single error in an 8-bit data unit, we need to consider eight possible error locations; if we need to correct two

errors in a data unit of the same size, we need to consider 28 (permutation of 8 by 2) possibilities. You can imagine the receiver's difficulty in finding 10 errors in a data unit of 1000 bits.

10.1.4 Coding

Redundancy is achieved through various coding schemes. The sender adds redundant bits through a process that creates a relationship between the redundant bits and the actual data bits. The receiver checks the relationships between the two sets of bits to detect errors. The ratio of redundant bits to data bits and the robustness of the process are important factors in any coding scheme.

We can divide coding schemes into two broad categories: **block coding** and **convolution coding**. In this book, we concentrate on block coding; convolution coding is more complex and beyond the scope of this book.

10.2 BLOCK CODING

In block coding, we divide our message into blocks, each of k bits, called ***datawords***. We add r redundant bits to each block to make the length $n = k + r$. The resulting n-bit blocks are called ***codewords***. How the extra r bits are chosen or calculated is something we will discuss later. For the moment, it is important to know that we have a set of datawords, each of size k, and a set of codewords, each of size of n. With k bits, we can create a combination of 2^k datawords; with n bits, we can create a combination of 2^n codewords. Since $n > k$, the number of possible codewords is larger than the number of possible datawords. The block coding process is one-to-one; the same dataword is always encoded as the same codeword. This means that we have $2^n - 2^k$ codewords that are not used. We call these codewords invalid or illegal. The trick in error detection is the existence of these invalid codes, as we discuss next. If the receiver receives an invalid codeword, this indicates that the data was corrupted during transmission.

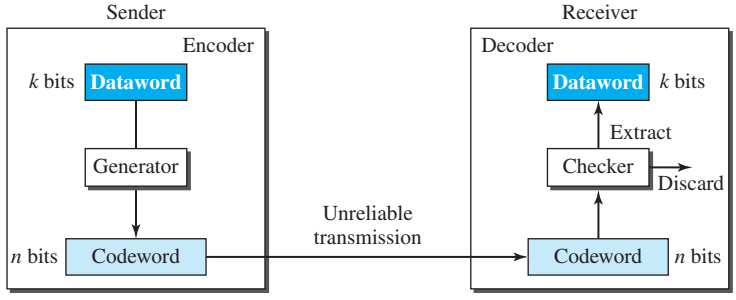
10.2.1 Error Detection

How can errors be detected by using block coding? If the following two conditions are met, the receiver can detect a change in the original codeword.

1. The receiver has (or can find) a list of valid codewords.
2. The original codeword has changed to an invalid one.

Figure 10.2 shows the role of block coding in error detection. The sender creates codewords out of datawords by using a generator that applies the rules and procedures of encoding (discussed later). Each codeword sent to the receiver may change during transmission. If the received codeword is the same as one of the valid codewords, the word is accepted; the corresponding dataword is extracted for use. If the received codeword is not valid, it is discarded. However, if the codeword is corrupted during transmission but the received word still matches a valid codeword, the error remains undetected.

Figure 10.2 *Process of error detection in block coding*



Example 10.1

Let us assume that $k = 2$ and $n = 3$. Table 10.1 shows the list of datawords and codewords. Later, we will see how to derive a codeword from a dataword.

Table 10.1 *A code for error detection in Example 10.1*

Dataword	*Codeword*	*Dataword*	*Codeword*
00	000	10	101
01	011	11	110

Assume the sender encodes the dataword 01 as 011 and sends it to the receiver. Consider the following cases:

1. The receiver receives 011. It is a valid codeword. The receiver extracts the dataword 01 from it.
2. The codeword is corrupted during transmission, and 111 is received (the leftmost bit is corrupted). This is not a valid codeword and is discarded.
3. The codeword is corrupted during transmission, and 000 is received (the right two bits are corrupted). This is a valid codeword. The receiver incorrectly extracts the dataword 00. Two corrupted bits have made the error undetectable.

An error-detecting code can detect only the types of errors for which it is designed; other types of errors may remain undetected.

Hamming Distance

One of the central concepts in coding for error control is the idea of the Hamming distance. The **Hamming distance** between two words (of the same size) is the number of differences between the corresponding bits. We show the Hamming distance between two words x and y as $\boldsymbol{d}(x, y)$. We may wonder why Hamming distance is important for error detection. The reason is that the Hamming distance between the received codeword and the sent codeword is the number of bits that are corrupted during transmission. For example, if the codeword 00000 is sent and 01101 is received, 3 bits are in error and the Hamming distance between the two is $\boldsymbol{d}(00000, 01101) = 3$. In other words, if the Hamming

distance between the sent and the received codeword is not zero, the codeword has been corrupted during transmission.

The Hamming distance can easily be found if we apply the XOR operation ($\oplus$) on the two words and count the number of 1s in the result. Note that the Hamming distance is a value greater than or equal to zero.

The Hamming distance between two words is the number of differences between corresponding bits.

Example 10.2

Let us find the Hamming distance between two pairs of words.

1. The Hamming distance $\boldsymbol{d}(000, 011)$ is 2 because $(000 \oplus 011)$ is 011 (two 1s).
2. The Hamming distance $\boldsymbol{d}(10101, 11110)$ is 3 because $(10101 \oplus 11110)$ is 01011 (three 1s).

Minimum Hamming Distance for Error Detection

In a set of codewords, the **minimum Hamming distance** is the smallest Hamming distance between all possible pairs of codewords. Now let us find the minimum Hamming distance in a code if we want to be able to detect up to s errors. If s errors occur during transmission, the Hamming distance between the sent codeword and received codeword is s. If our system is to detect up to s errors, the minimum distance between the valid codes must be $(s + 1)$, so that the received codeword does not match a valid codeword. In other words, if the minimum distance between all valid codewords is $(s + 1)$, the received codeword cannot be erroneously mistaken for another codeword. The error will be detected. We need to clarify a point here: Although a code with $d_{min} = s + 1$ may be able to detect more than s errors in some special cases, only s or fewer errors are guaranteed to be detected.

To guarantee the detection of up to s errors in all cases, the minimum Hamming distance in a block code must be $d_{min} = s + 1$.

We can look at this criteria geometrically. Let us assume that the sent codeword x is at the center of a circle with radius s. All received codewords that are created by 0 to s errors are points inside the circle or on the perimeter of the circle. All other valid codewords must be outside the circle, as shown in Figure 10.3. This means that d_{min} must be an integer greater than s or $d_{min} = s + 1$.

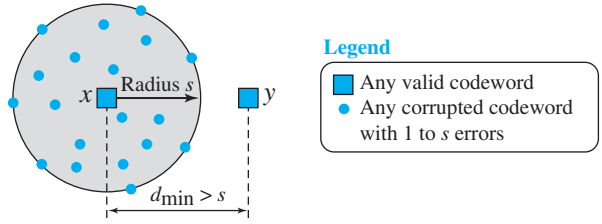
Example 10.3

The minimum Hamming distance for our first code scheme (Table 10.1) is 2. This code guarantees detection of only a single error. For example, if the third codeword (101) is sent and one error occurs, the received codeword does not match any valid codeword. If two errors occur, however, the received codeword may match a valid codeword and the errors are not detected.

Example 10.4

A code scheme has a Hamming distance $d_{min} = 4$. This code guarantees the detection of up to three errors ($d = s + 1$ or $s = 3$).

Figure 10.3 *Geometric concept explaining* d_{min} *in error detection*



Linear Block Codes

Almost all block codes used today belong to a subset of block codes called ***linear block codes***. The use of nonlinear block codes for error detection and correction is not as widespread because their structure makes theoretical analysis and implementation difficult. We therefore concentrate on linear block codes. The formal definition of linear block codes requires the knowledge of abstract algebra (particularly Galois fields), which is beyond the scope of this book. We therefore give an informal definition. For our purposes, a linear block code is a code in which the exclusive OR (addition modulo-2) of two valid codewords creates another valid codeword.

Example 10.5

The code in Table 10.1 is a linear block code because the result of XORing any codeword with any other codeword is a valid codeword. For example, the XORing of the second and third codewords creates the fourth one.

Minimum Distance for Linear Block Codes

It is simple to find the minimum Hamming distance for a linear block code. The minimum Hamming distance is the number of 1s in the nonzero valid codeword with the smallest number of 1s.

Example 10.6

In our first code (Table 10.1), the numbers of 1s in the nonzero codewords are 2, 2, and 2. So the minimum Hamming distance is $d_{\text{min}} = 2$.

Parity-Check Code

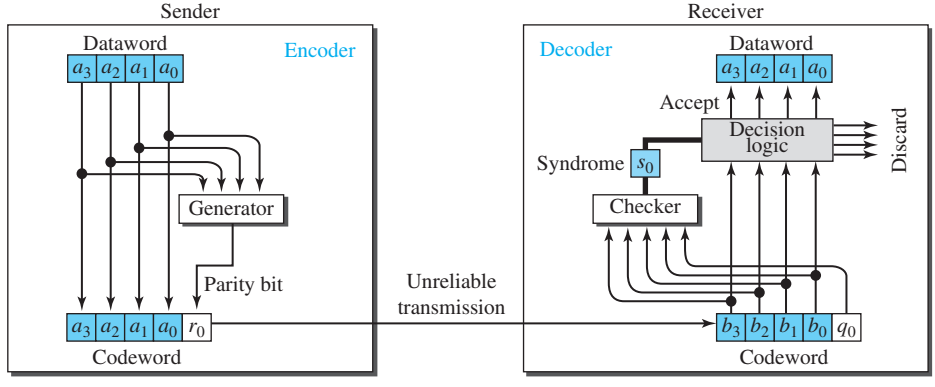
Perhaps the most familiar error-detecting code is the **parity-check code.** This code is a linear block code. In this code, a k-bit dataword is changed to an n-bit codeword where $n = k + 1$. The extra bit, called the *parity bit,* is selected to make the total number of 1s in the codeword even. Although some implementations specify an odd number of 1s, we discuss the even case. The minimum Hamming distance for this category is $d_{\text{min}} = 2$, which means that the code is a single-bit error-detecting code. Our first code (Table 10.1) is a parity-check code ($k = 2$ and $n = 3$). The code in Table 10.2 is also a parity-check code with $k = 4$ and $n = 5$.

Table 10.2 *Simple parity-check code C(5, 4)*

Dataword	*Codeword*	*Dataword*	*Codeword*
0000	**00000**	1000	**10001**
0001	**00011**	1001	**10010**
0010	**00101**	1010	**10100**
0011	**00110**	1011	**10111**
0100	**01001**	1100	**11000**
0101	**01010**	1101	**11011**
0110	**01100**	1110	**11101**
0111	**01111**	1111	**11110**

Figure 10.4 shows a possible structure of an encoder (at the sender) and a decoder (at the receiver).

