Michael A. Gallo
William M. Hancock

dp
Digital Press

Networking Explained

Second Edition

NETWORKING

Networking Explained, Second Edition

This Page Intentionally Left Blank

Networking Explained, Second Edition

MICHAEL A. GALLO, PH.D.
Florida Institute of Technology

WILLIAM M. HANCOCK, PH.D.
Exodus Communications Incorporated

Digital Press

Boston • Oxford • Auckland • Johannesburg • Melbourne • New Delhi

A member of the Reed Elsevier group

Recognizing the importance of preserving what has been written, Butterworth–Heinemann prints its books on acid-free paper whenever possible.

Gallo, *Networking Explained, Second Edition*
ISBN 1-55558-252-4

The publisher offers special discounts on bulk orders of this book.
For information, please contact:
Manager of Special Sales
Butterworth–Heinemann
225 Wildwood Avenue
Woburn, MA 01801–2041
Tel: 781-904-2500
Fax: 781-904-2620
For information on all Butterworth–Heinemann publications available, contact our World Wide Web home page at: http://www.bh.com

10 9 8 7 6 5 4 3 2 1
Printed in the United States of America

To Dee Dee Pannell . . .

a terrific computer systems and network supervisor,
and a remarkable human being

This Page Intentionally Left Blank

Contents

This Page Intentionally Left Blank

Preface

Welcome to the second edition of *Networking Explained*. In preparing this new edition, our goal was threefold: (1) to update the book so that it reflects, as best as possible, the current state of computer networking; (2) to enhance the discussion of key concepts so that you will have a greater understanding of how certain protocols work; and (3) to maintain the elements and features of the previous edition that contributed to its success. We believe that we have succeeded in accomplishing our goals.

Key Content Changes

Chapter 1: Fundamental Concepts of Computer Networks and Networking now contains an overview of networking and a discussion of the main issues involved in networking. We also provide additional information about storage area networks (SANs), discuss the World Wide Web as a networking communications model, and include information about file-sharing programs such as Napster.

Chapter 2: Network Topologies, Architectures, and the OSI Model includes new historical information about the development of the OSI model. We also rewrote the material on packet-switched networks to improve the clarity of the presentation.

Chapter 3: The Internet and TCP/IP was extensively rewritten and now contains information about TCP/IP's application layer protocols, including SMTP, MIME, POP, IMAP, TELNET, FTP, and HTTP. We also extended our discussion on TCP/IP's two transport layer protocols, UDP and TCP, and provided more detailed information about subnets and subnet masking.

Chapter 4: Physical Layer Concepts contains updated information on UTP cables and wireless media. We also revised the discussion on IEEE 802.11 and included new material about HomeRF.

Chapter 5: Data Link Layer Concepts and IEEE Standards contains new material on various flow control protocols, including stop-and-wait, sliding window, and go-back-*n*, and includes a new boxed text item that demonstrates how CRC checksums are calculated.

Chapter 7: Internetworks and Network Layer Concepts and Components contains a new boxed text item that discusses Nyquist's theorem and includes additional information about the pulse code-modulation (PCM) process. We also extended our coverage on the tunneling concept and included new information about multiprotocol label switching (MPLS) and virtual private networks (VPNs). Finally, we moved information about VPN security protocols from Chapter 16 and placed it in this chapter.

Chapter 8: Ethernet, Ethernet, and More Ethernet contains updated information on Gigabit Ethernet and includes new material on 10 Gigabit Ethernet.

Chapter 15: Dialup and Home Networking contains expanded coverage on cable modem service and the various digital subscriber line (DSL) protocols. We also updated our discussion on modem protocols to include V.44 and V.92, and rewrote the material dealing with establishing a home network.

Chapter 16: Network Security Issues was rewritten to improve topic clarity and organization. The chapter was also updated and now contains new information about denial of service (DoS) attacks, public key infrastructure (PKI), the advanced encryption standard (AES), and a new boxed text item that demonstrates the RSA algorithm.

New Chapters and Appendix

In addition to the content changes cited above, we also wrote two new chapters and one new appendix.

Chapter 17: Network Convergence contains a thorough discussion of the convergence concept from several perspectives, including its impact on networks, network media, businesses, and the home. The chapter also contains several examples of convergence applications, including Enum and the Bluetooth project, and provides information about multimedia networking from a convergence perspective as well as information about voice over IP (VoIP).

Chapter 18: Wireless Networking extends the wireless communications discussion from Chapter 4 and includes information about cellular telephone networks and protocols, paging networks, and wireless data networks. This chapter also expands Chapter 4's discussion on IEEE 802.11b and 802.11a and compares wireless LANs (WLANs) to the Bluetooth technology. Finally, the chapter concludes with a brief discussion of the future of wireless communications.

Appendix E: X.25 provides an overview of the X.25 standard for data communications.

Retained Features

One of the most redeeming features of the first edition of this book was its question and answer format, which we modeled after a terrific book on statistics by James Brewer: *Everything You Always Wanted To Know about Statistics But Didn't Know How to Ask.* This question-answer presentation was retained for the second edition. As was the case with the previous edition, we structured the question-answer format to emulate a conversation between a networking professional and the reader. The questions are representative of those asked by individuals who are interested in computer networks and those who wish to gain additional understanding of the subject. The answers are intended to give the reader a broad foundation in networking concepts.

A second retained feature is the collection of informative figures, tables, and boxed text items. We have included 32 new figures, 26 new tables, and 6 new boxed text items. In all total, this new edition contains nearly 300 pieces of art. The figures graphically represent key concepts, the tables summarize salient information, and the boxed text items explore topics in greater detail without getting in the way of the main discussion.

A third feature we retained in this edition is the glossary. A comprehensive 700-term glossary complements the book's nucleus.

A fourth feature we retained is the overall structure of the chapters. As was the case with the first edition, all the chapters are structured logically to follow each other and build on previous learned knowledge. At the same time, however, the chapters may be read in any order because we wrote them to also be independent of each other. Throughout every chapter, key terms or concepts that were either presented in an earlier chapter or discussed in a later chapter, are explained in the current context with either a forward or backward reference that directs the reader to the appropriate chapter where the term or concept is discussed more completely. These cross-references enable readers to take a break from the current discussion so they can either refresh their knowledge of previously presented material or jump ahead to gain further insight into a specific topic. All chapters are also formatted consistently. Each chapter begins with an introduction that includes a bullet list of the major topics discussed. Topics also contain corresponding question numbers, thereby making it easy to quickly locate information about a particular topic. Questions are categorized hierarchically by subject so that busy readers who seek answers to specific questions can find them easily. End-of-chapter commentaries are also provided. These commentaries consist of transitional material that identifies other chapters in the book containing additional information related to the current discussion.

Finally, in writing this new edition, we were careful to maintain our overall philosophy on which the first edition was grounded: to provide current and future network managers and administrators with an accurate and easy to read book on data communications standards and emerging networking technologies that is accessible by readers of all backgrounds. Given its format and level of coverage, we equate the book to a snorkeling adventure. We primarily stay at the surface to examine the features, attributes, technical issues, and concepts of networks. Occasionally, we hold our breath and dive under to explore a particular concept in more detail. We do not, however, discuss any particular topic in too much depth. Such an undertaking is better left for more technical books, which are analogous to a scuba diving expedition.

What This Book Is and Isn't About

After reading this book, you will accrue a greater understanding of and appreciation for networks and networking. This book will help you understand basic networking terminology, components, applications, protocols, architectures, standards, and implementation strategies. The reader is cautioned that this is *not* a "how-to" book. We do not provide specific information relative to network management or configurations. Thus, the material contained here will not help you perform such tasks as setting up a domain name server, configuring a network printer, or installing or managing an office network. However, your knowledge, appreciation, comprehension, and awareness of the concepts involved in such activities will be more acute after reading this book.

Secondary Audiences

In addition to its primary audience, this book lends itself to three secondary audiences. First, the book is appropriate for computer-networking hobbyists or nonprofessionals who desire to gain a working knowledge of the vocabulary, concepts, and current technologies related to networking. Second, the book is suitable for individuals who have a working knowledge of networks, but lack an understanding of the fundamental concepts and theoretical underpinnings of networks. Finally, the book can be used as a companion resource in an academic setting.

Acknowledgments

Many people contributed considerably, either directly or indirectly, in the preparation of this material. It is therefore justifiable that these individuals be acknowledged. First of all, we are grateful to the authors of the articles, books, RFCs, and other reference material listed in the Bibliography. These publications served as invaluable resources for confirming that our illustrations and material were accurate, complete, and up-to-date. Next, several people reviewed various aspects of the material in one form or another. They include Prasad Aloni of Drexel University, Ron Fulle of Rochester Institute of Technology, Arnold Meltzer of George Washington University, Krishna Sivalingam of Washington State University, Eugene Styer of Eastern Kentucky University, and Michael Whitman of Kennesaw State University. The feedback we received from these individuals was invaluable and we are grateful for their assistance. It is also with pleasure that we acknowledge and thank John Rhoton, who is a wireless technology consultant at Compaq Global Services. John provided us with immense support and timely assistance in the preparation of the wireless networking chapter. We are also extremely appreciative of the support and guidance provided by our editor, Pam Chester, and by Karen Forster, production director, for keeping this project on schedule. Thanks Pam, and thanks Karen. It was a pleasure working with you both. A sincere note of appreciation is also extended to Kristin Merz, who provided tremendous assistance with the artwork. Finally, on a personal note, one of us is indebted to Jane for always believing that "this will be the last one." Without her support, patience, understanding, and love, this project would never have been completed. Thank you sweetie. I love you.

Chapter 1

Fundamental Concepts of Computer Networks and Networking

In this chapter, we provide an overview of the various topics and concepts surrounding computer networks and networking. We begin by discussing some of the key issues related to networking. We follow this discussion with a formal definition of computer networks along with a description of the various terms related to this definition. As part of this discussion, we also introduce basic networking vocabulary as well as several key networking concepts. Much of the material we present here is done so from a general perspective and is elaborated in subsequent chapters. Understanding this material is important because it serves as the underlying foundation of the book. An outline of the terms and concepts we define and discuss follows:

- Concepts and Related Issues of Computer Networking (Questions 1–2)
- Definition and Components of Computer Networks (Questions 3–9)
- Network Protocols (Questions 10–12)
- Network Media (Questions 13–14)
- Applications vs. Application Protocols (Questions 15–19)
- Interoperability (Questions 20–21)
- Internet, internet (Internetwork), Intranet, Extranet (Questions 22–24)
- LAN, MAN, WAN, PAN, GAN, SAN (Questions 25–27)
- Decentralized vs. Centralized vs. Distributed Systems (Questions 28–30)
- Client/Server, Peer-to-Peer, Servent, and WWW Network Models (Questions 31–41)
- Network Appliances (Questions 42–44)
- Serial and Parallel Communications (Questions 45–48)
- Synchronous, Asynchronous, and Isochronous Communications (Questions 49–53)
- Simplex, Half-Duplex, and Full-Duplex Communications (Questions 54–55)
- Network Standards (Questions 56–64)

1. The title of this book is *Networking Explained*. Please explain the term "networking" and distinguish it from the term "network."

Before we respond directly, consider for a moment what some people do when they attend a professional conference or meeting—they usually engage in some form of *networking*. That is, they seek out people in an attempt to establish contacts so they can exchange or share information. Collectively, these individuals form a human network. (Have you ever heard someone say, "She has her own network of friends"?) Things are not that much different in the computer world. Specifically, the term "networking" refers to the concept of connecting a group of systems for the expressed purpose of sharing information. The systems that are connected form a network.

2. What are some of the issues involved in networking?

Computer networking involves many issues. A brief overview of some key ones follows.

- *Communication methodology and protocols.* In order for network members to understand each other's communication, they must first know and follow specific communication rules. In the world of computer networks, communication methodology and protocols provide these rules of communication. We will discuss the concept of communication methodology and protocols later in this chapter.
- *Topology and design.* A key issue of networking involves how to connect the individual systems that comprise a network. Thus, the topology and design criteria are important topics in any discussion of computer networks. Topology and design are the focus of Chapter 2.
- *Addressing.* Addressing describes how systems are assigned addresses and how systems locate each other within a network. It involves assigning a network node a unique address so that other systems or devices are able to locate it. It is similar to a house's street address—knowing the street helps to find where we want to go, but having the house number means we will eventually find the exact location of our destination. Addressing is first introduced in this chapter and is then revisited in nearly every subsequent chapter.
- *Routing.* Related to addressing is the concept of routing, which describes the manner in which data are transferred from one system to another across a network. Routing involves determining the path a message takes as it travels between source and destination nodes. Routing is usually performed by special dedicated hardware units called *routers*. The "best" route packets should take is a function of a specific criterion (or criteria), which is called a *metric*. Some of the issues involved in routing include the path a message follows, how a path is determined, and various policy issues that govern the control of a path. Routing is discussed in Chapter 3 and then again in more detail in Chapter 7.
- *Reliability.* Reliability refers to *data integrity*, which has to do with ensuring that the data received are identical to the data transmitted. Computer networks are nontrivial systems and are not infallible. In fact, most networks are highly fragile and easily disrupted. Thus, it is important that they be designed with the capability to resolve errors. A common error-control strategy is to provide

enough information in the transmitted data to enable the destination node to detect an error. This is called *error detection*. Once a destination node has detected an error, it can then do one of two things: request a retransmission from the sending node or determine what the correct data should be and change them accordingly. Both methods are forms of *error correction*. We call the first *error correction through retransmission* and the second *autonomous error correction*. Note that error correction implies error detection. Autonomous error correction is very expensive to implement. Hence, most networks today are designed with error-detection capabilities, and error correction is done by having the destination node request the sending node to retransmit the data in question. The concept of and issues related to reliability are first discussed in Chapter 2 and then re-examined from different perspectives in nearly every subsequent chapter. The two most popular error-detection strategies—parity check and cyclic redundancy check (CRC)—are discussed in Chapter 5.

- *Interoperability.* In any free enterprise system, consumers want the freedom to determine which products to purchase. Unfortunately, this can sometimes lead to disastrous results when purchasing networking products from different vendors. Thus, it is important that interoperability issues are considered. This refers to the degree in which software and hardware products developed by different vendors are able to communicate successfully with each other over a network. The topic of interoperability is first introduced later in this chapter and then highlighted in most subsequent chapters, particularly Chapter 7.
- *Security.* Whenever we engage in any networking activity, we would like some assurances that our communications are secure. Thus, security issues are of paramount concern and pertain to the safekeeping or protection of all components of a network. Network security refers to the proper safeguarding of everything associated with a network. This includes data, media, and equipment. Security involves administrative functions such as *threat assessment* ("What do you have and who would want it?"), technical tools and facilities such as cryptographic products, and network access control products such as *firewalls*, which are special hardware devices that serve to protect an internal network from the outside world. Security also involves making certain that network resources are used in accordance with a prescribed policy and only by people who are authorized to use these resources. Chapter 16 is dedicated to network security issues.
- *Standards.* The development and implementation of networks rely on the establishment of specific rules and regulations to be followed. This is the role of networking standards. We introduce the concept of network standards later in this chapter and then include a discussion of its impact in every chapter thereafter.

3. OK. Now that I have some idea of what networking is, what exactly is a computer network?

A computer network is a collection of computers and other devices that use a common network protocol to share resources with each other over a network medium.

4. That's a mouthful. Can you break this down for me, please? There are several terms I do not understand.

Sure. Where do you want to start?