Figure 10.4 *Encoder and decoder for simple parity-check code*



The calculation is done in **modular arithmetic** (see Appendix E). The encoder uses a generator that takes a copy of a 4-bit dataword (a_0, a_1, a_2, and a_3) and generates a parity bit r_0. The dataword bits and the parity bit create the 5-bit codeword. The parity bit that is added makes the number of 1s in the codeword even. This is normally done by adding the 4 bits of the dataword (modulo-2); the result is the parity bit. In other words,

$$r_0 = a_3 + a_2 + a_1 + a_0 \qquad \textbf{(modulo-2)}$$

If the number of 1s is even, the result is 0; if the number of 1s is odd, the result is 1. In both cases, the total number of 1s in the codeword is even.

The sender sends the codeword, which may be corrupted during transmission. The receiver receives a 5-bit word. The checker at the receiver does the same thing as the generator in the sender with one exception: The addition is done over all 5 bits. The result,

which is called the ***syndrome***, is just 1 bit. The syndrome is 0 when the number of 1s in the received codeword is even; otherwise, it is 1.

$$s_0 = b_3 + b_2 + b_1 + b_0 + q_0 \quad \textbf{(modulo-2)}$$

The syndrome is passed to the decision logic analyzer. If the syndrome is 0, there is no detectable error in the received codeword; the data portion of the received codeword is accepted as the dataword; if the syndrome is 1, the data portion of the received codeword is discarded. The dataword is not created.

Example 10.7

Let us look at some transmission scenarios. Assume the sender sends the dataword 1011. The codeword created from this dataword is 10111, which is sent to the receiver. We examine five cases:

1. No error occurs; the received codeword is 10111. The syndrome is 0. The dataword 1011 is created.
2. One single-bit error changes a_1. The received codeword is 10**0**11. The syndrome is 1. No dataword is created.
3. One single-bit error changes r_0. The received codeword is 1011**0**. The syndrome is 1. No dataword is created. Note that although none of the dataword bits are corrupted, no dataword is created because the code is not sophisticated enough to show the position of the corrupted bit.
4. An error changes r_0 and a second error changes a_3. The received codeword is **0**0110. The syndrome is 0. The dataword 0011 is created at the receiver. Note that here the dataword is wrongly created due to the syndrome value. The simple parity-check decoder cannot detect an even number of errors. The errors cancel each other out and give the syndrome a value of 0.
5. Three bits—a_3, a_2, and a_1—are changed by errors. The received codeword is **010**11. The syndrome is 1. The dataword is not created. This shows that the simple parity check, guaranteed to detect one single error, can also find any odd number of errors.

A parity-check code can detect an odd number of errors.

10.3 CYCLIC CODES

Cyclic codes are special linear block codes with one extra property. In a **cyclic code,** if a codeword is cyclically shifted (rotated), the result is another codeword. For example, if 1011000 is a codeword and we cyclically left-shift, then 0110001 is also a codeword. In this case, if we call the bits in the first word a_0 to a_6, and the bits in the second word b_0 to b_6, we can shift the bits by using the following:

$$b_1 = a_0 \qquad b_2 = a_1 \qquad b_3 = a_2 \qquad b_4 = a_3 \qquad b_5 = a_4 \qquad b_6 = a_5 \qquad b_0 = a_6$$

In the rightmost equation, the last bit of the first word is wrapped around and becomes the first bit of the second word.

10.3.1 Cyclic Redundancy Check

We can create cyclic codes to correct errors. However, the theoretical background required is beyond the scope of this book. In this section, we simply discuss a subset of

cyclic codes called the **cyclic redundancy check (CRC)**, which is used in networks such as LANs and WANs.

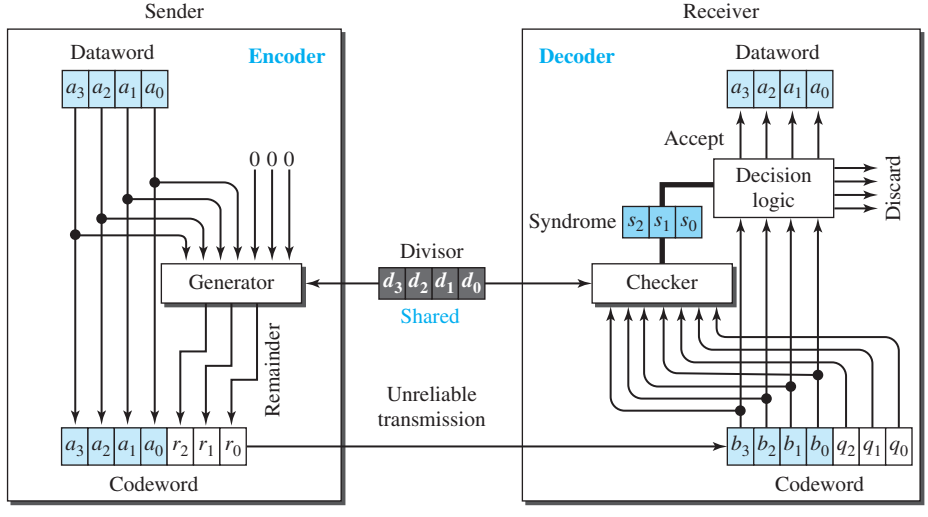
Table 10.3 shows an example of a CRC code. We can see both the linear and cyclic properties of this code.

Table 10.3 *A CRC code with C(7, 4)*

Dataword	*Codeword*	*Dataword*	*Codeword*
0000	0000000	1000	1000101
0001	0001011	1001	1001110
0010	0010110	1010	1010011
0011	0011101	1011	1011000
0100	0100111	1100	1100010
0101	0101100	1101	1101001
0110	0110001	1110	1110100
0111	0111010	1111	1111111

Figure 10.5 shows one possible design for the encoder and decoder.

Figure 10.5 *CRC encoder and decoder*



In the encoder, the dataword has k bits (4 here); the codeword has n bits (7 here). The size of the dataword is augmented by adding $n - k$ (3 here) 0s to the right-hand side of the word. The n-bit result is fed into the generator. The generator uses a divisor of size $n - k + 1$ (4 here), predefined and agreed upon. The generator divides the augmented dataword by the divisor (modulo-2 division). The quotient of the division is discarded; the remainder ($r_2 r_1 r_0$) is appended to the dataword to create the codeword.

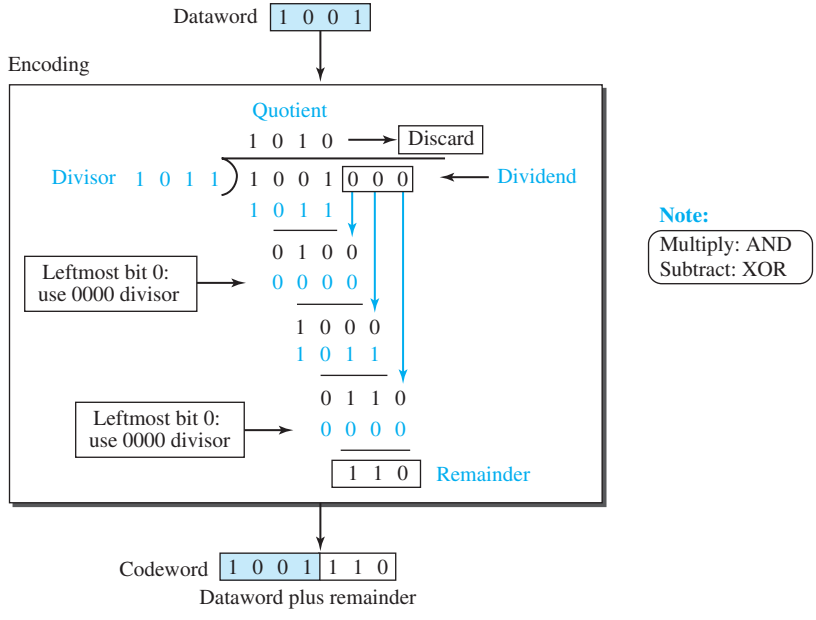
The decoder receives the codeword (possibly corrupted in transition). A copy of all n bits is fed to the checker, which is a replica of the generator. The remainder produced

by the checker is a syndrome of $n - k$ (3 here) bits, which is fed to the decision logic analyzer. The analyzer has a simple function. If the syndrome bits are all 0s, the 4 leftmost bits of the codeword are accepted as the dataword (interpreted as no error); otherwise, the 4 bits are discarded (error).

Encoder

Let us take a closer look at the encoder. The encoder takes a dataword and augments it with $n - k$ number of 0s. It then divides the augmented dataword by the divisor, as shown in Figure 10.6.

Figure 10.6 *Division in CRC encoder*



The process of modulo-2 binary division is the same as the familiar division process we use for decimal numbers. However, addition and subtraction in this case are the same; we use the XOR operation to do both.

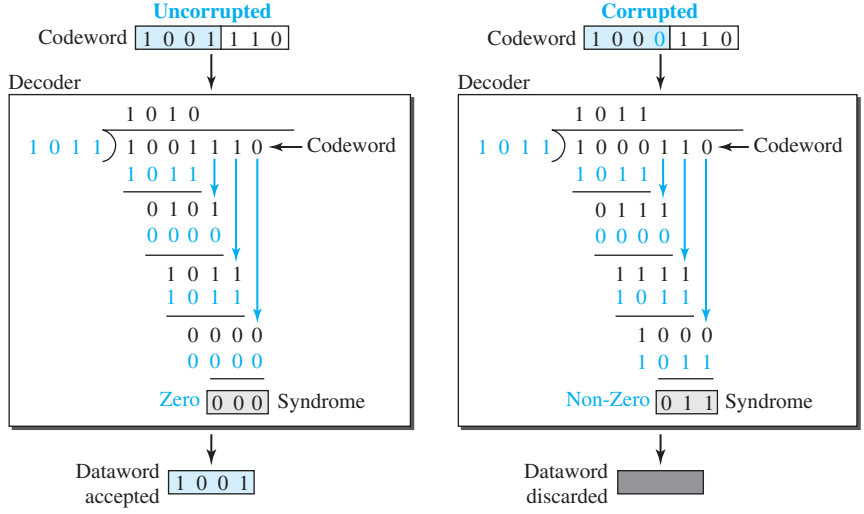
As in decimal division, the process is done step by step. In each step, a copy of the divisor is XORed with the 4 bits of the dividend. The result of the XOR operation (remainder) is 3 bits (in this case), which is used for the next step after 1 extra bit is pulled down to make it 4 bits long. There is one important point we need to remember in this type of division. If the leftmost bit of the dividend (or the part used in each step) is 0, the step cannot use the regular divisor; we need to use an all-0s divisor.

When there are no bits left to pull down, we have a result. The 3-bit remainder forms the **check bits** (r_2, r_1, and r_0). They are appended to the dataword to create the codeword.

Decoder

The codeword can change during transmission. The decoder does the same division process as the encoder. The remainder of the division is the syndrome. If the syndrome is all 0s, there is no error with a high probability; the dataword is separated from the received codeword and accepted. Otherwise, everything is discarded. Figure 10.7 shows two cases: The left-hand figure shows the value of the syndrome when no error has occurred; the syndrome is 000. The right-hand part of the figure shows the case in which there is a single error. The syndrome is not all 0s (it is 011).

Figure 10.7 *Division in the CRC decoder for two cases*



Divisor

We may be wondering how the divisor 1011 is chosen. This depends on the expectation we have from the code. We will show some standard divisors later in the chapter (Table 10.4) after we discuss polynomials.

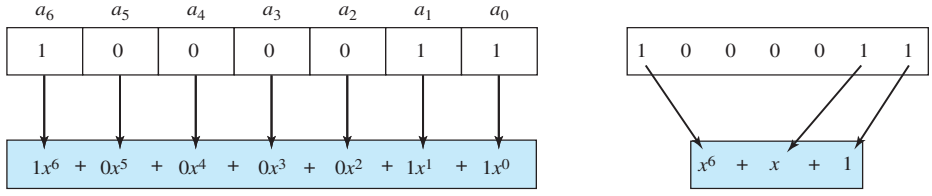
10.3.2 Polynomials

A better way to understand cyclic codes and how they can be analyzed is to represent them as polynomials. Again, this section is optional.

A pattern of 0s and 1s can be represented as a **polynomial** with coefficients of 0 and 1. The power of each term shows the position of the bit; the coefficient shows the value of the bit. Figure 10.8 shows a binary pattern and its polynomial representation. In Figure 10.8a we show how to translate a binary pattern into a polynomial; in Figure 10.8b we show how the polynomial can be shortened by removing all terms with zero coefficients and replacing x^1 by x and x^0 by 1.

Figure 10.8 shows one immediate benefit; a 7-bit pattern can be replaced by three terms. The benefit is even more conspicuous when we have a polynomial such as

Figure 10.8 *A polynomial to represent a binary word*



a. Binary pattern and polynomial

b. Short form

$x^{23} + x^3 + 1$. Here the bit pattern is 24 bits in length (three 1s and twenty-one 0s) while the polynomial is just three terms.

Degree of a Polynomial

The degree of a polynomial is the highest power in the polynomial. For example, the degree of the polynomial $x^6 + x + 1$ is 6. Note that the degree of a polynomial is 1 less than the number of bits in the pattern. The bit pattern in this case has 7 bits.

Adding and Subtracting Polynomials

Adding and subtracting polynomials in mathematics are done by adding or subtracting the coefficients of terms with the same power. In our case, the coefficients are only 0 and 1, and adding is in modulo-2. This has two consequences. First, addition and subtraction are the same. Second, adding or subtracting is done by combining terms and deleting pairs of identical terms. For example, adding $x^5 + x^4 + x^2$ and $x^6 + x^4 + x^2$ gives just $x^6 + x^5$. The terms x^4 and x^2 are deleted. However, note that if we add, for example, three polynomials and we get x^2 three times, we delete a pair of them and keep the third.

Multiplying or Dividing Terms

In this arithmetic, multiplying a term by another term is very simple; we just add the powers. For example, $x^3 \times x^4$ is x^7. For dividing, we just subtract the power of the second term from the power of the first. For example, x^5/x^2 is x^3.

Multiplying Two Polynomials

Multiplying a polynomial by another is done term by term. Each term of the first polynomial must be multiplied by all terms of the second. The result, of course, is then simplified, and pairs of equal terms are deleted. The following is an example:

$$\begin{aligned}(x^5 + x^3 + x^2 + x)(x^2 + x + 1) &= x^7 + x^6 + x^5 + x^5 + x^4 + x^3 + x^4 + x^3 + x^2 + x^3 + x^2 + x \\ &= x^7 + x^6 + x^3 + x\end{aligned}$$

Dividing One Polynomial by Another

Division of polynomials is conceptually the same as the binary division we discussed for an encoder. We divide the first term of the dividend by the first term of the divisor to get the first term of the quotient. We multiply the term in the quotient by the divisor and

subtract the result from the dividend. We repeat the process until the dividend degree is less than the divisor degree. We will show an example of division later in this chapter.

Shifting

A binary pattern is often shifted a number of bits to the right or left. Shifting to the left means adding extra 0s as rightmost bits; shifting to the right means deleting some rightmost bits. Shifting to the left is accomplished by multiplying each term of the polynomial by x^m, where m is the number of shifted bits; shifting to the right is accomplished by dividing each term of the polynomial by x^m. The following shows shifting to the left and to the right. Note that we do not have negative powers in the polynomial representation.

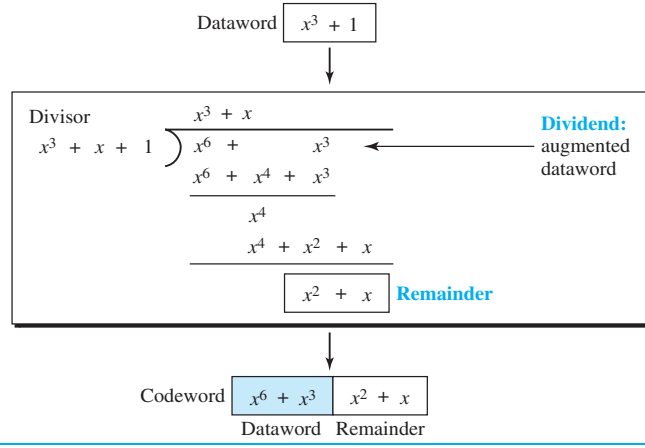
Shifting left 3 bits: 10011 becomes 10011000	$x^4 + x + 1$ becomes $x^7 + x^4 + x^3$
Shifting right 3 bits: 10011 becomes 10	$x^4 + x + 1$ becomes x

When we augmented the dataword in the encoder of Figure 10.6, we actually shifted the bits to the left. Also note that when we concatenate two bit patterns, we shift the first polynomial to the left and then add the second polynomial.

10.3.3 Cyclic Code Encoder Using Polynomials

Now that we have discussed operations on polynomials, we show the creation of a codeword from a dataword. Figure 10.9 is the polynomial version of Figure 10.6. We can see that the process is shorter. The dataword 1001 is represented as $x^3 + 1$. The divisor 1011 is represented as $x^3 + x + 1$. To find the augmented dataword, we have left-shifted the dataword 3 bits (multiplying by x^3). The result is $x^6 + x^3$. Division is straightforward. We divide the first term of the dividend, x^6, by the first term of the divisor, x^3. The first term of the quotient is then x^6/x^3, or x^3. Then we multiply x^3 by the divisor and subtract (according to our previous definition of subtraction) the result from the dividend. The result is x^4, with a degree greater than the divisor's degree; we continue to divide until the degree of the remainder is less than the degree of the divisor.

Figure 10.9 *CRC division using polynomials*



It can be seen that the polynomial representation can easily simplify the operation of division in this case, because the two steps involving all-0s divisors are not needed here. (Of course, one could argue that the all-0s divisor step can also be eliminated in binary division.) In a polynomial representation, the divisor is normally referred to as the ***generator polynomial*** $t(x)$.

The divisor in a cyclic code is normally called the *generator polynomial* or simply the *generator.*

10.3.4 Cyclic Code Analysis

We can analyze a cyclic code to find its capabilities by using polynomials. We define the following, where $f(x)$ is a polynomial with binary coefficients.

Dataword: $d(x)$ **Codeword: $c(x)$** **Generator: $g(x)$** **Syndrome: $s(x)$** **Error: $e(x)$**

If $s(x)$ is not zero, then one or more bits is corrupted. However, if $s(x)$ is zero, either no bit is corrupted or the decoder failed to detect any errors. (Note that ¦ means divide).

In a cyclic code,

1. If $s(x) \nmid 0$, one or more bits is corrupted.
2. If $s(x) = 0$, either
   a. No bit is corrupted, or
   b. Some bits are corrupted, but the decoder failed to detect them.

In our analysis we want to find the criteria that must be imposed on the generator, $g(x)$ to detect the type of error we especially want to be detected. Let us first find the relationship among the sent codeword, error, received codeword, and the generator. We can say

$$\textbf{Received codeword} = c(x) + e(x)$$

In other words, the received codeword is the sum of the sent codeword and the error. The receiver divides the received codeword by $g(x)$ to get the syndrome. We can write this as

$$\frac{\textbf{Received codeword}}{g(x)} = \frac{c(x)}{g(x)} + \frac{e(x)}{g(x)}$$

The first term at the right-hand side of the equality has a remainder of zero (according to the definition of codeword). So the syndrome is actually the remainder of the second term on the right-hand side. If this term does not have a remainder (syndrome = 0), either $e(x)$ is 0 or $e(x)$ is divisible by $g(x)$. We do not have to worry about the first case (there is no error); the second case is very important. Those errors that are divisible by $g(x)$ are not caught.

In a cyclic code, those $e(x)$ errors that are divisible by $g(x)$ are not caught.

Let us show some specific errors and see how they can be caught by a well-designed $g(x)$.

Single-Bit Error

What should the structure of $g(x)$ be to guarantee the detection of a single-bit error? A single-bit error is $e(x) = x^i$, where i is the position of the bit. If a single-bit error is caught, then x^i is not divisible by $g(x)$. (Note that when we say *not divisible,* we mean that there is a remainder.) If $g(x)$ has at least two terms (which is normally the case) and the coefficient of x^0 is not zero (the rightmost bit is 1), then $e(x)$ cannot be divided by $g(x)$.

If the generator has more than one term and the coefficient of x^0 is 1, all single-bit errors can be caught.

Example 10.8

Which of the following $g(x)$ values guarantees that a single-bit error is caught? For each case, what is the error that cannot be caught?

a. $x + 1$
b. x^3
c. 1

Solution

a. No x^i can be divisible by $x + 1$. In other words, $x^i/(x + 1)$ always has a remainder. So the syndrome is nonzero. Any single-bit error can be caught.