5. First of all, you say a computer network is a collection of computers and other devices. What other devices?

In our definition we use the term *device* to represent any entity that is connected to a network. Such entities may be terminals, printers, computers, or special network-related hardware units such as communication servers, repeaters, bridges, switches, routers, hand-held units such as the *PalmPilot*, and various other devices, most of which are discussed in detail in later chapters. Devices can be either local or remote. The device originating communication across a network is called the *local device* or *sending device*, and any device within the network that is accessed from this local device is called the *remote device* or *receiving device*. In a telephone network, the telephone handsets we all use are devices. So is the interconnecting hardware at the phone company that allows handsets to talk to each other. A network requires many diverse types of devices in order to work.

6. So *device* is a generic term.

Correct.

7. What about *node*? I see this term used frequently.

The word *node* is commonly used interchangeably with device; both terms refer to any equipment that can access a network. You also frequently see the term *station* as well as *appliance* used in the literature. Both are synonymous with device and node. All four terms refer to any equipment that can access a network.

8. If *device* and *node* are generic terms that refer to any entity connected to a network, then why do you use the term *computers* in your definition?

We prefer to distinguish between devices and computers. As network devices, computers are called *hosts* (or *servers*) or *workstations* (also called *desktops* or *clients*). This terminology refers to computer systems that have their own operating systems (such as Windows). Thus, a workstation might be a personal computer such as an Apple Macintosh or a Windows-based machine such as a Compaq, Dell, or Gateway PC; a graphic workstation such as those manufactured by Sun Microsystems, Silicon Graphics, IBM, or Hewlett Packard; a super-minicomputer such as an IBM AS/400 system; a super-microcomputer like Compaq's Alpha; or perhaps a mainframe such as an IBM ES-9000.

9. With so many different types of nodes connected to a network, what mechanism do they use to find each other?

You are referring to the concept of "addressing," which was mentioned earlier in our response to Question 2 and is discussed in more detail in later chapters. For now, though, suffice it to say that a network node is assigned a unique address that allows other systems or devices to locate them. It is similar to a house's street address—knowing the street helps to find where you want to go, but having the house number means you will eventually find the exact location of your destination. Another analogy is the phone system you use all the time. Each phone (a node) has an area code and a number (an address). The area code provides information about the node's location within the global telephone network, and the telephone number is the device's specific identification number within that locale. Systems and call "routers" at the phone company are programmed to provide information to other network devices to get the call from your phone handset to the proper destination (the phone number you are calling).

10. OK. Next is *network protocol*. What does this mean?

Let's take this in two parts, starting with the term *protocol*. From a general perspective, a *protocol* is an accepted or established set of procedures, rules, or formal specifications governing specific behavior or language. For example, when eating in a fancy, expensive restaurant, patrons are usually required to observe a specific dress protocol (e.g., men typically have to wear a jacket and tie). Other restaurants, such as the ones frequented by the authors, may have different dress protocols: no shoes, no shirt, no service. If you were to meet the Queen of England, once again you would need to observe a certain protocol. When applied to networking and data communications, a *network protocol* is a formal specification that defines how nodes are to "behave" or communicate with each other. Among other things, network protocols define how data are to be formatted, how data integrity is to be maintained, and how data are transmitted and received between nodes. In short, a network protocol specifies the vocabulary and rules of data communication.

11. Can you give me an example of a network protocol?

Yes. A good example consists of the individual protocols that are part of the TCP/IP suite. TCP/IP stands for "Transmission Control Protocol/Internet Protocol," which serves as the basis of the Internet. (See Chapters 2 and 3 for more information about TCP/IP and the Internet.) Although TCP/IP specifies two particular protocols (TCP and IP), it is also used to name the set of protocols that includes not only TCP and IP, but also many others. This set of protocols is called the TCP/IP *suite*. (When a group of related and interoperating protocols are put together in a package on a system, we call it a suite.)

Another protocol that is part of the TCP/IP suite is FTP, or "File Transfer Protocol," which specifies how to do file transfers. HTTP, the Hypertext Transport Protocol, is used for the World Wide Web (WWW), and defines how servers need to transfer documents (Web pages) to clients (Web browsers). Three protocols used for electronic mail (e-mail) with which you might already be familiar are the Post Office Protocol (POP), the Simple Mail Transfer Protocol (SMTP), and the Internet Mail Access Protocol (IMAP). All of the foregoing are network protocols that are also part of the TCP/IP suite. Today's networks employ a great multitude of protocols, ranging from very simple to quite complex. Protocols are the glue that binds together computer networks because they define how specific operations are to be performed. We will provide detailed examples of how some of these protocols actually work in later chapters of the book.

12. Are there other sets (or suites, as you called them) of network protocols?

Yes. One you might be familiar with is *AppleTalk*, which is a network protocol suite used by Apple Computer, Inc., originally in its line of Macintosh computers and now available in many other operating systems. Another example is the set of protocols that are part of Microsoft Corporation's Windows NT operating systems. Sometimes, computer networks are named by their protocols. For example, a network that consists of devices supporting AppleTalk is referred to as an AppleTalk network. Similarly, a TCP/IP network implies a set of devices linked together that uses the TCP/IP suite as its set of rules for communication.

13. OK. Getting back to your definition of computer networks, I now understand what you mean by devices and network protocol. One last thing you mentioned that I am unclear about is "network medium."

Besides protocols, nodes have to be connected to each other in some manner to share resources or receive services via a network. The physical environment used to connect members of a network is referred to as a *medium* (the plural of which is *media*). Network media come in two broad categories: cable and wireless. Examples of cable include twisted-pair, coaxial, and fiber-optic cable. Examples of wireless include radio waves (including microwave and satellite communication) and infrared radiation. Network media will be discussed in more detail in Chapter 4.

14. So, computer networks require media and protocols because without a link, resources cannot be shared, and without protocols, communication cannot be understood even if a link exists. Is this correct?

Right. Network media provide an environment in which communication can take place, while protocols are necessary to ensure that communications are understood. This is similar to a telephone conversation between one person who speaks only Italian and another who speaks only Russian. If a telephone circuit (i.e., network link) for this conversation is not available, then these two individuals cannot speak to each other (i.e., they cannot share resources). Given a circuit, the two individuals can now speak and hear each other's voices (i.e., transmission), but communication cannot take place because neither individual is capable of understanding the other's message—they speak different languages. Networking happens when a common wiring infrastructure connects nodes that share a common protocol infrastructure—just like human communication.

15. Regardless of the protocol used, don't most computer networks support similar network-related applications like e-mail?

Yes they do. Although the function of these applications across different networks is similar, the manner in which they are implemented is protocol-dependent. For example, e-mail messages can be exchanged between hosts connected to a TCP/IP network because they speak the same language. Similarly, e-mail messages can be exchanged between hosts of a Windows NT network because, once again, they speak the same language. However, e-mail messages cannot be exchanged directly between a host connected to a TCP/IP network and a host connected to a Windows NT network because they may use different application protocols for electronic mail. Consequently, although different networks might be functionally equivalent in that they support similar applications (e.g., TCP/IP and NT networks both support electronic mail), the manner in which these functions are implemented is not the same. As an example, UNIX systems support the e-mail protocol we previously mentioned, called SMTP, which is a component of the TCP/IP suite of protocols. NT systems support a different e-mail system called Exchange. By default, they cannot interoperate directly even though they are both connected to same network. However, Exchange can be configured to communicate not only "native" with its own protocols, but also to simultaneously support TCP/IP and SMTP so that UNIX and NT users can exchange e-mail.

A network may have many protocols and many applications. Not all of them necessarily talk directly with each other. Software functions called *gateways* (explained later) allow conversion (like a linguistic translation) between some application protocols. In other areas the problems are solved simply by supporting more than one application protocol at the same time. While this sounds complex, usually a little care in the planning cycle makes everything work well when activated for use.

16. So what you are saying then is that there is a difference between an application like e-mail and the protocol that defines it.

Exactly! To help understand this better, consider the hierarchy in Figure 1.1. At the root layer we have a network protocol suite (TCP/IP). The next layer shows three network applications that are part of this suite (e-mail, file transfer, and virtual terminal). The third layer contains the protocols that define these applications (SMTP and POP for e-mail, FTP for file transfer, and TELNET for virtual terminal). The last layer identifies a specific program that users can use for these applications. A similar tree diagram can be drawn for NT, or any other protocol suite. In short: A network protocol suite provides the specifications for network applications such as e-mail. These applications have specific protocols that define how the application is to be implemented on the network. The application protocols also include specific user programs that we use to interact with the application.

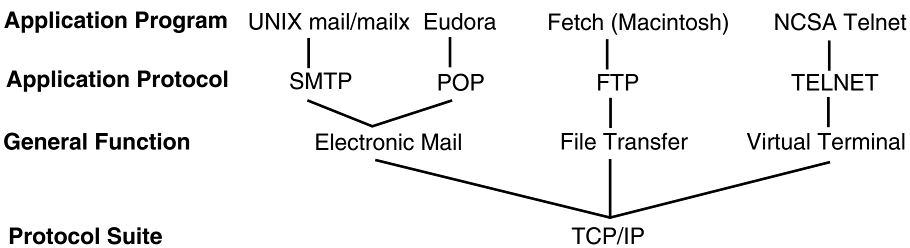


FIGURE 1.1 A protocol suite contains specific network applications (e.g., e-mail), which in turn are defined by specific application protocols (e.g., SMTP). These application protocols are part of specific application programs (e.g., UNIX mail) that provide a user with an interface to interact with an application. The application protocol also defines the manner in which an application is to be implemented between two hosts connected to a network.

17. Can you give me an example to make this a little more concrete?

Sure. Let's use e-mail as our example. E-mail is a network application that involves creating, sending, receiving, and storing messages electronically. These activities are performed by using a "mail program," which provides a utility for users to (among others) compose, read, save, delete, and forward e-mail messages. This mail program is an application program that resides on a host. A mail program is also concerned with issues such as how a host accepts or rejects mail, how mail is stored on a system, how a user is notified of the arrival of new mail messages, and so forth. A mail program does not, however, manage the network exchange of e-mail messages between two hosts. Instead, the method by which e-mail is transferred from one host to another is handled by an e-mail application protocol. Examples include SMTP, IMAP, and POP, which are part of the TCP/IP protocol suite.

Other network applications are similar to e-mail. They consist of an application program that provides the user with an interface to interact with the application, and they contain a related application protocol that defines the manner in which an application communicates over a network. Thus, file transfer programs provide users with an interface for copying files to or from a remote host, and virtual terminal programs provide users with an interface for establishing a login on a remote host. These applications also have corresponding protocols (FTP and TELNET, respectively) that define the rules local and remote hosts must follow to run the application across the network.

18. Is there a one-to-one correspondence between an application program and network application?

No. Some network applications support more than one application protocol. For example, the public domain package NCSA Telnet, supports both the virtual terminal protocol (TELNET) and the file transfer protocol (FTP). As another example, consider the software product from Netscape Communications, Netscape Communicator. This product supports several protocols including those for network news (NNTP), e-mail (SMTP, POP, and IMAP4), the World Wide Web (HTTP), and file transfers (FTP).

19. I can understand why nodes have to use the same protocol, but do users also have to use the same program to communicate with one another? For example, does everyone I send e-mail to have to use the same e-mail application I use?

No, not at all. Remember, an application program simply provides the user with an interface to interact with the application. Behind this application is an associated protocol, which is transparent to the user. As long as the application's corresponding protocol understands another application's protocol everything should work out. Thus, you might use Eudora as your e-mail package, but someone else might choose to use the e-mail program that is part of Netscape Communicator, and still a third person might use the e-mail application that is part of Microsoft's Internet Explorer. It doesn't matter—all these applications support protocols that understand each other. It's similar to a superhighway. Not all vehicles are the same, but they all have rules on what they can do in the lanes, how fast they can go, and how they get on and off the road. We all know what happens when someone violates the rules on a road—it's messy. The concern you have is a valid one. It also is part of a much larger network-related issue known as *interoperability*.

20. Interoperability? What's that?

Earlier we stated that interoperability refers to the degree in which products (software and hardware) developed by different vendors are able to communicate successfully with each other over a network. During the heyday of proprietary (private, vendor-specific, or in-house) networks, interoperability really wasn't an issue as long as one stayed with a specific vendor's products and protocols. Occasionally a third-party vendor would set up shop and develop an application that had more bells and whistles (called *valued-added features*) than your vendor was offering. To do so, though, this third-party vendor had to receive permission from the primary vendor; which usually implied paying a licensing fee. Today, however, with TCP/IP being an "open" standard, and with the Internet's extremely rapid growth, vendors who want to write and sell TCP/IP-based applications are free to do so without fear of violating any proprietary copyrights. Although the protocol specifications for the TCP/IP suite of applications are freely available, the interpretation of these protocols by different vendors is not always the same. This, coupled with the fact that there is no governing body to oversee the development of TCP/IP-based products, sometimes leads to incompatible products.

21. Thanks for the tip. Is this something I really need to be concerned about?

You should be cognizant of it, particularly if you are a network manager responsible for the applications that run across your network, at whom fingers will point when things go wrong. Most computer vendors strive for interoperability with other vendors' products. In fact, one of the largest networking trade shows in the world is called Networld+Interop (for interoperability). Each year at Networld+Interop competing vendors convene to display their products and to demonstrate how they can interoperate with other vendors' products. Still, the issue of interoperability is paramount, and you should exercise care when considering using network products from different vendors.

22. Since we're talking about it, is the Internet a computer network?

Although it might appear that way, the Internet is not a computer network. Recall our response to Question 3. We said that a computer network "is a collection of computers and other devices." The Internet does not consist of a collection of computers and other devices. Instead, it consists of a collection of computer networks.

Just as computers can be connected to one another to form a network, computer networks can be connected to one another creating what is known as a network of networks, or an *internet*. For example, a network located in an office on one floor of a building can be connected to another network located on a different floor of the same building. Collectively, these two interconnected networks represent an internet.

23. I noticed that you wrote "internet" and not "Internet?" Is there a difference between the two?

The term *internet* is an abbreviation for *internetwork*, which refers to a collection of interconnected networks that functions as a single network. When used as a proper noun and spelled with an uppercase *I*, the Internet refers to the world's largest internetwork, which consists of hundreds of thousands of interconnected networks worldwide and has associated with it a certain culture. The Internet also implies a set of networks that support the same network protocol, namely, TCP/IP. Thus, the Internet is a collection of computer networks based on a specific set of network standards (TCP/IP), which describe how the computers of each individual network are to communicate with each other. The Internet allows individual, autonomous networks to function and appear as a single large network. The Internet and TCP/IP are discussed in more detail in Chapter 3.

24. OK, I'm really confused now. I've heard of something called an intranet. How is that different from an internet?

It's easy to get confused with all the network buzzwords. An *intranet* is the internal network implementation of traditional Internet applications within a company or an institution. Examples of applications that run on corporate or institutional internets are Web servers, e-mail, newsgroups. There are many others. It is, in the strictest sense, still an internet (notice the lack of an uppercase *I*), but it is easier to understand that the speaker is referring to the internal corporate network by calling it an intranet. To make things even more confusing, a popular networking term for an interconnection from the internal intranet to a customer or noncompany network that is not the Internet connection is called an *extranet* connection. This may involve a leased-line connection or some other network type of connection; it may also involve the use of a secure protocol to "tunnel" through the Internet. This is discussed further in Chapter 7.

In summary, an *intranet* is an internal company network that implements traditional Internet services; an *extranet* is a network connection to noncompany entities that are not being accessed via an Internet connection; and the *Internet* is a series of worldwide network services available from an Internet service provider (ISP), which is discussed in more detail in Chapter 3.