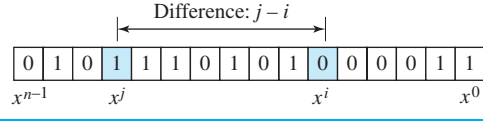
b. If i is equal to or greater than 3, x^i is divisible by $g(x)$. The remainder of x^i/x^3 is zero, and the receiver is fooled into believing that there is no error, although there might be one. Note that in this case, the corrupted bit must be in position 4 or above. All single-bit errors in positions 1 to 3 are caught.

c. All values of i make x^i divisible by $g(x)$. No single-bit error can be caught. In addition, this $g(x)$ is useless because it means the codeword is just the dataword augmented with $n - k$ zeros.

Two Isolated Single-Bit Errors

Now imagine there are two single-bit isolated errors. Under what conditions can this type of error be caught? We can show this type of error as $e(x) = x^j + x^i$. The values of i and j define the positions of the errors, and the difference $j - i$ defines the distance between the two errors, as shown in Figure 10.10.

Figure 10.10 *Representation of two isolated single-bit errors using polynomials*



We can write $e(x) = x^i(x^{j-i} + 1)$. If $g(x)$ has more than one term and one term is x^0, it cannot divide x^i, as we saw in the previous section. So if $g(x)$ is to divide $e(x)$, it must divide $x^{j-i} + 1$. In other words, $g(x)$ must not divide $x^t + 1$, where t is between 0 and $n - 1$. However, $t = 0$ is meaningless and $t = 1$ is needed, as we will see later. This means t should be between 2 and $n - 1$.

If a generator cannot divide $x^t + 1$ (t between 0 and $n - 1$), then all isolated double errors can be detected.

Example 10.9

Find the status of the following generators related to two isolated, single-bit errors.

a. $x + 1$
b. $x^4 + 1$
c. $x^7 + x^6 + 1$
d. $x^{15} + x^{14} + 1$

Solution

a. This is a very poor choice for a generator. Any two errors next to each other cannot be detected.
b. This generator cannot detect two errors that are four positions apart. The two errors can be anywhere, but if their distance is 4, they remain undetected.
c. This is a good choice for this purpose.
d. This polynomial cannot divide any error of type $x^t + 1$ if t is less than 32,768. This means that a codeword with two isolated errors that are next to each other or up to 32,768 bits apart can be detected by this generator.

Odd Numbers of Errors

A generator with a factor of $x + 1$ can catch all odd numbers of errors. This means that we need to make $x + 1$ a factor of any generator. Note that we are not saying that the generator itself should be $x + 1$; we are saying that it should have a factor of $x + 1$. If it is only $x + 1$, it cannot catch the two adjacent isolated errors (see the previous section). For example, $x^4 + x^2 + x + 1$ can catch all odd-numbered errors since it can be written as a product of the two polynomials $x + 1$ and $x^3 + x^2 + 1$.

A generator that contains a factor of $x + 1$ can detect all odd-numbered errors.

Burst Errors

Now let us extend our analysis to the burst error, which is the most important of all. A burst error is of the form $e(x) = (x^j + \cdots + x^i)$. Note the difference between a burst error and two isolated single-bit errors. The first can have two terms or more; the second can only have two terms. We can factor out x^i and write the error as $x^i(x^{j-i} + \cdots + 1)$. If our generator can detect a single error (minimum condition for a generator), then it cannot divide x^i. What we should worry about are those generators that divide $x^{j-i} + \cdots + 1$. In other words, the remainder of $(x^{j-i} + \cdots + 1)/(x^r + \cdots + 1)$ must not be zero. Note that the denominator is the generator polynomial. We can have three cases:

1. If $j - i < r$, the remainder can never be zero. We can write $j - i = L - 1$, where L is the length of the error. So $L - 1 < r$ or $L < r + 1$ or L ð r. This means all burst errors with length smaller than or equal to the number of check bits r will be detected.
2. In some rare cases, if $j - i = r$, or $L = r + 1$, the syndrome is 0 and the error is undetected. It can be proved that in these cases, the probability of undetected burst error of length $r + 1$ is $(1/2)^{r-1}$. For example, if our generator is $x^{14} + x^3 + 1$, in which $r = 14$, a burst error of length $L = 15$ can slip by undetected with the probability of $(1/2)^{14-1}$ or almost 1 in 10,000.
3. In some rare cases, if $j - i > r$, or $L > r + 1$, the syndrome is 0 and the error is undetected. It can be proved that in these cases, the probability of undetected burst error of length greater than $r + 1$ is $(1/2)^r$. For example, if our generator is $x^{14} + x^3 + 1$, in which $r = 14$, a burst error of length greater than 15 can slip by undetected with the probability of $(1/2)^{14}$ or almost 1 in 16,000 cases.

- **All burst errors with $L \leq r$ will be detected.**
- **All burst errors with $L = r + 1$ will be detected with probability $1 - (1/2)^{r-1}$.**
- **All burst errors with $L > r + 1$ will be detected with probability $1 - (1/2)^r$.**

Example 10.10

Find the suitability of the following generators in relation to burst errors of different lengths.

a. $x^6 + 1$
b. $x^{18} + x^7 + x + 1$
c. $x^{32} + x^{23} + x^7 + 1$

Solution

a. This generator can detect all burst errors with a length less than or equal to 6 bits; 3 out of 100 burst errors with length 7 will slip by; 16 out of 1000 burst errors of length 8 or more will slip by.
b. This generator can detect all burst errors with a length less than or equal to 18 bits; 8 out of 1 million burst errors with length 19 will slip by; 4 out of 1 million burst errors of length 20 or more will slip by.
c. This generator can detect all burst errors with a length less than or equal to 32 bits; 5 out of 10 billion burst errors with length 33 will slip by; 3 out of 10 billion burst errors of length 34 or more will slip by.

Summary

We can summarize the criteria for a good polynomial generator:

A good polynomial generator needs to have the following characteristics:

1. **It should have at least two terms.**
2. **The coefficient of the term x^0 should be 1.**
3. **It should not divide $x^t + 1$, for t between 2 and $n - 1$.**
4. **It should have the factor $x + 1$.**

Standard Polynomials

Some standard polynomials used by popular protocols for CRC generation are shown in Table 10.4 along with the corresponding bit pattern.

Table 10.4 *Standard polynomials*

Name	*Polynomial*	*Used in*
CRC-8	$x^8 + x^2 + x + 1$ **100000111**	ATM header
CRC-10	$x^{10} + x^9 + x^5 + x^4 + x^2 + 1$ **11000110101**	ATM AAL
CRC-16	$x^{16} + x^{12} + x^5 + 1$ **10001000000100001**	HDLC
CRC-32	$x^{32} + x^{26} + x^{23} + x^{22} + x^{16} + x^{12} + x^{11} + x^{10} + x^8 + x^7 + x^5 + x^4 + x^2 + x + 1$ **10000010011000001000111011011 0111**	LANs

10.3.5 Advantages of Cyclic Codes

We have seen that cyclic codes have a very good performance in detecting single-bit errors, double errors, an odd number of errors, and burst errors. They can easily be implemented in hardware and software. They are especially fast when implemented in hardware. This has made cyclic codes a good candidate for many networks.

10.3.6 Other Cyclic Codes

The cyclic codes we have discussed in this section are very simple. The check bits and syndromes can be calculated by simple algebra. There are, however, more powerful polynomials that are based on abstract algebra involving Galois fields. These are beyond the scope of this book. One of the most interesting of these codes is the **Reed-Solomon code** used today for both detection and correction.

10.3.7 Hardware Implementation

One of the advantages of a cyclic code is that the encoder and decoder can easily and cheaply be implemented in hardware by using a handful of electronic devices. Also, a hardware implementation increases the rate of check bit and syndrome bit calculation. In this section, we try to show, step by step, the process. The section, however, is optional and does not affect the understanding of the rest of the chapter.

Divisor

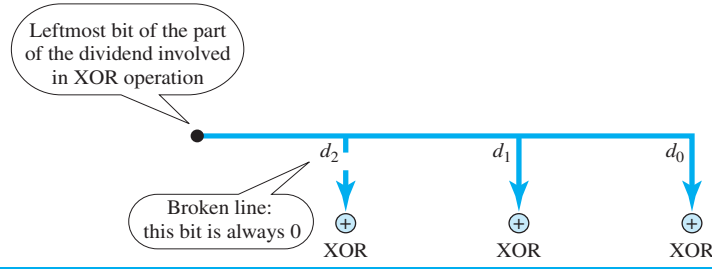
Let us first consider the divisor. We need to note the following points:

1. The divisor is repeatedly XORed with part of the dividend.
2. The divisor has $n - k + 1$ bits which either are predefined or are all 0s. In other words, the bits do not change from one dataword to another. In our previous example, the divisor bits were either 1011 or 0000. The choice was based on the leftmost bit of the part of the augmented data bits that are active in the XOR operation.

3. A close look shows that only $n - k$ bits of the divisor are needed in the XOR operation. The leftmost bit is not needed because the result of the operation is always 0, no matter what the value of this bit. The reason is that the inputs to this XOR operation are either both 0s or both 1s. In our previous example, only 3 bits, not 4, are actually used in the XOR operation.

Using these points, we can make a fixed (hardwired) divisor that can be used for a cyclic code if we know the divisor pattern. Figure 10.11 shows such a design for our previous example. We have also shown the XOR devices used for the operation.

Figure 10.11 *Hardwired design of the divisor in CRC*



Note that if the leftmost bit of the part of the dividend to be used in this step is 1, the divisor bits ($d_2d_1d_0$) are 011; if the leftmost bit is 0, the divisor bits are 000. The design provides the right choice based on the leftmost bit.

Augmented Dataword

In our paper-and-pencil division process in Figure 10.6, we show the augmented dataword as fixed in position with the divisor bits shifting to the right, 1 bit in each step. The divisor bits are aligned with the appropriate part of the augmented dataword. Now that our divisor is fixed, we need instead to shift the bits of the augmented dataword to the left (opposite direction) to align the divisor bits with the appropriate part. There is no need to store the augmented dataword bits.

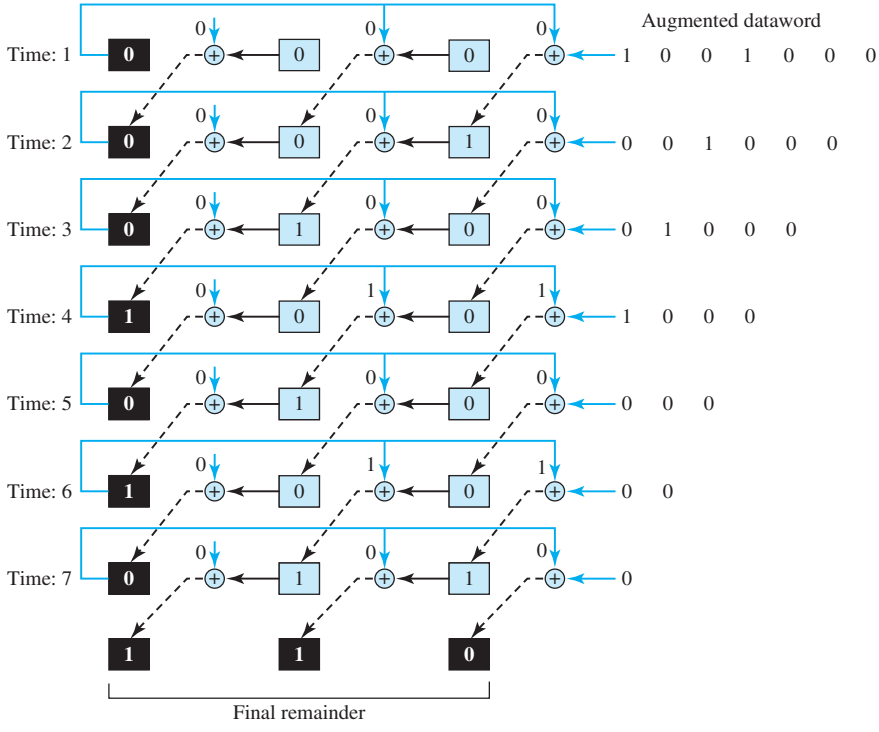
Remainder

In our previous example, the remainder is 3 bits ($n - k$ bits in general) in length. We can use three **registers** (single-bit storage devices) to hold these bits. To find the final remainder of the division, we need to modify our division process. The following is the step-by-step process that can be used to simulate the division process in hardware (or even in software).

1. We assume that the remainder is originally all 0s (000 in our example).
2. At each time click (arrival of 1 bit from an augmented dataword), we repeat the following two actions:
   a. We use the leftmost bit to make a decision about the divisor (011 or 000).
   b. The other 2 bits of the remainder and the next bit from the augmented dataword (total of 3 bits) are XORed with the 3-bit divisor to create the next remainder.

Figure 10.12 shows this simulator, but note that this is not the final design; there will be more improvements.

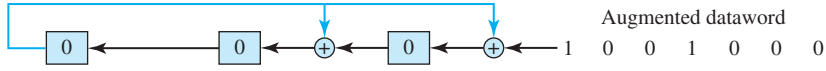
Figure 10.12 *Simulation of division in CRC encoder*



At each clock tick, shown as different times, one of the bits from the augmented dataword is used in the XOR process. If we look carefully at the design, we have seven steps here, while in the paper-and-pencil method we had only four steps. The first three steps have been added here to make each step equal and to make the design for each step the same. Steps 1, 2, and 3 push the first 3 bits to the remainder registers; steps 4, 5, 6, and 7 match the paper-and-pencil design. Note that the values in the remainder register in steps 4 to 7 exactly match the values in the paper-and-pencil design. The final remainder is also the same.

The above design is for demonstration purposes only. It needs simplification to be practical. First, we do not need to keep the intermediate values of the remainder bits; we need only the final bits. We therefore need only 3 registers instead of 24. After the XOR operations, we do not need the bit values of the previous remainder. Also, we do not need 21 XOR devices; two are enough because the output of an XOR operation in which one of the bits is 0 is simply the value of the other bit. This other bit can be used as the output. With these two modifications, the design becomes tremendously simpler and less expensive, as shown in Figure 10.13.

Figure 10.13 *The CRC encoder design using shift registers*

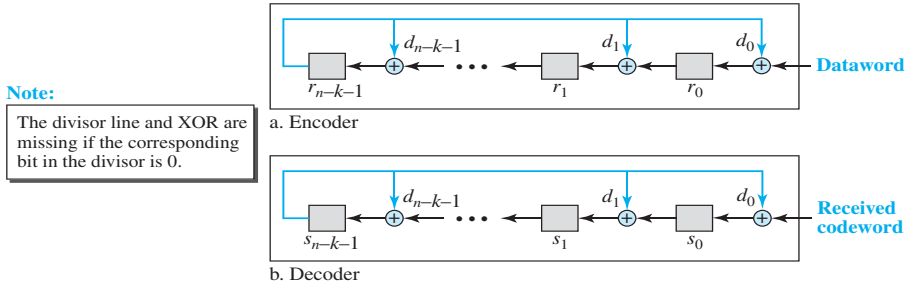


We need, however, to make the registers shift registers. A 1-bit shift register holds a bit for a duration of one clock time. At a time click, the shift register accepts the bit at its input port, stores the new bit, and displays it on the output port. The content and the output remain the same until the next input arrives. When we connect several 1-bit shift registers together, it looks as if the contents of the register are shifting.

General Design

A general design for the encoder and decoder is shown in Figure 10.14.

Figure 10.14 *General design of encoder and decoder of a CRC code*



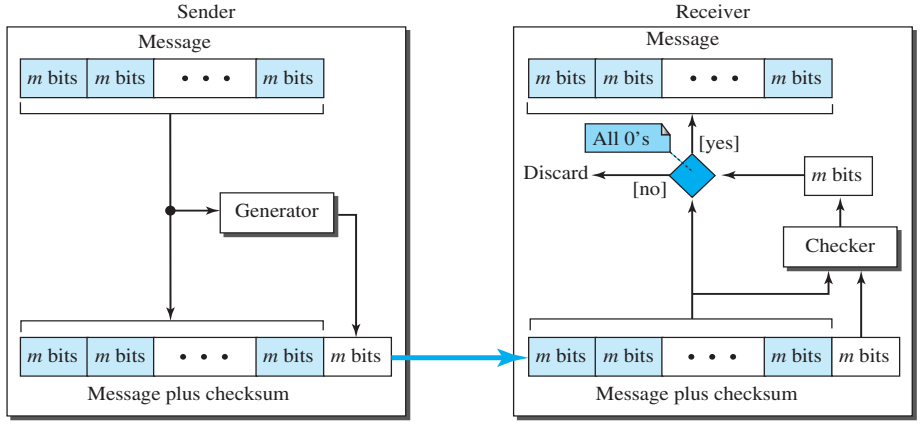
Note that we have $n - k$ 1-bit shift registers in both the encoder and decoder. We have up to $n - k$ XOR devices, but the divisors normally have several 0s in their pattern, which reduces the number of devices. Also note that, instead of augmented datawords, we show the dataword itself as the input because after the bits in the dataword are all fed into the encoder, the extra bits, which all are 0s, do not have any effect on the rightmost XOR. Of course, the process needs to be continued for another $n - k$ steps before the check bits are ready. This fact is one of the criticisms of this design. Better schemes have been designed to eliminate this waiting time (the check bits are ready after k steps), but we leave this as a research topic for the reader. In the decoder, however, the entire codeword must be fed to the decoder before the syndrome is ready.

10.4 CHECKSUM

Checksum is an error-detecting technique that can be applied to a message of any length. In the Internet, the checksum technique is mostly used at the network and transport layer rather than the data-link layer. However, to make our discussion of error-detecting techniques complete, we discuss the checksum in this chapter.

At the source, the message is first divided into *m*-bit units. The generator then creates an extra *m*-bit unit called the ***checksum,*** which is sent with the message. At the destination, the checker creates a new checksum from the combination of the message and sent checksum. If the new checksum is all 0s, the message is accepted; otherwise, the message is discarded (Figure 10.15). Note that in the real implementation, the checksum unit is not necessarily added at the end of the message; it can be inserted in the middle of the message.

Figure 10.15 *Checksum*



10.4.1 Concept

The idea of the traditional checksum is simple. We show this using a simple example.

Example 10.11

Suppose the message is a list of five 4-bit numbers that we want to send to a destination. In addition to sending these numbers, we send the sum of the numbers. For example, if the set of numbers is (7, 11, 12, 0, 6), we send (7, 11, 12, 0, 6, **36**), where 36 is the sum of the original numbers. The receiver adds the five numbers and compares the result with the sum. If the two are the same, the receiver assumes no error, accepts the five numbers, and discards the sum. Otherwise, there is an error somewhere and the message is not accepted.

One's Complement Addition

The previous example has one major drawback. Each number can be written as a 4-bit word (each is less than 15) except for the sum. One solution is to use **one's complement** arithmetic. In this arithmetic, we can represent unsigned numbers between 0 and $2^m - 1$ using only *m* bits. If the number has more than *m* bits, the extra leftmost bits need to be added to the *m* rightmost bits (wrapping).

Example 10.12

In the previous example, the decimal number 36 in binary is $(100100)_2$. To change it to a 4-bit number we add the extra leftmost bit to the right four bits as shown below.

$$(10)_2 + (0100)_2 = (0110)_2 \rightarrow (6)_{10}$$

Instead of sending 36 as the sum, we can send 6 as the sum (7, 11, 12, 0, 6, **6**). The receiver can add the first five numbers in one's complement arithmetic. If the result is 6, the numbers are accepted; otherwise, they are rejected.

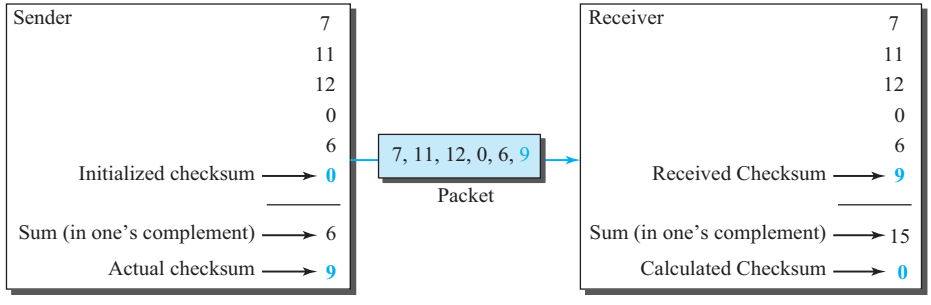
Checksum

We can make the job of the receiver easier if we send the complement of the sum, the checksum. In one's complement arithmetic, the complement of a number is found by completing all bits (changing all 1s to 0s and all 0s to 1s). This is the same as subtracting the number from $2^m - 1$. In one's complement arithmetic, we have two 0s: one positive and one negative, which are complements of each other. The positive zero has all m bits set to 0; the negative zero has all bits set to 1 (it is $2^m - 1$). If we add a number with its complement, we get a negative zero (a number with all bits set to 1). When the receiver adds all five numbers (including the checksum), it gets a negative zero. The receiver can complement the result again to get a positive zero.