25. Are the networks connected to the Internet called *local area networks* or *LANs*?

Some are and some are not. Computer networks frequently are classified by the geographical area they encompass. One classification is *local area network* (LAN). Another is *wide area network* (WAN). A LAN generally interconnects computing resources within a moderately sized geographical area. This can include a room, several rooms within a building, or several buildings of a campus. Since the term "moderately sized" is not well defined, some people quantify a LAN's range by restricting it from a few feet to several miles or kilometers (the IEEE usually relates this to 10 km or less in radius). In contrast to a LAN, a WAN interconnects computing resources that are widely separated geographically (usually over 100 km). This includes towns, cities, states, and countries. Following the quantification of a LAN's range, a WAN would span an area greater than five miles (eight kilometers). A WAN can be thought of as consisting of a collection of LANs.

26. So the Internet consists of a collection of WANs and LANs?

That's correct. You should note, though, that some people make further distinctions between LANs and WANs. One such distinction is *metropolitan area network* (MAN), which interconnects computing resources that span a metropolitan area. For example, consider a large business organization with buildings located throughout a local county or city. If each building has its own independent LAN, and if these LANs were interconnected to one another (thus forming an internet), the resulting network could be considered a MAN since all of the buildings are located within the same metropolitan area, namely, the local county. MANs generally refer to networks that span a larger geographical area than LANs but a smaller geographical area than WANs.

Another classification is *personal area network* (PAN), which refers to the small computer networks that are found in private homes. The relatively low cost of computers and the resulting growing number of multicomputer homes is driving the need for PANs as home computer users begin to realize the convenience of interconnecting their computers. For example, PANs can interconnect multiple home computers to the same printer, thereby eliminating the need to purchase separate printers for each computer. PANs can also enable home-based computer users to use a file server on which application software and user data can reside but are accessible from any machine connected to the home network. PANs also provide all members of a household with convenient access to home-based shared computing resources from their private rooms (e.g., a child's bedroom, home office, or kitchen). Still another classification is *global area network* (GAN), which refers to a collection of WANs that span the globe. For example, many businesses such as McDonald's Restaurants have operations in many different countries throughout the world. The interconnection of these individual business locations makes up a GAN. Finally, there is *storage area network* (SAN), which is a network dedicated exclusively for storing data. Figure 1.2 provides a pictorial representation of some of these types of networks. Note that many of the devices shown in Figure 1.2 are discussed later in this chapter or in subsequent chapters.

As an interesting note, IBM has created a device that is worn like a pager by humans and is connected to the wearer's skin via small sensors. Using the conductivity of the skin of the wearer as the network medium, the device can communicate with another wearer of a comparable device to exchange information between the personal devices. IBM calls this type of network a personal area network as well, although it is somewhat different than the above definition of a PAN.

27. Could you expand on the concept of storage are networks for a moment?

Sure. A SAN can be thought of as a "back-end" network that connects data storage devices via specialized channels such as the small computer systems interface (SCSI, pronounced "scuzzy") or Fibre channel. We will discuss these channels later in Chapter 4. For the time being, you may think of these channels simply as a method for connecting devices. A SAN can be configured in a centralized or distributed manner. A centralized SAN configuration (Figure 1.3a) links multiple hosts to a single storage system. Consistent with the centralized concept, a centralized SAN makes data storage management simpler and easier but also represents a single source of failure. A distributed SAN configuration (Figure 1.3b) connects multiple hosts with multiple storage devices. Finally, and for completeness, there is also something called *network attached storage* (NAS), which interconnects hosts and storage devices via a network device such as a a repeater hub or network switch (Figure 1.3c). We will discuss repeater hubs and switches later in Chapter 6.

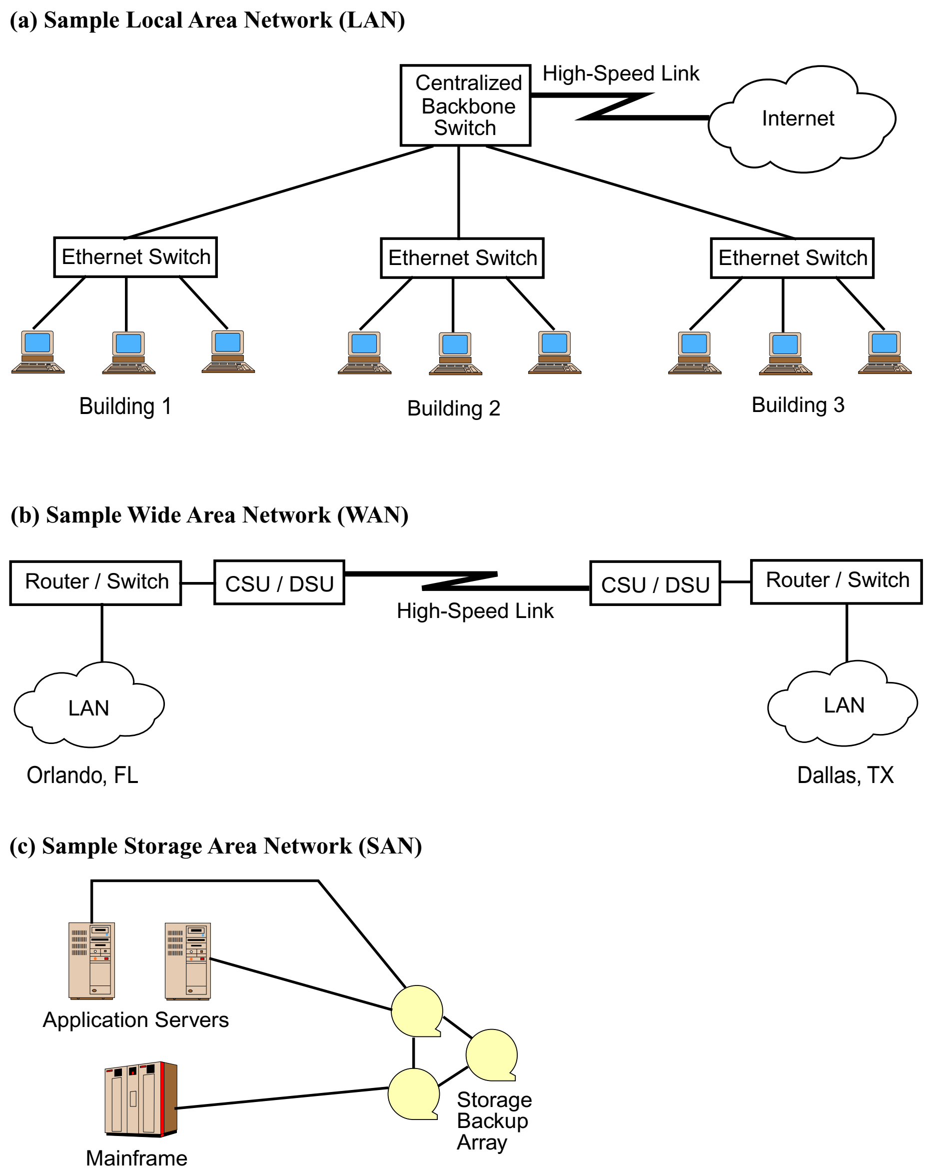


FIGURE 1.2 Examples of various types of networks. In (a) a local area network (LAN) interconnects the computing resources housed in several buildings of an organization. Note that if the buildings were located throughout a metropolitan area such as a city or county, then we would have a metropolitan area network (MAN). In (b) a wide area network (WAN) interconnects LANs that are located in different states. If LANs, MANs, and WANs that span the globe are interconnected, then we have a global area network (GAN). In (c) a storage area network (SAN) is a back-end network that connects storage devices to multiple application servers or mainframes.

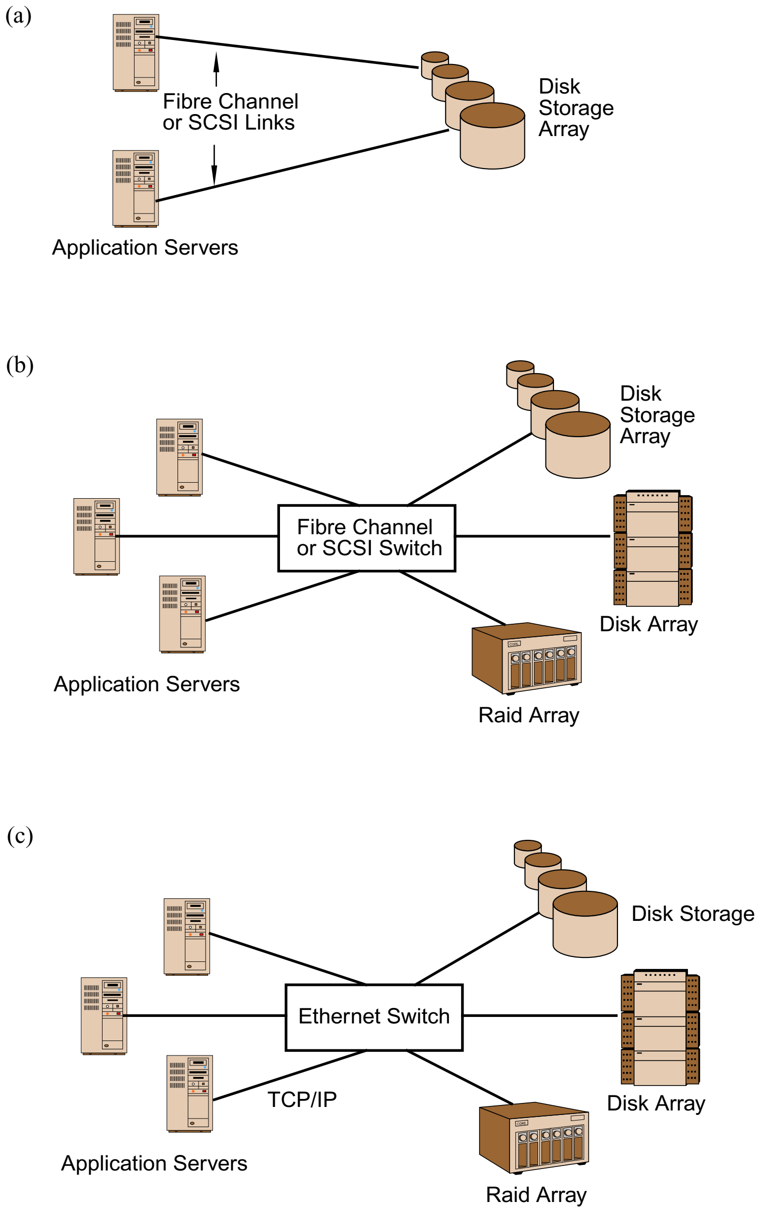


FIGURE 1.3 A storage area network (SAN) comprises several components, including an interface such as a Fibre channel or SCSI link, an interconnect such as a switch or hub, a protocol such as SCSI or IP, and storage devices such as disks or tapes. All are connected to multiple application servers. In (a) a centralized SAN (channel attached) configuration is shown; in (b) a distributed SAN (channel attached is shown; and in (c) a networked attached SAN is shown.

28. In Figure 1.2(a), I see the word "centralized" is used. Sometimes when I hear people talk about computer networks, they often use terms like *centralized*, *decentralized*, and *distributed*. What do these terms mean?

The terms *centralized*, *decentralized*, and *distributed* are part of the old MIS (management information systems) vocabulary and denote specific data communications models. These terms are more applicable to the use of computing and not networking resources. However, they are still used today to describe various communication models and hence warrant consideration. To understand the differences among them, think of an organization's typical computing needs. For example, there is accounting, inventory, and maintenance of personnel records. Computer communication models can be designed to handle these functions in several ways. In a decentralized model, offices or departments have their own systems, independent of each other, and maintain separate databases germane to their specific activities. In a centralized model, a single computer provides all the computing resources for all offices and departments within an organization. Finally, in a distributed model, computers are linked together to provide, in a transparent manner, the required computing resources and information processing needs of the entire organization. Distributed systems bear the greatest resemblance to computer networks. A pictorial description that distinguishes among these three terms is given in Figure 1.4. Note that although these terms are still used today, their corresponding computer communication models have since evolved into various networking models. These include client/server, peer-to-peer, Web-based, and servent. We discuss these models later in the chapter.

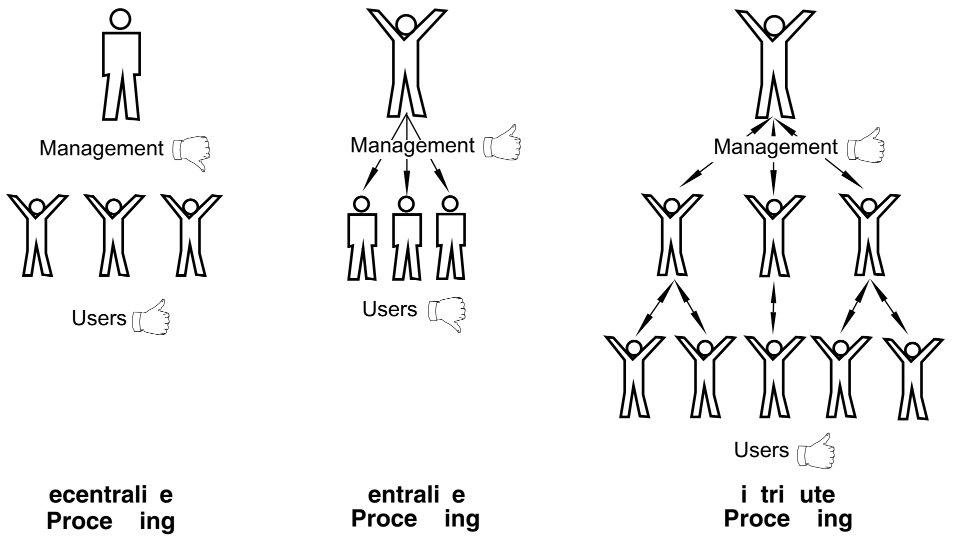


FIGURE 1.4 In a decentralized environment, users are "happy" but managers are not; in a centralized environment, managers are "happy" but users are not; in a distributed environment, both managers and users are "happy." Source: Adapted from Distributed Processing, by F. Kaufman, p. 14. Copyright © 1977 Coopers and Lybrand. Found in Data Center Operations: A Guide to Effective Planing, Processing, and Performance, 2e, by H. Schaeffer, p. 272. Copyright @ 1987 Prentice-Hall.

29. In what way are distributed systems similar to networks?

A distributed system consists of independent computers interconnected to one another. The primary difference between the two is that in a distributed environment resources are made available to the user in a transparent manner. What this means is that users are unaware that the computers are interconnected. From a user's perspective, a distributed system appears as if it were a single system. Using specially designed software, all functions of a distributed system are handled without users ever having to explicitly request a specific service. In a networked environment, though, users must explicitly identify what it is they want done.

30. I need an example of how a distributed system differs from a computer network.

No problem. Consider the task of editing a file that resides on a remote system. In a distributed environment a user would simply "call up" the file to be edited and the system would make the file available. In a computer network, however, the user must first know on what remote host the file resides, and then either (a) transfer the file to the local host (which involves running a file transfer program), or (b) establish a login to the remote host on which the file is located (which involves running a virtual terminal program to log into the remote host). Thus, in a distributed system the file appears local to the user regardless of where the file actually resides within the system, whereas in a computer network the user must be cognizant of the file's residence and then explicitly perform some function to gain access to the file. Although not exactly a computer network, there is considerable overlap between distributed systems and computer networks. Suffice it to say that a distributed system represents a special case of a network, with the major distinction being the software as opposed to the hardware.

31. Now that I have some understanding of these legacy communication models, tell me about the current networking models that you mentioned earlier, namely, client-server, peer-to-peer, Web-based, and servent.