Example 10.13

Let us use the idea of the checksum in Example 10.12. The sender adds all five numbers in one's complement to get the sum = 6. The sender then complements the result to get the checksum = 9, which is 15 − 6. Note that $6 = (0110)_2$ and $9 = (1001)_2$; they are complements of each other. The sender sends the five data numbers and the checksum (7, 11, 12, 0, 6, **9**). If there is no corruption in transmission, the receiver receives (7, 11, 12, 0, 6, **9**) and adds them in one's complement to get 15. The sender complements 15 to get 0. This shows that data have not been corrupted. Figure 10.16 shows the process.

Figure 10.16 *Example 10.13*



Internet Checksum

Traditionally, the Internet has used a 16-bit checksum. The sender and the receiver follow the steps depicted in Table 10.5. The sender or the receiver uses five steps.

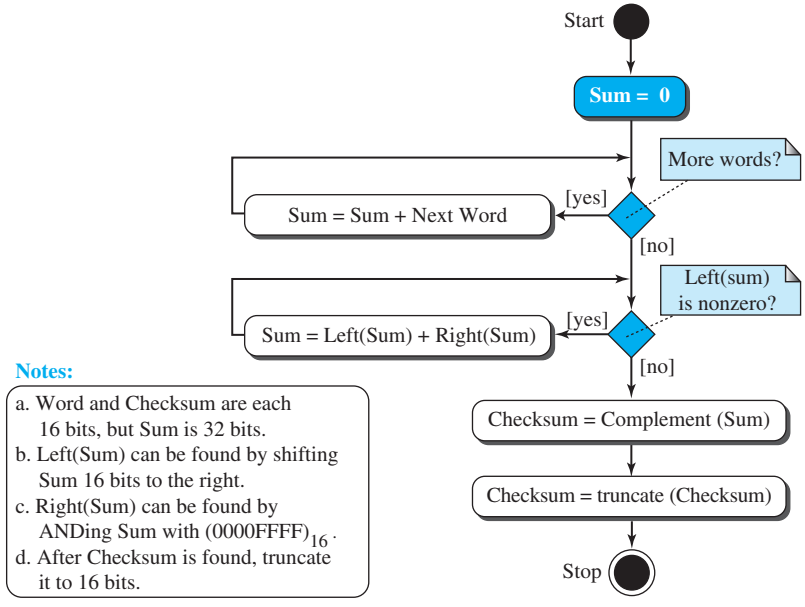
Table 10.5 *Procedure to calculate the traditional checksum*

Sender	*Receiver*
1. The message is divided into 16-bit words.	1. The message and the checksum are received.
2. The value of the checksum word is initially set to zero.	2. The message is divided into 16-bit words.
3. All words including the checksum are added using one's complement addition.	3. All words are added using one's complement addition.
4. The sum is complemented and becomes the checksum.	4. The sum is complemented and becomes the new checksum.
5. The checksum is sent with the data.	5. If the value of the checksum is 0, the message is accepted; otherwise, it is rejected.

Algorithm

We can use the flow diagram of Figure 10.17 to show the algorithm for calculation of the checksum. A program in any language can easily be written based on the algorithm. Note that the first loop just calculates the sum of the data units in two's complement; the second loop wraps the extra bits created from the two's complement calculation to simulate the calculations in one's complement. This is needed because almost all computers today do calculation in two's complement.

Figure 10.17 *Algorithm to calculate a traditional checksum*



Performance

The traditional checksum uses a small number of bits (16) to detect errors in a message of any size (sometimes thousands of bits). However, it is not as strong as the CRC in error-checking capability. For example, if the value of one word is incremented and the value of another word is decremented by the same amount, the two errors cannot be detected because the sum and checksum remain the same. Also, if the values of several words are incremented but the sum and the checksum do not change, the errors are not detected. Fletcher and Adler have proposed some weighted checksums that eliminate the first problem. However, the tendency in the Internet, particularly in designing new protocols, is to replace the checksum with a CRC.

10.4.2 Other Approaches to the Checksum

As mentioned before, there is one major problem with the traditional checksum calculation. If two 16-bit items are transposed in transmission, the checksum cannot catch this error. The reason is that the traditional checksum is not weighted: it treats each data item equally. In other words, the order of data items is immaterial to the calculation. Several approaches have been used to prevent this problem. We mention two of them here: Fletcher and Adler.

Fletcher Checksum

The Fletcher checksum was devised to weight each data item according to its position. Fletcher has proposed two algorithms: 8-bit and 16-bit. The first, 8-bit Fletcher, calculates on 8-bit data items and creates a 16-bit checksum. The second, 16-bit Fletcher, calculates on 16-bit data items and creates a 32-bit checksum.

The 8-bit Fletcher is calculated over data octets (bytes) and creates a 16-bit checksum. The calculation is done modulo 256 (2^8), which means the intermediate results are divided by 256 and the remainder is kept. The algorithm uses two accumulators, L and R. The first simply adds data items together; the second adds a weight to the calculation. There are many variations of the 8-bit Fletcher algorithm; we show a simple one in Figure 10.18.

The 16-bit Fletcher checksum is similar to the 8-bit Fletcher checksum, but it is calculated over 16-bit data items and creates a 32-bit checksum. The calculation is done modulo 65,536.

Adler Checksum

The Adler checksum is a 32-bit checksum. Figure 10.19 shows a simple algorithm in flowchart form. It is similar to the 16-bit Fletcher with three differences. First, calculation is done on single bytes instead of 2 bytes at a time. Second, the modulus is a prime number (65,521) instead of 65,536. Third, L is initialized to 1 instead of 0. It has been proved that a prime modulo has a better detecting capability in some combinations of data.

Figure 10.18 *Algorithm to calculate an 8-bit Fletcher checksum*

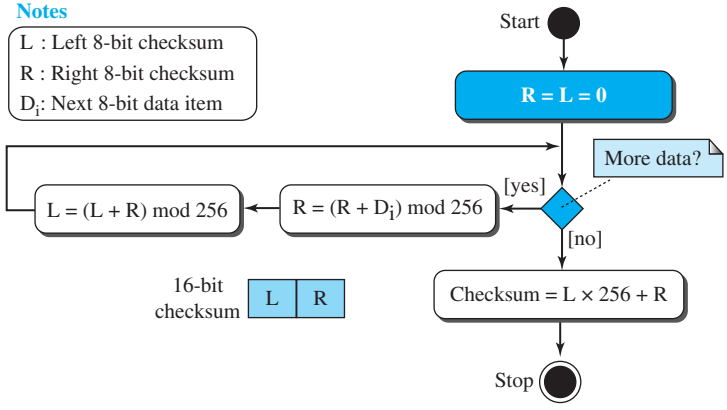
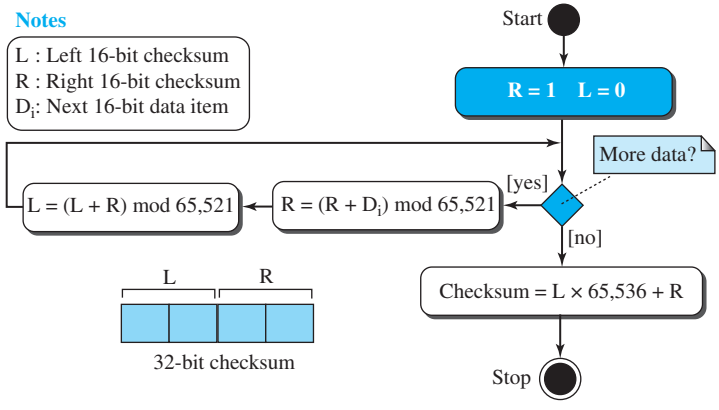


Figure 10.19 *Algorithm to calculate an Adler checksum*



To see the behavior of the different checksum algorithms, check some of the applets for this chapter at the book website.

10.5 FORWARD ERROR CORRECTION

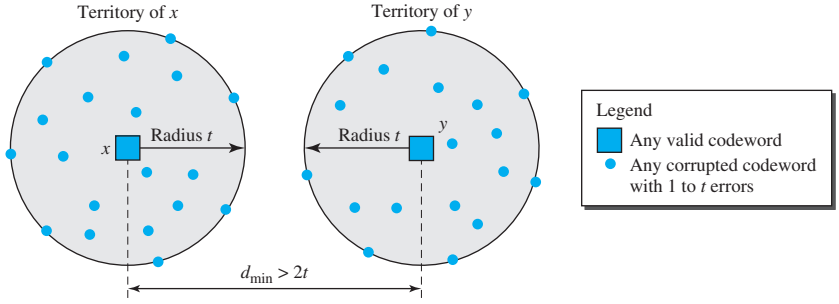
We discussed error detection and retransmission in the previous sections. However, retransmission of corrupted and lost packets is not useful for real-time multimedia transmission because it creates an unacceptable delay in reproducing: we need to wait until the lost or corrupted packet is resent. We need to correct the error or reproduce the

packet immediately. Several schemes have been designed and used in this case that are collectively referred to as **forward error correction** (**FEC**) techniques. We briefly discuss some of the common techniques here.

10.5.1 Using Hamming Distance

We earlier discussed the Hamming distance for error detection. We said that to detect s errors, the minimum Hamming distance should be $d_{min} = s + 1$. For error detection, we definitely need more distance. It can be shown that to detect t errors, we need to have $d_{min} = 2t + 1$. In other words, if we want to correct 10 bits in a packet, we need to make the minimum hamming distance 21 bits, which means a lot of redundant bits need to be sent with the data. To give an example, consider the famous BCH code. In this code, if data is 99 bits, we need to send 255 bits (extra 156 bits) to correct just 23 possible bit errors. Most of the time we cannot afford such a redundancy. We give some examples of how to calculate the required bits in the practice set. Figure 10.20 shows the geometrical representation of this concept.