OK. We'll begin with client/server. Most network communications and applications today are based on a paradigm called the *client/server model*. This model describes network services (e.g., file transfers, terminal connections, electronic mail, and printing) and the programs used by end users to access these services. The client/server model can be thought of as dividing a network transaction into two parts: The client side (or front end) provides a user with an interface for requesting services from the network, and the server side (or back end) is responsible for accepting user requests for services and providing these services transparent to the user. Both terms—client and server—can be applied to either application programs or actual computing devices.

32. Please give me an example.

Okay. Let's assume you are using Microsoft Word on a networked PC. Let's further assume that the printer you will use to print this document is a networked device accessible by your PC. Thus, we have a common scenario in which an end user using an application program (Word) needs to access a specific network service (printing). In this context, the application program becomes a client when it relays the print request to the printer, while the printer is the server, which accepts and services this request. It's similar to being served in a restaurant. You are the client who issues a request (you order a salad), and your waiter or waitress is the server who services your request (he or she brings you the salad).

33. I understand the concept. What I was hoping for was a little more detail. For example, would you walk me through an Internet-based client/server example?

Sure. A typical Internet-based client/server interaction works as follows:

- A server process is started on a host. This process notifies the host that it is ready to accept client requests. The server process then waits for a client process to contact it to request a specific network application service.
- Independent of the server process, a client process is started. This process can be invoked either on the same system that is hosting the server process, or on another computer that is connected to the same network to which the computer supporting the server process is connected. Regardless of which system is involved, a client process is usually initiated by a user through an application program. A request for service is sent by the client process to the host that is providing the requested service and the server program running on that host responds to the request.
- When the server process has fully honored the client's request, the server returns to a "wait" state and waits for another client request from the same or another client.

On UNIX systems, a server process is commonly referred to as a *daemon* and is designated by the letter *d* at the end of a program's name. (*Note:* On operating systems such as Windows NT, OpenVMS, and OS/400, different nomenclature is used.) For example, the virtual terminal program *telnet* represents the client side and its companion *telnetd* (pronounced "telnet dee") is the server side. Similarly, the file transfer program has both a client and a server side, *ftp* and *ftpd*. An example of an Internet-based client/server interaction is shown in Figure 1.5. Host A is running an HTTP server process (*httpd*), and a user on B is requesting a specific document from this server using a Web browser such as Microsoft's Internet Explorer or Netscape's Communicator. The Web browser is the application that supports the HTTP client protocol. When the server receives the request, it processes it by transferring the requested Web page or document to the client.

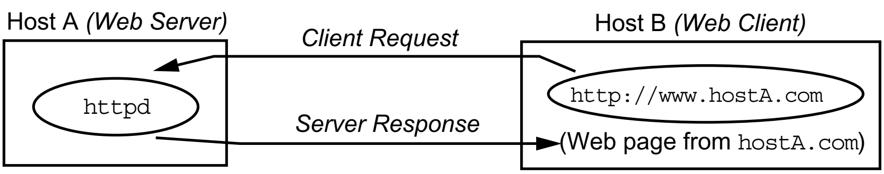


FIGURE 1.5 Example of an HTTP client/server process. Host A is running a Web server process, and host B is running a Web client process (i.e., a Web browser). When a user on host B opens a location (i.e., a Web address or URL), a connection is established to the machine at that address. The server accepts this request and services it by transferring the requested document to the client machine that made the request.

Computer systems that run specific server programs are commonly referred to by the application service they provide. For instance, a host that accepts and provides mail service is known as a *mail server*; a computer that provides users access to files remotely is known as a *file server*; a computer running httpd is known as a *Web server*; and a computer that runs a network news protocol (e.g., NNTP) is known as a *news server.*

34. Thanks. That was helpful. How does the client/server model compare to the *peer-to-peer* model?

Peer-to-peer is another model on which some network communications and applications are based. In a peer-to-peer environment, each networked host runs both the client and server parts of an application. (Contrast this to a client/server environment in which a host is capable of running only client-based applications, only server applications, or both client and server applications, thereby acting as a server for one application but a client for another application.) This is accomplished by installing the same *network operating system* (NOS) on all hosts within the network, enabling them to provide resources and services to all other networked hosts. For example, each networked host in a peer-to-peer environment can allow any other connected host to access its files or print documents on its printer while it is being used as a workstation. Once again, the key distinction between peer-to-peer computing and client/server computing is the former requires each networked host to run both client and server parts of all applications. Examples of peer-to-peer networks include Microsoft's Windows NT and Windows 95/98/2000, Apple's AppleTalk, Artisoft's LANtastic, and Novell's Personal NetWare. Peer-to-peer networks are relatively inexpensive to purchase and operate, and are fairly easy and straightforward to configure. They represent a cost-effective model for small organizations or departments that want to enjoy some of the benefits of networking but do not have the requisite resources (financial, human, or equipment). However, peer-to-peer networks can be less reliable than client/server based networks. They usually also require the use of more powerful workstations for certain activities (e.g., sharing a database) than a client/server based network.

35. Time out. What is a network operating system and how different is it from a regular operating system?

The term *network operating system* (NOS) refers to software that is installed on a system to make it network-capable. Examples include IBM's LAN Server, Banyan's VINES, and Novell's NetWare (also known as IntranetWare). In each of these cases, the NOS is independent of a computer's native operating system—it is loaded "on top" of the computer's operating system and provides the computer with networking capability based on a particular protocol. If, on the other hand, a computer's native operating system includes built-in network support, then a NOS refers to that particular OS. Examples include Sun Microsystems' Solaris Operating System, Hewlett-Packard's HP-UX Operating System, and Microsoft's NT Server. You can think of the NOS in this case meaning *networkable* operating system.

36. OK. Getting back to the peer-to-peer model, if my PC is running Windows 95/98/2000 and is connected to a network, then any other PC running Windows 95/98/2000 that is connected to the same network can access files on my machine because of this peer-to-peer thing. Is that correct?

Yes, provided your machine has been configured for this access. It's just like Apple Macintoshes, which support AppleTalk. If you go into the Control Panel and invoke "file sharing," you can make your Macintosh accessible to any other machine connected to the same network that also is running AppleTalk. In Windows, you select My Computer, followed by the disk you are interested in sharing, and then right-click the mouse. If your network Control Panel applet has been configured for file sharing, you will see the word "Sharing" in the menu and you simply select it and then set up the parameters for sharing that will be allowed. (If sharing is already enabled on a Windows 95/98/2000 system, you will see a picture of a hand holding the device icon.) This is the concept of peer-to-peer networking—it enables users to easily share resources on a network.

Network printing is implemented in a similar manner. For example, a locally connected printer in Windows NT 4.*x* can be configured as a network device. During the initial configuration process, the "Add Printer Wizard" provides two radio buttons—Shared and Not Shared. Selecting "Shared" makes the printer accessible network-wide. (You also have to give it a name and identify the operating systems that will print to this printer so the proper device drivers are installed.) Now, assuming users have the proper access permissions, this printer appears as an available resource whenever users browse network resources in Windows NT Explorer.

37. This sounds like the way Napster works.

Exactly. As you probably know, Napster was the brainchild of Shawn Fanning, who, as a 19-year old college student, was interested in making it easy for music listeners (especially other college students) to share copies of their favorite recordings. Prior to Napster's development, electronic versions of songs were accessible via the Internet and could be downloaded using a Web browser and TCP/IP's file transfer protocol (FTP). Recordings were saved using the MPEG-1 audio layer III (MP3) compression format (see Chapter 17). This approach was deficient, though, because there was no current list that identified which recordings were available and where they were located.

Napster addressed this deficiency. Using a locally installed client application along with a custom networking protocol, users specify to a Napster server their local host's Internet address and the names of the recordings they want to share. Other Napster clients can then search this server for the files they want. The actual file transfer, however, does not involve the Napster server. Instead, the transfer is between two Napster clients—the host on which the requested file resides and the host that is requesting the file. Thus, the Napster server simply brokers the file transfer between clients.

Using our previously developed vocabulary, in the Napster model the local and remote systems are both client and server, similar to the peer-to-peer model. In this model, however, all the Napster hosts are concurrently servers and clients, and the term that is used to denote this is *servent*, which is a combination of SERVer and cliENT. Thus, unlike the peer-to-peer model, which is one-to-one, or the client/server model, which is one-to-many, the Napster model is many-to-many. The servent concept is expected to be expanded into a general file-sharing model that will enable all interconnected hosts to exchange any type of file residing on their hard drives.

38. Isn't this sort of what Gnutella is like? How does Gnutella relate to Napster?

Yes. Gnutella is similar to Napster. It too is an open, decentralized, peer-to-peer file search system. Unlike Napster, though, Gnutella is not a company, it is not an application, and it is not a Web site. It is simply a name for a file-sharing technology.

39. What's the future of the servent model given the court's ruling against Napster?

This is a difficult question to answer because of the problematic nature of current law as applied to the Internet. Nevertheless, some believe that once all the copyright related issues and attending legal matters are resolved, the servent concept is expected to be expanded into a general file-sharing model that will enable all interconnected hosts to exchange any type of file residing on their hard drives.

40. Is this sharing model concept also true for client/server? For example, can other machines access my machine if they are running the same NOS?

No. In the client/server model, it is important to note that a network service can only be provided if a server program responsible for servicing a request is running on a particular host. For example, look back at Figure 1.5. If host A was not running the httpd process, then the request from host B would not be honored. This is why PCs used to access Internet services such as e-mail or the World Wide Web are relatively secure from being compromised by "outside" users. These machines usually run client versions of Internet-related applications. For example, Eudora is an Internet-based mail client program that requests mail service from a mail server. It makes a connection to another machine running the server process to retrieve mail for a user. If you are running Eudora on your PC, Eudora users on other machines cannot connect to your machine just because you are running Eudora as well; your machine is not running an appropriate mail *server* program (Eudora is a client, not a server). Although some PC- or Macintosh-based Internet applications can be configured as servers (e.g., there are mail, ftp, gopher, and WWW server applications available for PCs and Macintoshes), most users only run the client side of these applications. As a result, without a server process running on a system, a network connection to that system cannot be made by another machine.

41. OK. I now understand the difference among client/server, peer-to-peer, and servent. And I even see how they are different than the earlier computer communications models of decentralized, centralized, and distributed. What I still don't know, though, is where the World Wide Web fits into all of this.

The World Wide Web (WWW or Web for short) is yet another networking model that emerged as a result of the Internet. From a communications model perspective, the Web can be viewed as a collection of Internet-based clients and servers that speak the same language, namely, the Hypertext Transfer Protocol (HTTP). The Web is accessible through Web browsers such as Netscape's Communicator or Microsoft's Internet Explorer. With the introduction of a programming language called Java in the mid-1990s by Sun Microsystems, animated and interactive Web pages can be designed in addition to the standard fare of displaying text and images. A direct result of the Web's evolution is a Web-based communications model that fosters the notion of *The network is the computer*, a phrase coined in the late 1980s by Sun Microsystems' president, Scott McNealy. "The network is the computer" implies that by making resources available to users via a network, the network essentially becomes the single, most powerful computer accessible. Thus, the network gives users access to more computing power than their desktop models. Although this communications model is still client/server based, it deserves separate recognition because of its potential for reshaping the manner in which resources are provisioned to the end user.

42. Is this the networking model that is used by all of those speciality networking devices you see today like dedicated e-mail devices?

Yes, and for the record, those devices are formally called *network appliances*. (Recall from an earlier response we stated that an appliance was another term commonly used to denote a network node or device.) Other names commonly used include *netappliances*, *information appliances*, or *Internet appliances*. Unlike traditional laptop or desktop computers, network appliances usually support a single, dedicated application such as Web browsing or e-mail, and in some cases do not have keyboards or monitors. Nevertheless, network appliances are powerful computing devices designed for the average consumer who wants Internet access but does not want to be burdened with the attending problems and maintenance issues often related to personal computer ownership.

Network appliances usually rely on *application service providers* (ASPs) (also known as *content service providers* or CSPs) to furnish users with computing resources via the Internet. Examples of network appliances include: handheld devices such as the *PalmPilot*; two-way "smart pagers" that can send and receive e-mail and provide Web access; "smart phones," which are specially designed corded desktop phones or wireless devices that provide standard voice communications as well as Internet access for e-mail service and Web browsing; and TV/set-top boxes such as *WebTV*, which make Web access and e-mail service available to consumers via standard television sets. Other network appliances currently being developed or deployed as of this writing include: automobile dashboard-installed Internet connections and service; refrigerators with an in-door screen-based PC and modem for Internet access, which will enable the device to monitor food quantities and automatically order food from Internet-based grocers; and Internet-enabled fax machines that can send and receive e-mail. (See http://devices.internet.com for additional information about the rapidly evolving consumer network appliance marketplace.)

43. Would you consider a *network computer* a netappliance?

Yes. A *network computer* (NC) is a business-based network appliance instead of a consumer-based appliance. A NC promotes the concept of *thin client computing*. It is an inexpensive ($500 or less) network access device with functionality that allows some applications to be run, but not as complete as what would typically be found on a PC or a workstation of some sort. Notice that we use the term "device" here.

Although the term "computer" is part of their name, NCs are not computers; they have a specialized, proprietary (and highly restricted) operating system and are usually diskless (i.e., most have no hard disk drives for local storage). NCs are stripped-down systems that use the network to access their applications dynamically. For example, if you need a word processor, a copy of a word processing application is downloaded from a network server to your NC and stored in its memory (RAM) for you to use. Any documents you create are uploaded to and saved on the server. The idea behind NCs is to offer businesses a tremendous reduction in cost-of-ownership for each desktop location where a more expensive traditional terminal or PC would otherwise be used. By incorporating a massive server, or server "farm," with user NCs, companies can save money compared with purchasing fully-loaded PCs for each user and dealing with their management and maintenance.

44. Given what I've been able to understand so far, this sounds like centralized computing to me. Is this correct?

In a sense it is—what goes around comes around. What do you call a computing device that relies on the network for its application? A terminal. The NC concept is very reminiscent of the era when terminals (dumb or otherwise) were connected to a mainframe. (The MIS people called this a network; we know better now.) It also is similar in concept to diskless UNIX workstations and X-terminals.

45. OK. What's next?

Since we just discussed the various communication models, this is probably a good time to extend our discussion to include the various communication service methods and data transmission modes. We'll begin with *serial* and *parallel communication*.

46. I am pretty familiar with these two terms already. Serial communication means sending data one bit at a time; parallel communication means sending data in parallel, like eight bits at a time. Is this right?

Yes. Should we skip these terms then?

47. No. Go ahead and review them for me. It can't hurt.

As you said, *serial* communication (also referred to as *serial transmission*) is a data transmission method in which the bits representing a character of data are transmitted in sequence, one bit at a time, over a single communications channel. Serial transmission is limited to the speed of the line. *Parallel* communication (also called *parallel transmission*) refers to the simultaneous transmission, each on a separate channel, of all the bits representing a character. In contrast to serial communications, a parallel link transmits a group of bits at one time. The number of bits varies from device to device. Consequently, assuming the line speeds were the same, in the same amount of time required to transmit one bit of information to a remote node over a serial line we can transmit eight bits (or more) of data over a parallel line. (See Figure 1.6.)

Serial ommunication

0 →1 → 0 → 0 → 0 → 1 → 0 → 1 →

(1 channel transmits 1 bit at a time)

Parallel ommunication

0 →
1 →
0 →
0 →
0 →
1 →
0 →
1 →

(8 parallel channels transmit 1 bit)

FIGURE 1.6 Serial and parallel communications. Here, the character E, which is 01000101 in binary, is transmitted 1 bit at a time (serial communication) and 8 bits at once in parallel.