Figure 10.20 *Hamming distance for error correction*



10.5.2 Using XOR

Another recommendation is to use the property of the exclusive OR operation as shown below.

$$\mathbf{R} = \mathbf{P_1} \oplus \mathbf{P_2} \oplus \ldots \oplus \mathbf{P}_i \oplus \ldots \oplus \mathbf{P}_N \quad \rightarrow \quad \mathbf{P}_i = \mathbf{P_1} \oplus \mathbf{P_2} \oplus \ldots \oplus \mathbf{R} \oplus \ldots \oplus \mathbf{P}_N$$

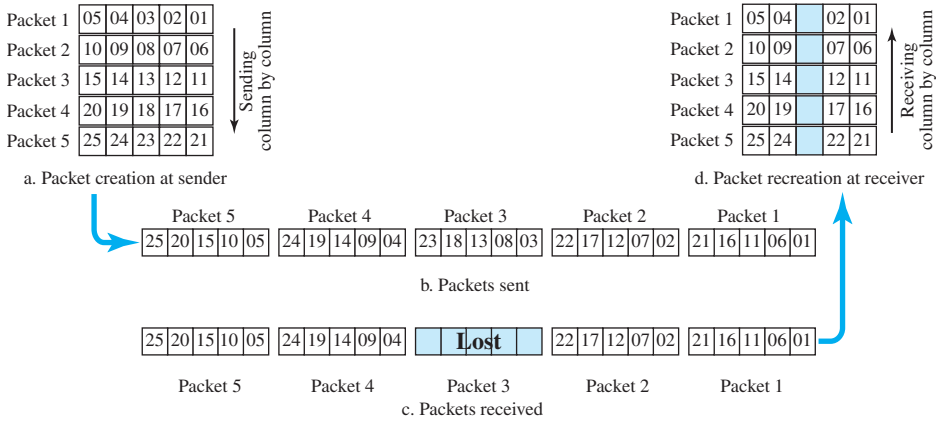
In other words, if we apply the exclusive OR operation on N data items (P_1 to P_N), we can recreate any of the data items by exclusive-ORing all of the items, replacing the one to be created by the result of the previous operation (R). This means that we can divide a packet into N chunks, create the exclusive OR of all the chunks and send $N + 1$ chunks. If any chunk is lost or corrupted, it can be created at the receiver site. Now the question is what should the value of N be. If $N = 4$, it means that we need to send 25 percent extra data and be able to correct the data if only one out of four chunks is lost.

10.5.3 Chunk Interleaving

Another way to achieve FEC in multimedia is to allow some small chunks to be missing at the receiver. We cannot afford to let all the chunks belonging to the same

packet be missing; however, we can afford to let one chunk be missing in each packet. Figure 10.21 shows that we can divide each packet into 5 chunks (normally the number is much larger). We can then create data chunk by chunk (horizontally), but combine the chunks into packets vertically. In this case, each packet sent carries a chunk from several original packets. If the packet is lost, we miss only one chunk in each packet, which is normally acceptable in multimedia communication.

Figure 10.21 *Interleaving*



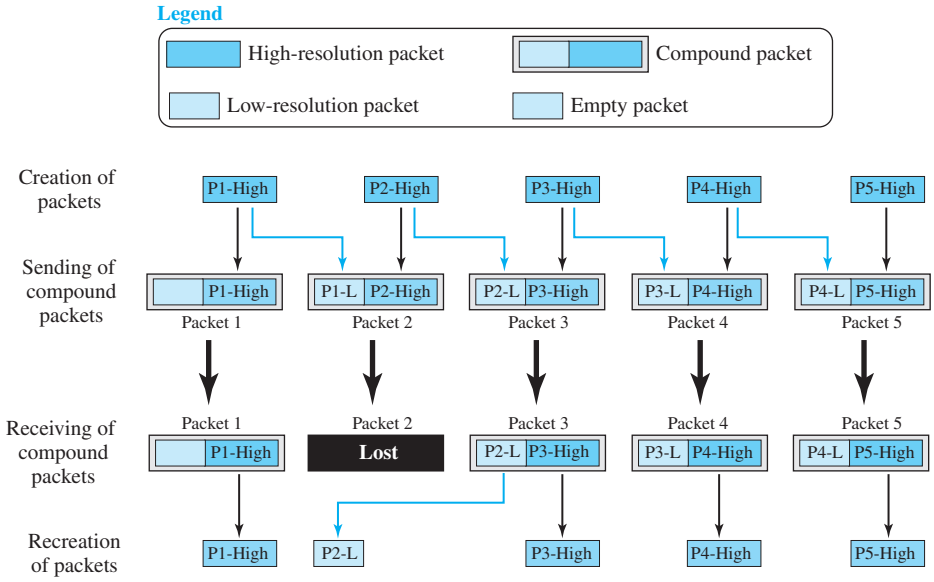
10.5.4 Combining Hamming Distance and Interleaving

Hamming distance and interleaving can be combined. We can first create n-bit packets that can correct t-bit errors. Then we interleave m rows and send the bits column by column. In this way, we can automatically correct burst errors up to $m \times t$-bit errors.

10.5.5 Compounding High- and Low-Resolution Packets

Still another solution is to create a duplicate of each packet with a low-resolution redundancy and combine the redundant version with the next packet. For example, we can create four low-resolution packets out of five high-resolution packets and send them as shown in Figure 10.22. If a packet is lost, we can use the low-resolution version from the next packet. Note that the low-resolution section in the first packet is empty. In this method, if the last packet is lost, it cannot be recovered, but we use the low-resolution version of a packet if the lost packet is not the last one. The audio and video reproduction does not have the same quality, but the lack of quality is not recognized most of the time.

Figure 10.22 *Compounding high- and low-resolution packets*



10.6 END-CHAPTER MATERIALS

10.6.1 Recommended Reading

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

Books

Several excellent books discuss link-layer issues. Among them we recommend [Ham 80], [Zar 02], [Ror 96], [Tan 03], [GW 04], [For 03], [KMK 04], [Sta 04], [Kes 02], [PD 03], [Kei 02], [Spu 00], [KCK 98], [Sau 98], [Izz 00], [Per 00], and [WV 00].

RFCs

A discussion of the use of the checksum in the Internet can be found in RFC 1141.

10.6.2 Key Terms

block coding
burst error
check bit
checksum
codeword
convolution coding
cyclic code
cyclic redundancy check (CRC)
dataword
error
error correction
error detection
forward error correction (FEC)
generator polynomial
Hamming distance
interference
linear block code
minimum Hamming distance
modular arithmetic
one's complement
parity-check code
polynomial
redundancy
register
Reed-Solomon code
single-bit error
syndrome

10.6.3 Summary

Data can be corrupted during transmission. Some applications require that errors be detected and corrected. In a single-bit error, only one bit in the data unit has changed. A burst error means that two or more bits in the data unit have changed. To detect or correct errors, we need to send extra (redundant) bits with data. There are two main methods of error correction: forward error correction and correction by retransmission.

We can divide coding schemes into two broad categories: block coding and convolution coding. In coding, we need to use modulo-2 arithmetic. Operations in this arithmetic are very simple; addition and subtraction give the same results. We use the XOR (exclusive OR) operation for both addition and subtraction. In block coding, we divide our message into blocks, each of k bits, called *datawords*. We add r redundant bits to each block to make the length $n = k + r$. The resulting n-bit blocks are called *codewords*.

In block coding, errors be detected by using the following two conditions:

a. The receiver has (or can find) a list of valid codewords.

b. The original codeword has changed to an invalid one.

The Hamming distance between two words is the number of differences between corresponding bits. The minimum Hamming distance is the smallest Hamming distance between all possible pairs in a set of words. To guarantee the detection of up to s errors in all cases, the minimum Hamming distance in a block code must be $d_{min} = s + 1$. To guarantee correction of up to t errors in all cases, the minimum Hamming distance in a block code must be $d_{min} = 2t + 1$.

In a linear block code, the exclusive OR (XOR) of any two valid codewords creates another valid codeword.

A simple parity-check code is a single-bit error-detecting code in which $n = k + 1$ with $d_{min} = 2$. A simple parity-check code can detect an odd number of errors.

All Hamming codes discussed in this book have $d_{min} = 3$. The relationship between m and n in these codes is $n = 2m - 1$.

Cyclic codes are special linear block codes with one extra property. In a cyclic code, if a codeword is cyclically shifted (rotated), the result is another codeword. A category

of cyclic codes called the cyclic redundancy check (CRC) is used in networks such as LANs and WANs.

A pattern of 0s and 1s can be represented as a polynomial with coefficients of 0 and 1. Traditionally, the Internet has been using a 16-bit checksum, which uses one's complement arithmetic. In this arithmetic, we can represent unsigned numbers between 0 and $2^n - 1$ using only n bits.

10.7 PRACTICE SET

10.7.1 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.

10.7.2 Questions

Q10-1. How does a single-bit error differ from a burst error?

Q10-2. What is the definition of a linear block code?

Q10-3. In a block code, a dataword is 20 bits and the corresponding codeword is 25 bits. What are the values of k, r, and n according to the definitions in the text? How many redundant bits are added to each dataword?

Q10-4. In a codeword, we add two redundant bits to each 8-bit data word. Find the number of

a. valid codewords. **b.** invalid codewords.

Q10-5. What is the minimum Hamming distance?

Q10-6. If we want to be able to detect two-bit errors, what should be the minimum Hamming distance?

Q10-7. A category of error detecting (and correcting) code, called the ***Hamming code,*** is a code in which $\boldsymbol{d}_{\text{min}} = 3$. This code can detect up to two errors (or correct one single error). In this code, the values of n, k, and r are related as: $n = 2^r - 1$ and $k = n - r$. Find the number of bits in the dataword and the codewords if r is 3.

Q10-8. In CRC, if the dataword is 5 bits and the codeword is 8 bits, how many 0s need to be added to the dataword to make the dividend? What is the size of the remainder? What is the size of the divisor?

Q10-9. In CRC, which of the following generators (divisors) guarantees the detection of a single bit error?

a. 101 **b.** 100 **c.** 1

Q10-10. In CRC, which of the following generators (divisors) guarantees the detection of an odd number of errors?

a. 10111 **b.** 101101 **c.** 111

Q10-11. In CRC, we have chosen the generator 1100101. What is the probability of detecting a burst error of length

a. 5? **b.** 7? **c.** 10?

Q10-12. Assume we are sending data items of 16-bit length. If two data items are swapped during transmission, can the traditional checksum detect this error? Explain.

Q10-13. Can the value of a traditional checksum be all 0s (in binary)? Defend your answer.

Q10-14. Show how the Fletcher algorithm (Figure 10.18) attaches weights to the data items when calculating the checksum.

Q10-15. Show how the Adler algorithm (Figure 10.19) attaches weights to the data items when calculating the checksum.

10.7.3 Problems

P10-1. What is the maximum effect of a 2-ms burst of noise on data transmitted at the following rates?

a. 1500 bps **b.** 12 kbps **c.** 100 kbps **d.** 100 Mbps

P10-2. Assume that the probability that a bit in a data unit is corrupted during transmission is p. Find the probability that x number of bits are corrupted in an n-bit data unit for each of the following cases.

a. $n = 8, x = 1, p = 0.2$

b. $n = 16, x = 3, p = 0.3$

c. $n = 32, x = 10, p = 0.4$

P10-3. Exclusive-OR (XOR) is one of the most used operations in the calculation of codewords. Apply the exclusive-OR operation on the following pairs of patterns. Interpret the results.

a. (10001) $\oplus$ (10001) **b.** (11100) $\oplus$ (00000) **c.** (10011) $\oplus$ (11111)

P10-4. In Table 10.1, the sender sends dataword 10. A 3-bit burst error corrupts the codeword. Can the receiver detect the error? Defend your answer.

P10-5. Using the code in Table 10.2, what is the dataword if each of the following codewords is received?

a. 01011 **b.** 11111 **c.** 00000 **d.** 11011

P10-6. Prove that the code represented by the following codewords is not linear. You need to find only one case that violates the linearity.

{(00000), (01011), (10111), (11111)}

P10-7. What is the Hamming distance for each of the following codewords?

a. d (10000, 00000) **b.** d (10101, 10000)
c. d (00000, 11111) **d.** d (00000, 00000)

P10-8. Although it can be formally proved that the code in Table 10.3 is both linear and cyclic, use only two tests to partially prove the fact:

a. Test the cyclic property on codeword 0101100.

b. Test the linear property on codewords 0010110 and 1111111.

P10-9. Referring to the CRC-8 in Table 5.4, answer the following questions:

a. Does it detect a single error? Defend your answer.

b. Does it detect a burst error of size 6? Defend your answer.

c. What is the probability of detecting a burst error of size 9?

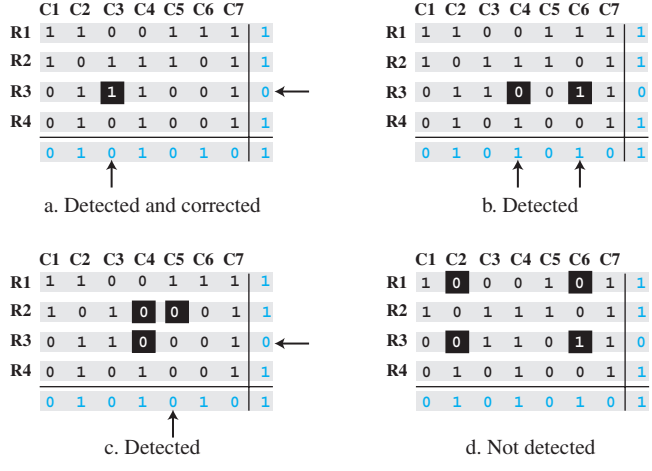
d. What is the probability of detecting a burst error of size 15?

P10-10. Assuming even parity, find the parity bit for each of the following data units.

a. 1001011 **b.** 0001100 **c.** 1000000 **d.** 1110111

P10-11. A simple parity-check bit, which is normally added at the end of the word (changing a 7-bit ASCII character to a byte), cannot detect even numbers of errors. For example, two, four, six, or eight errors cannot be detected in this way. A better solution is to organize the characters in a table and create row and column parities. The bit in the row parity is sent with the byte, the column parity is sent as an extra byte (Figure 10.23).

Figure 10.23 *P10-11*



Show how the following errors can be detected:

a. An error at (R3, C3).

b. Two errors at (R3, C4) and (R3, C6).

c. Three errors at (R2, C4), (R2, C5), and (R3, C4).

d. Four errors at (R1, C2), (R1, C6), (R3, C2), and (R3, C6).

P10-12. Given the dataword 101001111 and the divisor 10111, show the generation of the CRC codeword at the sender site (using binary division).

P10-13. Apply the following operations on the corresponding polynomials:

a. $(x^3 + x^2 + x + 1) + (x^4 + x^2 + x + 1)$

b. $(x^3 + x^2 + x + 1) - (x^4 + x^2 + x + 1)$

c. $(x^3 + x^2) \times (x^4 + x^2 + x + 1)$

d. $(x^3 + x^2 + x + 1) / (x^2 + 1)$

P10-14. Answer the following questions:

a. What is the polynomial representation of 101110?

b. What is the result of shifting 101110 three bits to the left?

c. Repeat part b using polynomials.

d. What is the result of shifting 101110 four bits to the right?

e. Repeat part d using polynomials.

P10-15. Which of the following CRC generators guarantee the detection of a single bit error?

a. $x^3 + x + 1$ **b.** $x^4 + x^2$ **c.** 1 **d.** $x^2 + 1$

P10-16. Referring to the CRC-8 polynomial in Table 10.7, answer the following questions:

a. Does it detect a single error? Defend your answer.

b. Does it detect a burst error of size 6? Defend your answer.

c. What is the probability of detecting a burst error of size 9?

d. What is the probability of detecting a burst error of size 15?

P10-17. Referring to the CRC-32 polynomial in Table 10.4, answer the following questions:

a. Does it detect a single error? Defend your answer.

b. Does it detect a burst error of size 16? Defend your answer.

c. What is the probability of detecting a burst error of size 33?

d. What is the probability of detecting a burst error of size 55?

P10-18. Assume a packet is made only of four 16-bit words $(A7A2)_{16}$, $(CABF)_{16}$, $(903A)_{16}$, and $(A123)_{16}$. Manually simulate the algorithm in Figure 10.17 to find the checksum.

P10-19. Traditional checksum calculation needs to be done in one's complement arithmetic. Computers and calculators today are designed to do calculations in two's complement arithmetic. One way to calculate the traditional checksum is to add the numbers in two's complement arithmetic, find the quotient and remainder of dividing the result by 2^{16}, and add the quotient and the remainder to get the sum in one's complement. The checksum can be found by subtracting the sum from $2^{16} - 1$. Use the above method to find the checksum of the following four numbers: 43,689, 64,463, 45,112, and 59,683.

P10-20. This problem shows a special case in checksum handling. A sender has two data items to send: $(4567)_{16}$ and $(BA98)_{16}$. What is the value of the checksum?

P10-21. Manually simulate the Fletcher algorithm (Figure 10.18) to calculate the checksum of the following bytes: $(2B)_{16}$, $(3F)_{16}$, $(6A)_{16}$, and $(AF)_{16}$. Also show that the result is a weighted checksum.

P10-22. Manually simulate the Adler algorithm (Figure 10.19) to calculate the checksum of the following words: $(FBFF)_{16}$ and $(EFAA)_{16}$. Also show that the result is a weighted checksum.

P10-23. One of the examples of a weighted checksum is the ISBN-10 code we see printed on the back cover of some books. In ISBN-10, there are 9 decimal digits that define the country, the publisher, and the book. The tenth (rightmost) digit is a checksum digit. The code, $D_1D_2D_3D_4D_5D_6D_7D_8D_9\mathbf{C}$, satisfies the following.

$$[(10 \times D_1) + (9 \times D_2) + (8 \times D_3) + \ldots + (2 \times D_9) + (1 \times \mathbf{C})] \bmod 11 = 0$$

In other words, the weights are 10, 9, . . .,1. If the calculated value for C is 10, one uses the letter X instead. By replacing each weight w with its complement in modulo 11 arithmetic $(11 - w)$, it can be shown that the check digit can be calculated as shown below.

$$\mathbf{C} = [(1 \times D_1) + (2 \times D_2) + (3 \times D_3) + \ldots + (9 \times D_9)] \bmod 11$$

Calculate the check digit for ISBN-10: **0-07-296775-C**.

P10-24. An ISBN-13 code, a new version of ISBN-10, is another example of a weighted checksum with 13 digits, in which there are 12 decimal digits defining the book and the last digit is the checksum digit. The code, $D_1D_2D_3D_4D_5D_6D_7D_8D_9D_{10}D_{11}D_{12}\mathbf{C}$, satisfies the following.

$$[(1 \times D_1) + (3 \times D_2) + (1 \times D_3) + \ldots + (3 \times D_{12}) + (1 \times \mathbf{C})] \bmod 10 = 0$$

In other words, the weights are 1 and 3 alternately. Using the above description, calculate the check digit for ISBN-13: **978-0-07-296775-C**.

P10-25. In the interleaving approach to FEC, assume each packet contains 10 samples from a sampled piece of music. Instead of loading the first packet with the first 10 samples, the second packet with the second 10 samples, and so on, the sender loads the first packet with the odd-numbered samples of the first 20 samples, the second packet with the even-numbered samples of the first 20 samples, and so on. The receiver reorders the samples and plays them. Now assume that the third packet is lost in transmission. What will be missed at the receiver site?

P10-26. Assume we want to send a dataword of two bits using FEC based on the Hamming distance. Show how the following list of datawords/codewords can automatically correct up to a one-bit error in transmission.

$$00 \rightarrow 00000 \qquad 01 \rightarrow 01011 \qquad 10 \rightarrow 10101 \qquad 11 \rightarrow 11110$$

P10-27. Assume we need to create codewords that can automatically correct a one-bit error. What should the number of redundant bits (r) be, given the number of bits in the dataword (k)? Remember that the codeword needs to be $n = k + r$ bits, called C(n, k). After finding the relationship, find the number of bits in r if k is 1, 2, 5, 50, or 1000.

P10-28. In the previous problem we tried to find the number of bits to be added to a dataword to correct a single-bit error. If we need to correct more than one bit, the number of redundant bits increases. What should the number of redundant bits (r) be to automatically correct one or two bits (not necessarily contiguous) in a dataword of size k? After finding the relationship, find the number of bits in r if k is 1, 2, 5, 50, or 1000.

P10-29. Using the ideas in the previous two problems, we can create a general formula for correcting any number of errors (m) in a codeword of size (n). Develop such a formula. Use the combination of n objects taking x objects at a time.

P10-30. In Figure 10.22, assume we have 100 packets. We have created two sets of packets with high and low resolutions. Each high-resolution packet carries on average 700 bits. Each low-resolution packet carries on average 400 bits. How many extra bits are we sending in this scheme for the sake of FEC? What is the percentage of overhead?

10.8 SIMULATION EXPERIMENTS

10.8.1 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.

10.9 PROGRAMMING ASSIGNMENTS

For each of the following assignments, write a program in the programming language you are familiar with.

Prg10-1. A program to simulate the calculation of CRC.

Prg10-2. A program to simulate the calculation of traditional checksum.

Prg10-3. A program to simulate the calculation of Fletcher checksum.

Prg10-4. A program to simulate the calculation of Adler checksum.