48. If parallel communication is so much faster than serial communication, why are most network links serial-based?

Although parallel communication is capable of transmitting data more quickly than serial communication, it does have its limitations. For instance, parallel communication requires a relatively complex communication link, which is achieved through the use of large, multiwire cables. Also, the longer the parallel link, the worse the degradation of the electrical signal from the most distant nodes. Consequently, in most networking applications, parallel communication is limited to peripherals directly connected to a system and for communication between systems that are relatively close to each other (in many cases, within a few yards or meters of each other). Serial communication, on the other hand, with its simpler data path, is slower but enables data transmission to occur over existing communications systems that were not originally designed for such transmission. As a result, serial communications are seen nearly everywhere, including in terminal-to-systems connections, via leased phone lines for data transfers, dialup lines, and satellite links.

49. Wait a minute. How does the receiver of a serial transmission know when a complete unit of data has been received? For example, in Figure 1.6, a single character, E, was transferred. What happens if a second character (say, X) is transferred immediately after the first? How does the receiving node identify the beginning (or ending) of a character when all it is seeing is a stream of bits (0s and 1s)?

That's a good question. Obviously, without some way of identifying the beginning (or ending) of a character, the transmitted data would be indecipherable, resulting in some sort of communication breakdown. Two methods can be employed to resolve this problem: We can either synchronize the sending and receiving nodes so that the receiving node always knows when a new character is being sent, or we can insert within the bit stream special "start" and "stop" bits that signify the beginning and end of a character. The former technique is called *synchronous* communication and the latter is called *asynchronous* communication. Synchronous communication is also tied to the clocking inherent on the link.

50. Oh yes. I recall these two terms, but I always have trouble remembering which is which. Can you help?

You bet. *Synchronous communication* implies that communication between two nodes is monitored by each node. That is, all actions resulting in data transmission (and general link conditions) are closely synchronized between the nodes. If data are to be transmitted or received, then the nodes are aware of this transmission almost immediately and prepare for the exchange based on ordered data rates and sizes. Thus, the sending and receiving nodes are "in sync" with each other. *Asynchronous communication* (commonly referred to as *async*) is characterized by the encapsulation of data within special *start* and *stop bits*. Hence, asynchronous communication is sometimes called *start-stop transmission*. A direct consequence of the inclusion of these start-stop bits in the bit stream is that data can be transferred at any time by the sending node without the receiving node having any advance notification of the transfer. Thus, a receiving node does not necessarily know when data are being sent or the length of the message. An example of async communications is a computer terminal (sender) connected to a system (receiver). The system does not know when someone will begin entering data on a terminal. As a result the system must always be in a "ready" state. Async communication lines remain in an idle state until the hardware on the line is ready to transmit data. Since the line is idle, a series of bits have to be sent to the receiving node to notify it that there are data coming. At the conclusion of a transmission, the node has to be notified that the transmission is complete so that it can return to an idle state, hence the stop bits. This pattern continues for the duration of the time the link is operative. We like to view the difference between these two terms from the perspective of a mugging on a television crime drama. Are you interested?

51. In getting mugged, no. In hearing how they can be related to a mugging, yes.

In an asynchronous mugging, you know when the actor is going to be attacked and hence are ready for it, but you do not know when it will occur. In a synchronous mugging, you not only know the actor is going to be mugged, but you also know when, so again, you are ready. Note that the term asynchronous is commonly used in the context of distance education. Through distance education technologies, education can be delivered "asynchronously," namely, at any time or place.

52. Where is each type of communication found with respect to networks?

A variety of both types of communications is found on most computer networks. Most terminals, dialup modems, and local links are asynchronous in nature. Synchronous communication tends to be more expensive than asynchronous because of the need for sophisticated clocking mechanisms in the hardware. However, synchronous communication can eliminate up to 20 percent of associated overhead inherent in asynchronous communication. (This overhead percentage is easy to compute. If we need two bits—one start and one stop bit—for each eight bit character transmitted on a serial link, then two out of every ten bits are "wasted" on overhead functions.) This allows for greater data throughput (i.e., the amount of real data that can be transferred in a given period) and better error detection. Synchronous communications are typically seen in higher speed connections.

53. What about *isochronous communication*? I'm starting to hearing a lot about it. What is it and how does it relate to the synchronous and asynchronous communications?

Isochronous communications is a term used to describe the delivery of time sensitive data such as voice or video transmissions. Networks that are capable of delivering isochronous service preallocate a specific amount of bandwidth over regular intervals to ensure that the transmission is not interrupted. Isochronous communications was originally intended to service the requirements of constant and complete delivery of video communications over a transmission medium. For example, television signaling in the United States requires that 30 frames per second of video be delivered to the receiver for full motion video. Not 29, not 31, but exactly 30. By establishing in the communications path that a session is going to require a specific data transmission rate on the path and also communicating what the data rate will be, a continual and uninterrupted flow of data can be realized. This is critical for the delivery of applications such as video, which requires a constant-bit rate (CBR) of information to be sent and delivered over the communications interface. Isochronous communications makes this happen. Networks such as ATM, SONET and a special full-duplex version of Ethernet have isochronous capabilities. To extend the previous mugging analogy, isochronous is like being mugged all the time, but the mugger and muggee have agreed on the mugging interval and constancy before the mugging actually started.

54. All this talk about serial, parallel, synchronous asynchronous, and isochronous communications reminds me to ask about simplex and duplex communications.

Serial, parallel, synchronous, asynchronous, and isochronous communications represent different techniques for transferring data. Associated with these techniques are three different modes of data transmission used for communication purposes; each corresponds to a specific type of circuit—simplex, half-duplex, and full-duplex. These modes specify the protocols sending and receiving nodes follow when transferring data. Figure 1.7 contains a summary of these three transmission modes.

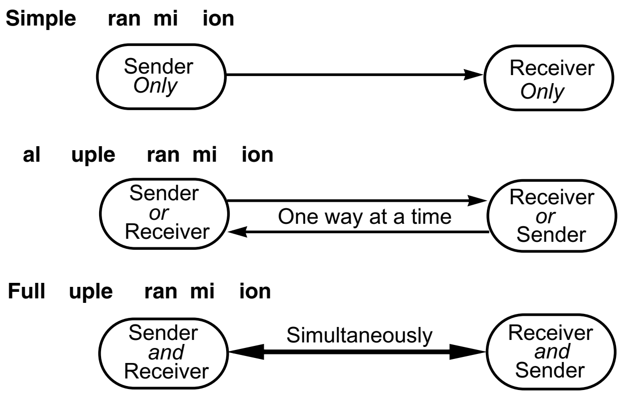


FIGURE 1.7 Simplex is a "fixed" one-way transmission; half-duplex is a two-way transmission, but only one way at a time; full-duplex is a simultaneous two-way transmission.

55. Would you please review these for me?

Simplex communications imply a simple method of communicating. In a simplex communications mode, data may flow in only one direction; one device assumes the role of sender and the other assumes the role of receiver. Furthermore, these roles may not be reversed. An example of a simplex communication is a television transmission—the main transmitter sends out a signal (broadcast), but it does not expect a reply since the receiving devices cannot issue a reply back to the transmitter. It's like a very boring person telling you his or her life story and you can neither interrupt it nor get away from it. . . . Sigh!

In *half-duplex* transmission, data may travel in either direction, but only one unit can send at *any one time*. While one node is in send mode, the other is in receive mode. Half-duplex communication is analogous to a conversation between two polite people—while one talks, the other listens, but neither talks at the same time. An example of a half-duplex communication is a citizens band (CB) transceiver. Users of a CB transceiver can either be senders or receivers but not both at the same time. Another example of half-duplex communication is like a very polite game of tag between two people. Only two can play, only one can be "it" at a time, and you know who is going to be "it" next, don't you?

A *full-duplex* transmission involves a link that allows simultaneous sending and receiving of data in both directions. Imagine, if you can, two people talking at the same time and each one understanding the other one perfectly. Compound this idea with the added benefit of not having to talk about the same thing. This is the realization of full-duplex communications—two separate but parallel transmissions occurring simultaneously. A full-duplex line can be thought of as the combination of two simplex lines, one in each direction.

56. OK. I am starting to get a headache and really need some time to assimilate all of this material. Before we take a break, though, I would like to know something about network standards.

A veritable plethora of network standards has been developed defining such things as hardware interfaces, communication protocols, and network architectures. Network standards establish specific rules or regulations to be followed. Standards also promote interoperability among different hardware and software vendors' products.

57. How are network standards created and by whom?

Standards are developed in several ways. First, they can be developed through formal standards organizations. These organizations can be classified into four major categories: (1) National, (2) Regional, (3) International, and (4) Industry, Trade, and Professional. A list of some of the influential organizations within these categories is given in Table 1.1. Standards organizations are composed of delegates from the government, academia, and from vendors who will be developing products based on the proposed standards.

58. How long does it take to get a standard approved?

The formal standards process, which is designed to ensure that a consensus is reached, is often lengthy and sometimes can take years before a proposed standard is approved. The process also is politically charged. A summary of this process is given in Table 1.2.

TABLE 1.1 Network Standards Organizations

National Standards Organizations

(Generally responsible for standards within a nation and usually participate in that nation's international activity)

- American National Standards Institute (ANSI)
- British Standards Institute (BSI)
- French Association for Normalization (AFNOR)
- German Institute for Normalization (DIN)

Regional Standards Organizations

(Restrict their activity to a specific geographical region but generally influence standards outside their regions)

- Committee of European Posts and Telegraph (CEPT)
- European Committee for Standardization (CEN)
- European Computer Manufacturers' Association (ECMA)

International Standards Organizations

(Promote standards for worldwide use)

- International Standards Organization (ISO)
- International Telecommunications Union (ITU)—consists of ITU-T, which is responsible for communications, interfaces, and other standards related to telecommunications; and ITU-R, which is responsible for allocating frequency bands in the electromagnetic spectrum for telecommunications, and for making recommendations relating to radio communications. (Note: ITU-T is the former CCITT—Consultative Committee for International Telephony and Telegraphy.)

Industry, Trade, and Professional Standards Organizations

(Restrict their activity to member interest areas but generally influence other areas)

- Electronic Industries Association (EIA)
- Telecommunications Industries Association (TIA)
- Institute for Electrical and Electronics Engineers (IEEE)
- Internet Engineering Task Force (IETF)

Source: Adapted from Conrad, 1988.

59. What kind of standards are there?

Standards can be viewed from four different perspectives. There are *de jure* standards, *de facto* standards, *proprietary* standards, and *consortia* standards.

60. Wow. I didn't realize there were so many. Could you summarize them for me?

Sure. *De jure* standards are approved by a formal, accredited standards organization. ("De jure" is Latin for "by right, according to law.") Examples include modem protocols developed by the International Telecommunications Union (ITU), the EIA/TIA-568 Standard for Commercial Building Telecommunications Wiring developed by the Electronic Industries Association (EIA) and Telecommunications Industries Association (TIA), and standards for local area networks developed by the Institute for Electrical and Electronic Engineers (IEEE). (*Note*: These standards are discussed in subsequent chapters.)

TABLE 1.2 The Network Standardization Process

Planning Phase
- Proposals submitted by vendors or organizations are examined.
- A determination is made whether there is a need to establish a standard.
- If a need is found, the development of a project is authorized and assigned to a technical committee.

Development Phase
- Committee prepares a working paper describing the scope of the proposed work.
- Liaisons with other standards groups are established.
- A draft proposal of the standard is produced.
- Draft is voted on and all negative comments are addressed.
- Draft is submitted to parent organization for discussion and approval.

Approval Phase
- All members of the organization vote on draft.
- Draft is made available to the public for review.
- Draft is ultimately approved as a standard.

Publication Phase
- The new standard is published.

Source: Adapted from Conrad, 1988.

De facto standards are those that have come into existence without any formal plan by any of the standards organizations. Rather, they are developed through the industry's acceptance of a specific vendor's standard, which is placed in the public domain. ("De facto" is Latin for "from the fact.") One example is Network File System, (NFS), a de facto file-sharing protocol standard developed by Sun Microsystems. Sun placed the specifications of this protocol in the public domain so that other vendors could implement it. This resulted in widespread use of NFS and established NFS as a de facto standard. NFS is now implemented on a variety of UNIX systems (including those from Sun, IBM, Silicon Graphics, and HP), as well as Macintosh and Intel-based systems. Another de facto standard is Java, which was also developed by Sun Microsystems.

Proprietary standards are those developed in a manufacturer-specific manner. This implies that their specifications are not in the public domain and are only used and accepted by a specific vendor. In the early days of networking, proprietary standards were the rule of the day. Although such standards are now frowned upon, many still exist. Some of the most well-known are from IBM (e.g., IBM's Systems Network Architecture or SNA). Novell's IPX protocol, which is based on Xerox's XNS protocol, is also proprietary in nature. Proprietary standards lock a customer into a vendor-specific solution and make it difficult for customers to use products (software or hardware) from other vendors (see Box 1.1).

Consortia standards are similar to de jure standards in that they too are the product of a formal planning process. The difference is the planning process and development of such standards are not conducted under the auspices of a formal standards organization. Instead, specifications for standards are designed and agreed upon by a group of vendors who have formed a consortium for the express purpose of achieving a common goal. These vendors pledge their support for the standards being developed by the consortium and also develop

BOX 1.1: Open Standards versus Closed Standards

- Open is for everyone.
- Closed is vendor specific.
- Open means that everyone has a chance to implement and benefit from the same standards.
- Closed means that the vendor feels that its standard has value and will not share the technology with other vendors.
- Open allows for the creation and modifications of standards by committee.
- Closed means that the vendor can effect repair or modifications without agreement of other vendors and without comment from customers.
- Open implies that any modifications take a long time.
- Closed implies that modifications are made in a timely manner but they are nonstandard.

and market products based on these mutually agreed upon set of standards. An example of consortium-based standards include Fast Ethernet, the early efforts for Asynchronous Transfer Mode (ATM Forum), and Gigabit Ethernet, all of which are discussed in later chapters of this book.

61. What about Internet standards? You listed the Internet Engineering Task Force in Table 1.1 but didn't say anything about how these standards are developed.

You're right. This is because the Internet Engineering Task Force (IETF) is not a traditional standards organization; it only recommends the standardization of protocols and protocol usage in the Internet. We listed IETF in Table 1.1, though, because many of the protocol specifications IETF produces become standards.

Internet standards are initially developed by IETF, which, according to The Tao of IETF (see http://www.ietf.org/tao.html), "is a loosely self-organized group of people who make technical and other contributions to the engineering and evolution of the Internet and its technologies." The Internet standards development process involves the generation of special documents called *Request for Comments* (RFCs), which initially were written comments about resolving certain Internet-related problems. Today, however, RFCs are formal documents that comprise two sub-series. The first contains For Your Information (FYI) documents, which provide, in a less technical manner, general overviews and introductory information about various Internet topics. The second sub-series, STD, references those RFCs that specify Internet standards. Before an RFC is published, it is first developed as an Internet Draft, which enables the Internet community to read and comment on proposed Internet-related documents before they are officially published as an RFC. Internet Drafts are considered temporary documents and have a shelf life of only six months; hence, they are not archived. To facilitate the dissemination process and to maintain a spirit of openness, RFCs and current Internet Drafts are available online at http://www.rfc-editor.org. It should be noted that approved Internet standards are promoted internationally by ISO.

However, corresponding RFC-STDs never become international standards, although the ISO may take information from them. For more information about the Internet standards process, see RFC 2026, “The Internet Standards Process—Revision 3.”

62. As a network manager, how do I know which standard to accept?

Obviously, you want to try to avoid proprietary standards and adopt de jure standards. Unfortunately, this is not so easy to do because standards are starting to fall victim to the relatively short life cycle of a technology. Even worse, the standards organizations (especially the big ones like ISO, ITU, ANSI, IEEE and others) must reaffirm, remove, or change a standard within five years of its creation. This can result in multiple versions of a standard, depending upon which year is being addressed. Further, a standard developed today for directory services (which is known in the business as X.500) may be completely rewritten and different four years hence when the next meeting of the ITU comes around to discuss the standard. A concept for a standard exists for a long time; however, the actual technical detail may last a short time, depending upon the standard.

63. What do you mean by this?

Consider the changes in modem technology. Within a 36-month period in the 1990s, data transmission rates for modems increased from 9600 bits per second (bps) to 14,400 bps to 28,800 bps to 33,600 bps to 56,000 bps. The standards on which each new modem technology was based were originally proprietary. Although users (in most cases) were given vendor assurances that their modems would be compliant with the forthcoming de jure standard, users still faced a purchasing dilemma: Should I invest in the newer technology now, even though it is proprietary, or should I wait for the technology to be approved by a formal standards organization before I commit my resources?

64. That’s a good question. What’s the answer?

Unfortunately, there is no easy answer to this question. You should be cognizant of the relatively short life cycle of technology and understand that technology will always experience different stages of maturity as it evolves. This situation, coupled with the lengthy process of formal standardization, means we cannot rely solely on de jure standards to achieve interoperability. The realistic approach to achieving interoperability will most likely involve a combination of de jure, de facto, proprietary, and consortia standards.

END-OF-CHAPTER COMMENTARY

The terms discussed and defined in this chapter serve as the basis for understanding many of the concepts presented in the remaining chapters of the book. These terms will be further expanded and additional terms will be defined as new concepts are introduced. In the event that you need a quick review of these or any other terms, see the glossary. We now turn our attention to Chapter 2, which addresses the subjects of network topologies, architectures, and the infamous OSI and TCP/IP models.

This Page Intentionally Left Blank

Chapter 2

Network Topologies, Architectures, and the OSI Model

In this chapter we present several fundamental networking concepts. We begin with an overview of network topologies, giving examples of the two most general designs on which most topologies are based: point-to-point and broadcast. Next, we examine the concept of network architecture. As part of this discussion we introduce the idea of "layering" and present the Open Systems Interconnect (OSI) networking model as an example of layered architecture. We conclude the chapter with an introduction to TCP/IP and show how it relates to the OSI model. An outline of the material we discuss follows:

- Concept of Network Topologies (Questions 1–2)
- Point-to-Point Networks: Star, Loop, Tree (Questions 3–9)
- Broadcast Networks: Bus, Ring, Satellite (Questions 10–22)
- Multidrop Networks (Question 23)
- Circuit- and Packet-Switched Networks (Questions 24–32)
- Network Architectures (Questions 33–34)
- The OSI Model (Questions 35–41)
- Connection-Oriented and Connectionless Services (Questions 42–46)
- The TCP/IP Model (Questions 47–52)

1. What is a network topology?

A *network topology* is the basic design of a computer network. It is very much like a road map. It details how key network components such as nodes and links are interconnected. A network's topology is also comparable to the blueprints of a new home in which components such as the electrical system, heating and air conditioning system, and plumbing are integrated into the overall design.

2. In which ways can nodes and links be interconnected?

There are three general interconnection schemes: *point-to-point*, *broadcast*, and *multidrop*.

3. Start with point-to-point. What's that?

A *point-to-point network* consists of nodes that can only communicate with *adjacent nodes*. It's like looking into a telescope and seeing only one planet out the eyepiece.

4. What do you mean by adjacent nodes?

Adjacent nodes are nodes that are next to each other (Figure 2.1). Adjacency is typically expressed by stating the number of *hops* required for data to travel from the source node to the destination node. A *hop* is a connection to or from an intermediate node on the path from the source to the destination. Adjacent nodes are always one hop from each other. Thus, one hop implies two directly connected (*line-of-sight*) nodes. In a more complex form, a point-to-point network might consist of thousands of nodes connected to adjacent nodes, with these adjacent nodes connected to other adjacent nodes, and so on.

5. What happens if a node needs to communicate with a nonadjacent node?

It does so indirectly via other adjacent nodes. The source node first transmits a message to its adjacent node. This message is passed serially through each intervening node until it finally reaches the destination node. Passing data through an adjacent node to another node is typically called *bridging* or *routing*, depending on the passing technique used to transfer the information.

6. Which network topologies are based on the point-to-point design?

There are several. Three very common ones are *star*, *loop*, and *tree*.

7. What does a star topology look like?

A *simple star* configuration is shown in Figure 2.2(a). A key characteristic of a star is the presence of a central processing hub, which serves as a wire center for connecting nodes. In order for nodes to communicate with each other, all data must pass through the hub. Consequently, a hub represents a single source of failure. A typical star configuration is shown in Figure 2.2(b). This is a 10BASE-T network (a type of Ethernet) consisting of nodes connected to an "Ethernet switch" via unshielded twisted-pair cable (UTP). (10BASE-T networks, Ethernet switches, and UTP cable are discussed in detail in subsequent chapters.)

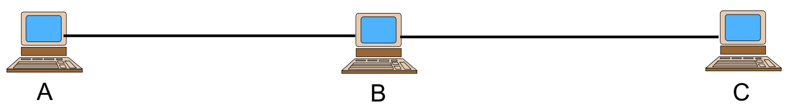


FIGURE 2.1 Example of a point-to-point network design. A characteristic of point-to-point networks is adjacency—nodes can only communicate with nodes they are next to. Thus, node A can only communicate with node B, and node C can only communicate with node B. If nodes A and C need to communicate with each other, they do so using node B, which is adjacent to both A and C.

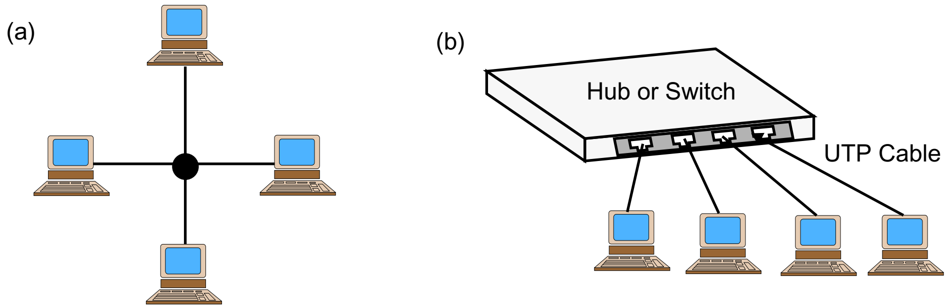
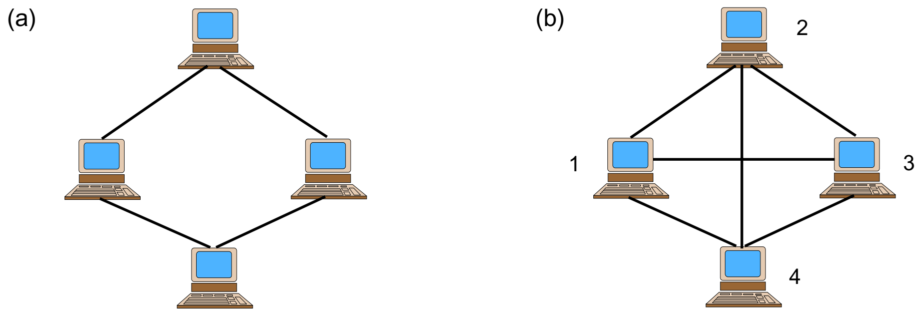


FIGURE 2.2 A simple star configuration (a) involves a wiring center (or hub) to which all nodes are connected and through which all data must pass. The hub represents a single source of failure because, if it fails, then all connected nodes will not be able to communicate. A typical hub configuration is shown in (b).



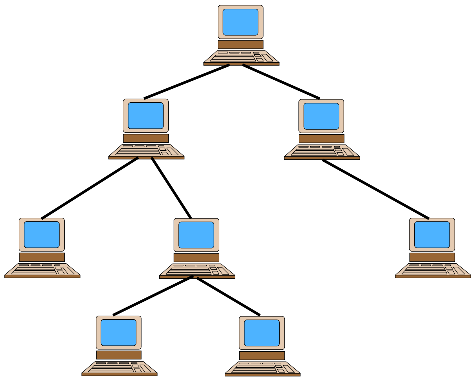
IGURE 2.3 A loop design (a) is a modified star configuration. Instead of using a wiring hub, nodes re connected directly by dedicated wiring. If every node has a link to every other node, then we have complete loop (b). (A complete loop is also called fully meshed.) Note that in a complete loop the umber of links each node has is one fewer than the number of nodes in the network. Loops are more eliable than stars because the potential for a single source of network failure is removed.

8. What does a loop look like?

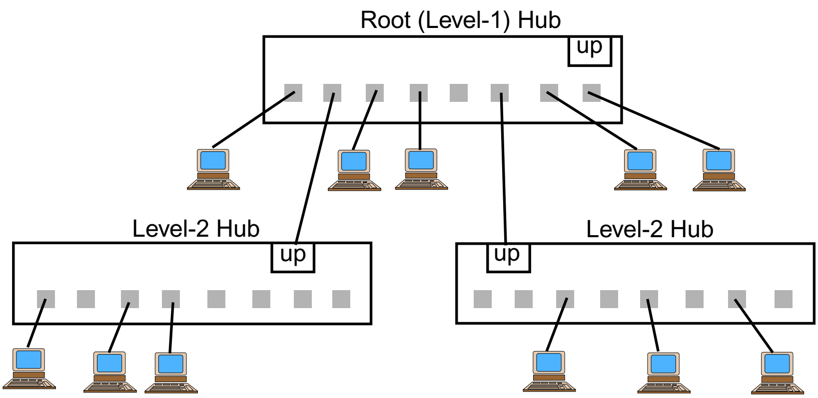
A *loop* is a modified version of a star. In a loop, nodes are connected via dedicated wiring instead of through a centralized hub. An example of a simple loop is shown in Figure 2.3(a). This involves only one connection between any two nodes. Note that a single link failure does not cause the entire network to fail. Thus, loops are more reliable than stars. A highly reliable and more expensive loop design involves each node being connected to every other node. This is called a *complete loop* and is shown in Figure 2.3(b). Note that in a complete loop every node is adjacent to every other node. A complete loop is also referred to as a *fully-meshed* design.

9. What about a tree?

A *tree* topology is a hierarchical configuration. It consists of a root node or hub that is connected to second level nodes or hubs. These "level 2" devices are connected to "level 3" devices, which in turn are connected to level 4 devices, and so forth. A simple tree topology is shown in Figure 2.4. One application of a tree topology is IEEE 802.12, known as 100VG-AnyLAN, in which hubs are cascaded to form a hierarchical topology. An example of this network is shown in Figure 2.5.



IGURE 2.4 A simple tree topology consists of nodes interconnected in a hierarchical con-guration.



IGURE 2.5 An example of a two-level hierarchical topology in a 100VG-AnyLAN network. Each ub has at least one uplink port, which connects to a higher-level hub; every port can be used as a ownlink port to connect to an end node or a lower-level hub.

10. What about broadcast. What is it and what distinguishes it from point-to-point?

A *broadcast* design consists of nodes that share a single communications channel. In a point-to-point design nodes do not share a common channel but instead are directly connected to each other.

11. Is this like a telephone party line?

Yes it is, and you are dating yourself. In contrast to point-to-point design, data sent by one machine are received by all other nodes connected to the shared channel. Hosts receiving a transmission check the destination address of the message to determine if it is intended for them. If not, they discard the message. Thus, only the destination node responds. As an illustration, consider a classroom setting with a teacher and 23 students. If the teacher asks one student, George, a question, all 23 students hear the question but only George responds. This is analogous to a broadcast network.

Compare this to point-to-point communication. Let us now assume that one student, Patti, wants to tell her friend, Janie, who is sitting two rows over and three seats behind Patti, to wait for her after class. To get this message to Janie, Patti turns to John who is sitting behind her and says, "Tell Janie to wait for me after class." John turns to the person behind him and says, "Tell Janie to wait for Patti after class." This message continues being passed from one person to the next until it finally reaches Janie. This is an example of a point-to-point design.

12. Since all nodes hear a transmission in a broadcast design, is it possible for a node to send the same data to more than one node during the same transmission?

Yes. In fact, there are three different types of messages. The first is a *unicast message*, which is destined to only one recipient. The second is a *multicast* message, which is destined to a group of recipients. It is important to note that a node "knows" that it is in a multicast group by its networking software "telling" it to listen to the multicast messages for the group. In many cases, the sending system to the multicast group does not know which nodes are actually members of the group. The third is a *broadcast* message, which is destined to all hosts connected to the network. A broadcast message is a special multicast message. (*Note*: In IPv6, discussed in Chapter 3, there is also the concept of *anycast*.)

13. How is one type of broadcast message distinguished from another?

This is protocol-dependent. Some protocol suites do not use broadcast and only use multicast. Others do not use multicast and use broadcast for group addressing needs. Would you like an example?

14. Yes. Tell me how it is done in Ethernet/802.3 networks.

An Ethernet/IEEE 802.3 address consists of 48 bits (eight bytes) that are represented as 12 hexadecimal digits (0-9, A-F) and partitioned into six groups of two. For example, 08:00:20:01:D6:2A is a valid Ethernet/802.3 address. If the second hexadecimal digit (from the left) of a destination address is 0 or an even digit (2, 4, 6, 8, A, C, E), then the message is unicast. Thus, 08:00:20:01:D6:2A is a unicast address because its second digit is 8, which is even. If this second hexadecimal digit is odd (1, 3, 5, 7, 9, B, D, F), then the message is multicast. Ethernet broadcast messages, which are special multicast messages, use the address FF:FF:FF:FF:FF:FF. See Box 5.1 and Appendix A for additional information about Ethernet/802.3 addresses.

15. Since it is possible for a node to send the same data to more than one node during the same transmission, is it also possible for more than one node to send data at the same time?

Well, they can try but they are not going to be successful. Since all nodes share the same communication channel, they must contend for the channel when they transmit. Thus, broadcast networks promote the concept of *contention* and hence require some sort of method for governing those cases when two or more nodes attempt to transmit data at the same time.

16. What sorts of protocols are there for resolving such squabbles among nodes?

There are many. We provide an overview of some of the more popular ones in Chapter 5 and describe them in more detail in subsequent chapters.

17. OK. Now that I know a little bit about broadcast designs, which topologies are based on this design?

Broadcast networks employ several topologies. They are *bus*, *ring*, and *satellite*.

18. Could you give an illustration of each of these broadcast-based designs?

Yes. A typical bus configuration is shown in Figure 2.6, a ring is shown in Figure 2.7, and a satellite is shown in Figure 2.8.

19. It's quite apparent from Figure 2.6 how a bus is a broadcast topology. It clearly shows all the nodes connected to the same channel. What about the ring and satellite configurations? How are they broadcast-based?

If you look closely at Figure 2.7 you will notice that all nodes are connected to the same ring, which serves as the shared medium. Remember, a broadcast design means all nodes share a single communications channel, and that messages sent by one node are received by all others connected to this channel.

20. How is a ring different from a loop, which is shown in Figure 2.3, and which is point-to-point? They almost look the same.

You are right. They do look similar. Looks are deceiving, though. The distinction can be made by looking at things from a logical and physical perspective. In a ring configuration, all nodes are connected to the same ring, which serves as the shared medium. Ring-based networks can be designed physically as a star (Figure 2.7a), or as a simple loop (Figure 2.7b). The star design is formally referred to as a *logical ring over a physical star*, and the simple loop design is formally referred to as a *logical ring over a physical ring*.

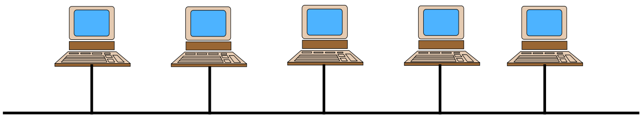
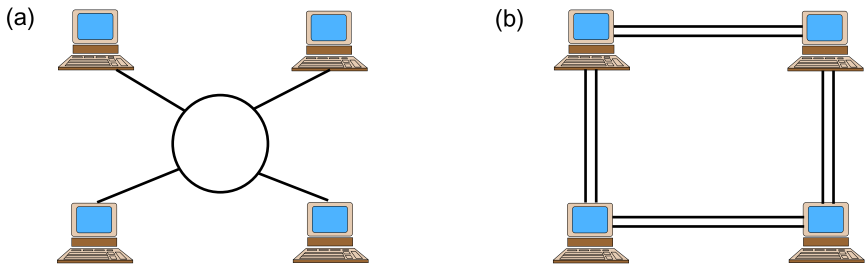


FIGURE 2.6 Typical bus configuration. An example of a topology based on a broadcast design.



IGURE 2.7 Ring-based networks represent a topology based on a broadcast design. Rings can be onfigured as (a) a logical ring over a physical star or (b) as a logical ring over a physical ring.

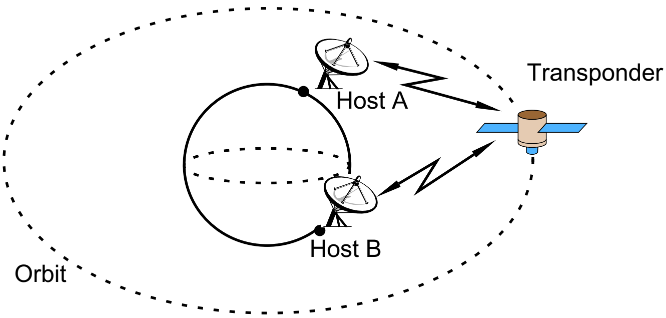


FIGURE 2.8 Typical satellite configuration. Nodes use an antenna to send and receive data.

The key difference between a ring-based "loop" (Figure 2.7b) and a point-to-point-based loop (Figure 2.3a) is that the ring-based loop is a broadcast system and hence, nodes share a single communications channel. In a point-to-point system, however, we have the concept of adjacency and hence nodes can only communicate with the node next to them. If you examine the loop topology in Figure 2.3a, you will notice that the nodes do not share a single channel. There is a specific, dedicated link between any two nodes.

21. How does a ring work?

In a classic ring topology, messages are passed from node to node around the ring. The direction of rotation can either be clockwise or counterclockwise (or both), depending on the technology. As is the case for the bus topology and for all broadcast systems, some method must govern simultaneous ring access. Note that although data are passed from node to node, this is not a point-to-point topology because all the nodes share the same communications channel. Thus, logically, the topology of a ring involves all nodes sharing the same communications channel. Physically, though, the communications are point-to-point.

22. OK. Now what about satellite networks? What makes them broadcast systems?

In a satellite communication system, data transmissions from a land-based antenna to the satellite are generally point-to-point. However, all nodes that are part of the network can receive the satellite's *downlink* transmissions. (A downlink is the communication link from the orbiting satellite to one or more ground stations.) This classifies them as broadcast systems. For example, many schools in the United States have satellite downlink capabilities. Whenever an educational program is broadcast via a satellite system, school sites wishing to receive this transmission simply tune their receivers to the proper frequency.

23. What's a multidrop network design?

In some types of networks, especially factory networks and those used to control real-time activities like power company networks, a particular design concept called a *multidrop* network is frequently used. Multidrop networks typically employ a master–slave concept with one node being assigned the network master and all other nodes being slaves. In this setting, the master controls the network functions and the slaves request network access from the master. Nodes are connected to a common cable plant similar to a bus design, but unlike bus networks multidrop nodes are assigned a specific number for communication purposes. This number also is used to establish priority of when a system is permitted to communicate with the master control system. This allows total control over the prioritization of traffic on the network as well as total control over the use of the network. Multidrop networks are popular only in factories because they are not terribly fast networks and would not work well in offices where users might want to share large disk drives and applications. They are typically used for command-and-control operations and some light data transfer of material information or tracking information (like bar codes). Multidrop networks are also often seen on older (circa early 1980s), legacy dumb terminal networks to reduce network costs. An illustration of a multidrop design is given in Figure 2.9.

24. What other network classifications are there?

In addition to geographical area and topology, networks also can be classified by the type of communications path they use and the manner in which data are transmitted across this path. Two particular classifications are *circuit-switched* and *packet-switched*. Switched networks involve a partially- or fully-meshed topology (see Figure 2.3) and use special network devices called switches to interconnect the links between source and destination nodes.

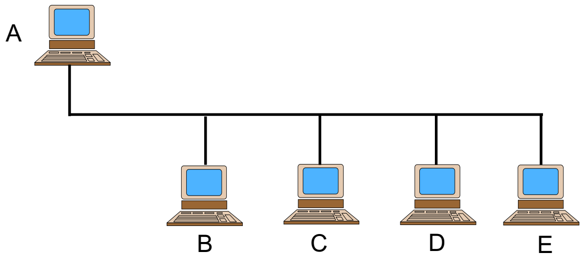


FIGURE 2.9 In a multidrop network topology, all nodes are connected to a common medium. Using a master–slave concept, where one node is the designated master and all other nodes are slaves, the master controls all network functions and the slaves request network access from the master. In this illustration, node A is the master, and nodes B, C, D, and E are slaves.

25. What's the difference between the two types of switched network?

In a *circuit-switched* network, a dedicated, physical circuit is first established between the source and destination nodes before any data transmission takes place. Furthermore, this circuit remains in place for the duration of a transmission. The public telephone system, known formally as the Public Switched Telephone Network (PSTN), is an example of a circuit-switched network. When we place a telephone call, a direct physical communications path is established between our telephone set and the receiver's telephone set. This path is a point-to-point connection that interconnects the telephone company's switches, which are located throughout the telephone network. Once established, the circuit is dedicated exclusively to the current transmission. After the transmission is completed, this dedicated circuit is then released and made available for another communication transmission. Thus, circuit-switching promotes link sharing since the same circuits can be used for different transmissions, although not at the same time.

In a *packet-switched* network, messages are first partitioned into smaller units called *packets*, which are then sent to the destination node one at a time via intermediate switches. A packet represents the smallest unit of data that can be transferred within a given network. Each packet carries the destination node's address as well as a sequence number. When a packet arrives at an intermediate switch, the switch examines the packet's destination address to determine which path the packet should take to the next switch. This switching technique in which data are stored on one node of a point-to-point link and then forwarded to the next node repeatedly en route to the destination node is called *store-and-forward*. The concept of a store-and-forward transmission requires that the entire contents of a transmitted message (or packet) be received by each intermediate node before it is forwarded to a succeeding node.

26. So circuit-switching sets up a link between the two end nodes and then transmits the data across this link. Packet-switching, on the other hand, partitions the data into packets and then transmits the packets. Is this right so far?

Yes.

27. If this is the case, then on what circuit are the packets transmitted in a packet-switching environment? How can the packets be transmitted if there is no link established between the two end nodes as there is with circuit-switching?

You're right, and something is missing. We need to mention that packet-switched networks also promote the concept of link sharing, which can be effected in two ways: Link sharing can be accomplished by using *virtual circuits* or a *datagram transport* scheme. In *virtual-circuit packet-switching*, instead of using a dedicated, physical circuit for every node-to-node communication (as is the case with circuit-switching), nodes share a communications channel via a virtual circuit. A virtual circuit is a *logical* communications path instead of a physical one. That is, it is a nondedicated logical connection through a shared medium that gives the high-level user the appearance of a dedicated, direct physical connection from the source node to the destination node. In this transport mechanism, individual packets follow the same communications path in sequence as if they were traveling along a dedicated circuit. The difference is that links within this circuit also can be used for other transmissions at the same time.

28. Wait a minute. Before we go any further, I need an example.

OK. Let's compare circuit-switching to virtual-circuit packet-switching. As an illustration of circuit switching, consider the fully meshed, point-to-point design shown in Figure 2.3(b). If a communications path between nodes 1 and 3 is established by interconnecting links 1-2-4-3, then these links collectively form a *dedicated*, *physical circuit* between the two nodes, and this circuit is monopolized by these two nodes for the duration of their communications. However, once their transmission ends, the circuit is released and the individual links that made up this circuit can now be used to construct a new dedicated circuit between two other nodes (e.g., 4-1-2). In a virtual-circuit packet-switching environment, node 1 can be communicating with node 3 along the 1-2-4-3 path, and at the same time, node 4 can be communicating with node 2 along the 4-1-2 path. Both paths represent virtual circuits that simultaneously use the 1-2 link. Contrast this with circuit-switching in which the circuit is dedicated and the individual links comprising the circuit cannot be used simultaneously to construct any other circuit.

29. Oh! I get it. So that's the difference between a dedicated link versus link sharing. OK. Now what about this datagram transport scheme you mentioned?

As we stated earlier, in addition to virtual circuits, packet-switched networks can also use a datagram transport mechanism for path selection. In *datagram packet-switching*, packets are transmitted independently of one another at any time. Thus, it is possible for different packets from the same message to be transported across different communications paths. Furthermore, the packets are not necessarily transmitted in a specific order, which implies that the specified destination node is responsible for reassembling them in the correct order. (This is why packets also carry sequence numbers.) Most modern-day computer networks, including the Internet, are packet-switched.

30. How are virtual circuits formed?

A virtual circuit is created by *multiplexing* a physical link so that it can be shared by multiple network programs or data transmissions. We will not go into the details of multiplexing here. If you're really interested in it at this stage, though, feel free to jump ahead to Chapter 4 where it is discussed.

Multiplexing is extremely valuable for providing low-cost communications capabilities because it is very expensive to provide dedicated links for every data transmission, as in circuit-switched networks. A definition of *virtual* has been memorably coined in this way: "If you can see it and touch it, it's *physical*; if you can see it but can't touch it, it's *virtual*; if you can't see it and can't touch it, it's *gone*."

31. So, is the main difference between circuit-switching and packet-switching the type of link that is used, namely, one is dedicated and the other is virtual?

Not quite, virtual reader. The main difference between circuit- and packet-switched networks is the use of *bandwidth*, which is the maximum capacity of a communications channel. For example, a circuit might have a data transmission capacity of 100 megabits per second. We will discuss the concept of bandwidth in more detail in Chapter 4. For the present, though, just understand that in a circuit-switched network a circuit's performance is predetermined and fixed. This means that bandwidth is allocated in advance and guaranteed for the entire transmission. Once a circuit is established, the full capacity of the circuit is available and the capacity of the circuit will never be reduced due to other network activity. This advantage of circuit-switched networks also gives rise to a disadvantage. Specifically, circuit costs are independent of the amount of data being transmitted; therefore, any unused bandwidth is wasted.

On the other hand, packet-switched networks acquire and release bandwidth dynamically as needed. One major advantage is that several communications can occur between nodes concurrently using the same channel. Again, this advantage becomes a disadvantage when, as packet-switched networks become overloaded with more traffic, delays and congestion are introduced. Nevertheless, packet-switched networks are cheaper and offer better performance than circuit-switched networks. Furthermore, given recent developments in high-speed switching hardware, the channel capacity issue has eased a bit. Table 2.1 provides a summary of the differences between circuit- and packet-switching.

32. Given the advantages and disadvantages of circuit- and packet-switched networks, why not combine the two so you could have the best of both worlds?

You are referring to *hybrid switching*, which combines the principles of both circuit- and packet-switching. This technique first partitions a message into packets (packet-switching) and transmits each packet via a dedicated circuit (circuit-switching). As soon as a packet is ready for transmission, a circuit meeting appropriate bandwidth requirements is established between the sending and receiving nodes. When the packet reaches its destination, the circuit is terminated (i.e., it is "torn down" using telephone terminology) so that it can be used again. This scenario has many advantages but it also requires extremely fast circuit-switching equipment.

TABLE 2.1 Circuit-Switching versus Packet-Switching

Circuit-Switched	Packet-Switched
1. Bandwidth is allocated in advance and is guaranteed for the entire transmission.	1. Bandwidth is acquired and released dynamically on an as-needed basis.
2. Once circuit is established, the full capacity of the circuit is available for use, and the capacity of the circuit will never be reduced due to other network activity.	2. Several communications can occur between nodes concurrently using the virtual links over the same physical channel.
3. Circuit costs are independent of the amount of data being transmitted and hence any unused bandwidth is wasted.	3. As packet-switched networks become overloaded with more traffic, delays and congestion are introduced.
	4. Packet-switched networks are more cost-effective and offer better performance than circuit-switched networks.

Thus far in this chapter, we have examined various network topologies. Many of the designs we discussed will be reintroduced in later chapters that deal with specific networking technologies such as Ethernet and token ring. The concepts of circuit-switching, packet-switching, and multiplexing will also be re-examined in more detail in subsequent chapters. We now focus our attention to network architectures and the OSI and TCP/IP models.

33. What is a network architecture?

Network architecture is a formal, logical structure that defines how network devices and software interact and function. It defines communication protocols, message formats, and standards required for interoperability. New hardware or software products created within a specific architecture are generally compatible with other products created within the same architecture.

34. Who creates or designs network architectures?

Network architectures are designed by standards organizations and manufacturers. For example, IBM designed the Systems Network Architecture (SNA), the former Digital Equipment Corporation (DEC) designed the Digital Network Architecture (DNA), and the International Organization for Standardization, which is always abbreviated "ISO," designed the Open Systems Interconnect (OSI) architecture.

35. I've heard of these architectures before and am familiar with SNA and DNA. Tell me, what's the big deal about OSI?

Well, OSI really isn't much of a big deal anymore, although it did once have a very pronounced role in the networking community. The genesis of OSI can be thought of as follows: Once upon a time—a long, long time ago (circa 1970s)—there was no such thing as a network architecture. Companies designed rather rude, crude, and socially unacceptable proprietary software and hardware communications products without any consideration to the implementation of a coherent architecture—the long-term technical effect of decisions made when something is constructed. Eventually, issues of interoperability and design began to emerge as new networks were being developed.

To address these issues, and to accommodate interconnection of the various proprietary and heterogeneous networks, ISO developed in 1978 a seven-layer architecture and reference model intended to serve as the foundation for future standards activities. The resulting model was formally called the *Basic Reference Model for Open Systems Interconnection*, or the OSI model for short. The OSI model provides a detailed set of standards for describing a network; it is a framework for the development of network protocol standards. The OSI model formally defines and codifies the concept of *layered* network architecture. It uses well-defined operationally descriptive layers that describe what happens at each stage in the processing of data for transmission.

36. What's so great about layers?

Networks are nontrivial systems. Given a network's complex nature, it is extremely difficult to design an architecture that (a) has a high degree of connectivity, (b) is reliable, and (c) is easy to implement, use, and modify. Layers help reduce the design complexity of a network. By organizing a network's functions into a series of hierarchical layers, the design of a network is greatly simplified. For example, a layered approach enables the functions and services of one layer to be completely independent of and isolated from other layers. Thus, a layered approach effectively screens from other layers the actual implementation details of the services one layer is offering to another layer. This allows us to change a layer's capabilities without significantly modifying the entire architecture. So as new technologies become available for one layer, they can be implemented without affecting the operation of the other layers. In theory, a layer can be completely removed, dramatically changed, and reinserted without affecting the operation of the layers above or below it.

37. Is this like modular programming?

You bet it is. Just as large computer programs are partitioned into separate, independent program modules, layers partition a network architecture into separate, independent components. Each layer is responsible for performing a specific set of functions and for providing a specific set of services. Specific protocols define both the services and the manner in which these services are provided. Another analogy is an automobile, which comprises several independent components, including the electrical, braking, cooling, and ignition systems. Working on one component does not affect another component. Thus, if we need to replace our car's brakes, we do not have to be concerned with how it will impact the car's ignition system.

38. What are the layers of the OSI model?

The layers of the OSI model are (from top to bottom): application, presentation, session, transport, network, data link, and physical. These layers are numbered in descending order from seven to one and define the communication capabilities needed to effect communication between any two devices. Figures 2.10 and 2.11 provide additional information about OSI layers.

Application

¥ Consists of protocols that define specific user-oriented applications such as e-mail, file transfers, and virtual terminal.
¥ Examples include FTAM (File Transfer, Access, and Management) for remote file handling, X.400 (for e-mail), and CMIP (Common Management Information Protocol) for network management.

Pre entation

¥ Provides data formats, translations, and code conversions.
¥ Concerned with syntax and semantics of data being transmitted.
¥ Encodes messages in a format that is suitable for electronic transmission.
¥ Performs data compression and encryption.
¥ Receives/formats message from application layer and passes it to session layer.
¥ In practice, this layer is usually incorporated within the application layer.

Se ion

¥ Provides coordination between communicating processes between nodes.
¥ Responsible for enforcing the rules of dialog (e.g., Does a connection permit half-duplex or full-duplex communication?), synchronizing the flow of data, and reestablishing a connection in the event a failure occurs.
¥ Examples include AppleTalk Data Stream Protocol for reliable data transfer between two nodes, NetBEUI (an extension of NetBIOS), and Printer Access Protocol for accessing a PostScript printer in an AppleTalk network.
¥ Uses the presentation layer above it and the transport layer below it.

ran port

¥ Provides error-free delivery of data.
¥ Accepts data from the session layer, partitions data into smaller packets if necessary, passes these packets to the network layer, and ensures that packets arrive completely and correctly at their destination.
¥ Examples involve varying classes of the OSI Transfer Protocol TP *x*, where *x* = {0, 1, 2, 3, 4}. Each class describes a specific level of service quality such as whether a transmission provides for error detection or correction, or if the service is connection-oriented or connectionless.

et or

¥ Provides end-to-end routing or switching, which establishes a connection for the transparent delivery of data.
¥ Addresses and resolves all inherent problems related to data transmission between heterogeneous networks.
¥ Uses the transport layer above it and the data link layer below it.
¥ Formatted messages are referred to as *packets*.

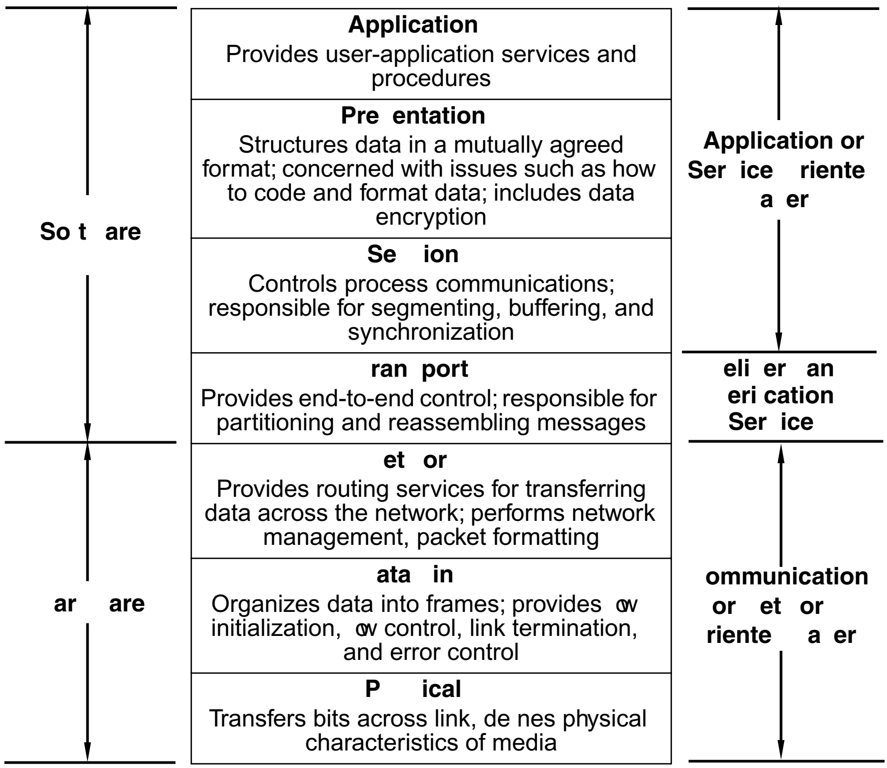
ata in

¥ Responsible for end-to-end data transfer across a physical link.
¥ Provides error detection, framing, and flow control.
¥ Resolves problems due to damaged, lost, or duplicate frames.
¥ Formatted messages are referred to as *frames*.

P ical

¥ Responsible for transmitting raw bits over a link; it moves energy.
¥ Accepts frames from the data link layer and translates the bit stream into signals on the physical medium, which lies below it.
¥ Concerned with issues such as the type of wire being used, the type of connector (i.e., interface) used to connect a device to the medium, and signaling scheme.

FIGURE 2.10 Summary of the OSI layers and functions.



IGURE 2.11 Another view of the OSI layers. Source: Adapted from Conrad, 1988b.

39. Why seven layers? Why not 5 or 20?

Seven is an interesting number. For example, there are the seven wonders of the world, the universe was created in seven days (or so we are told), there is supposed to be something mystical about being born a seventh son, seven is presumably a lucky number (although there is the slogan "seven years bad luck"), and major league baseball, basketball, and hockey have seven-game playoff series. So, why not seven? Seriously, though, the selection of seven layers was based on specific principles and design criteria. Two specific layers-related criteria were (a) a layer should be created only when a different level of abstraction is required, and (b) the number of layers should be large enough to keep distinct functions separate but not too large such that the architecture becomes too cumbersome or unmanageable. Following is the logic used in creating each layer based on the established criteria:

- The designers first agreed that a separation between applications and media was needed. This established two layers—an application layer and a media layer.
- If true global networking services were going to be provided, then the media layer required further refinement. Specifically, one layer was needed to specify the

media and corresponding signaling. This resulted in the creation of the physical layer. Another layer was needed to specify the operation of a single link. This is the role of the data link layer. Finally, a network layer was needed to provide end-to-end network service that specified the operation of end nodes as well as intermediate nodes across different links and topologies. Thus, the original media layer was supplanted by the physical, data link, and network layers. This now gives us four layers: the three "media-related" layers and the application layer.

The media-related layers effectively addressed technology issues and hence were considered technology-dependent and mutually exclusive of network applications. With these issues aside, the designers then focused on the actual data transmission relative to various aspects of an application. This resulted in three additional layers:

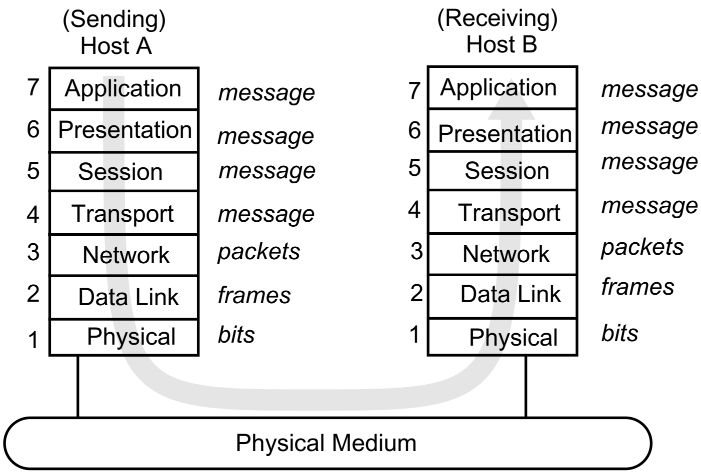
- First, the architecture designers agreed that the sending and receiving nodes of a data transmission should have some assurance that a transmitted message was received correctly and that the sending node was informed of this. Thus, what was needed was a mechanism for reliable, end-to-end transmissions. This led to the establishment of the transport layer, which effectively provides a *quality of service*. The inclusion of a transport layer brings the count to five layers.
- Second, the designers decided there should be a mechanism for the sending and receiving nodes to establish a *dialogue* about how to exchange messages. For example, the end nodes should be able to communicate with each other about how a message was going to be delimited (i.e., where it starts and ends) and how the virtual session between the two end processes was going to be established, monitored, and terminated. The end nodes also needed to exchange *message synchronization* information, which provides checkpoints throughout the transmission so that the two nodes are cognizant of buffer space for message storage as well as when the entire message has been received. This led to the session layer which brings the total number of layers to six.
- Finally, the designers recognized that application messages could be represented in many different ways depending on the circumstances in which a message was created and transmitted. For example, the encoding of a message might be in ASCII or EBCDIC, the message might be in integer format or floating point, an end-of-line character could be a carriage return (CR) or a carriage return followed by a line feed (CR/LF), or the data might be compressed, which reduces the number of bits needed to be transmitted. This led to the development of the presentation layer (which probably should have been more appropriately named representation or encoding layer). In short, the presentation layer is concerned with bit-pattern representation during transfer. The presentation layer was later modified to incorporate a standard language for specifying abstract syntaxes along with a set of encoding rules associated with use of that language. Thus, the presentation layer on the sending node translates the application data format it receives from the application layer into a common language format. The presentation layer on the receiving node then translates this common format to the application format, which is then passed to the application layer.

40. All of this certainly makes sense, but what about network management or security? Shouldn't there have been separate layers for them?

Good question. The issues of management and security did not escape the designers of the OSI model. For example, they felt that management did not warrant a separate layer because the functions of network management (e.g., traffic monitoring) were considered applications themselves. As for security, the designers decided that security features should be provided by the layers and hence incorporated various aspects of security into most of them.

41. How do layers communicate with each other?

Look at Figures 2.12 and 2.13. The data transmission of a message begins on the sending node. An application process first creates a message, attaches an application header to it, and then passes the message to the presentation layer. The presentation layer, if necessary, transforms the message into a different format (e.g., translating the data from the format sent by the application layer into a common format), attaches a header to it, and passes it to the session layer. This process is then repeated from layer to layer. When the message reaches the physical layer, it is then transmitted to the destination node. In some cases a "trailer" is also appended to a message. For example, at layer 2 a trailer is added to facilitate frame synchronization. This is covered in more detail in Chapter 5. This process is then reversed on the receiving node with each layer's header (or trailer) being stripped off one by one as the message ascends the layers. Note from Figure 2.12 that at layer 3 a message is called a packet, at layer 2 it's called a frame, and at layer 1 it's referred to as bits.



IGURE 2.12 The OSI layering process begins at the application layer of the source machine where message is created by an application program. This message moves down through the layers until it eaches layer 1. Underlying layer 1 is the actual physical medium. Data are then transmitted across this edium to the receiving host, where the information works its way up through the layers. As messages ove down the layers, they are encapsulated with headers that are germane to a specific layer. These eaders are removed as the data are passed upward through corresponding layers at the receiving host.

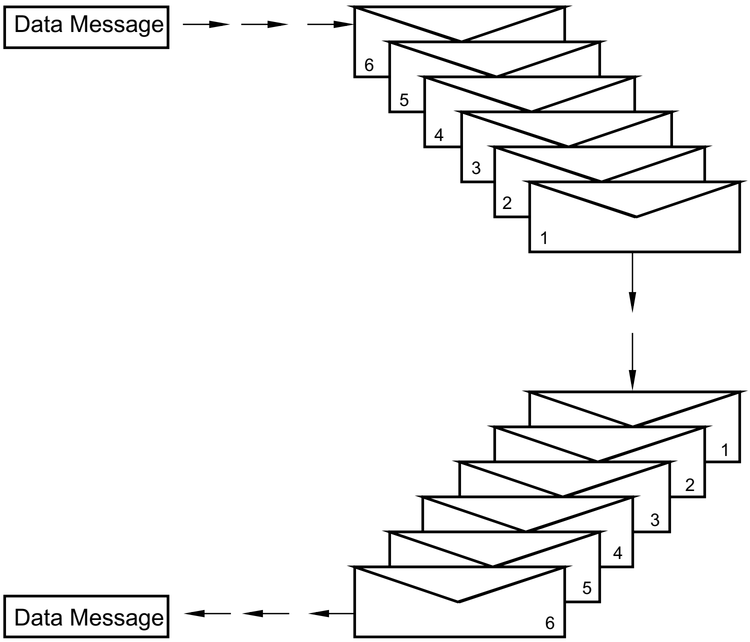


FIGURE 2.13 How layers work. Each layer "envelops" the data with its protocol. Each layer has a corresponding layer on the remote (destination) node, which is called a peer.

Additionally, each layer consists of two parts: a *service definition*, which defines the type of service a layer provides, and a *protocol specification* that details the rules governing the implementation of a particular service. Lower layers provide services to upper layers.

42. What kinds of services do layers provide to each other?

There are two different types of services: *connection-oriented* and *connectionless*. Some layers have an additional type of service called *multiplexing*, but this does not necessarily transcend all layers of the architecture. Services are available at *service access points* (SAPs), with each SAP having a corresponding address. (*Note:* In UNIX, a SAP is called a *socket*, and a SAP address is a *socket number*.)

43. Define connection-oriented service.

Connection-oriented implies that prior to the transfer of data a physical (and virtual) link is established between the sending and receiving nodes. This link remains in effect for the duration of the session. After the session is completed, the link is removed. Characteristics of a connection-oriented service include: wasted bandwidth, because the link must remain established even during idle periods of a transmission; a high potential for a hung network, since there is always a possibility that a link will not be terminated; and (on the bright side) guaranteed sequential arrival of packets at the destination node.

The telephone system is an example of connection-oriented service. You establish a connection (you dial a number); you transfer data over this circuit when the connection is made (you begin talking when the receiver is picked up); communication occurs in the proper sequence (words and sentences are received in the correct order); and you release the connection at the conclusion of the transfer (you hang up the phone, which frees the circuit). Note also the issues of wasted bandwidth and a hung network. If a telephone connection has been made but no one is talking, bandwidth is wasted because the circuit is established but not being used. Anyone trying to contact your house during this period of silence would be greeted by a busy signal—a "hung" connection.

44. What about connectionless service?

Connectionless service differs from connection-oriented service in that no physical link is established between sending and receiving nodes prior to data transmission. Instead, a message is partitioned into packets and routed through the network. Each packet is independent of the other packets that carry parts of the message and hence must carry a destination address. Note that addressing is not necessary for connection-oriented service because a physical, dedicated link is established between sending and receiving nodes before transmitting data. Packets can arrive out of order. Think of the post office as providing connectionless service. If you send someone five separate letters numbered one through five, you must place the recipient's address on each letter. Once mailed, the letters do not necessarily follow exactly the same delivery route, and it is possible for the recipient to receive the letters out of sequence (e.g., letter three is received before letter two).

Connectionless service is also either *reliable* or *unreliable*. Unreliable service requires no acknowledgment of receipt of data from the receiving node to the sending node. This is called a *datagram service*. Reliable service requires an acknowledgment. This is called an *acknowledged datagram service*. Using our post office metaphor, these services compare with mailing a "regular" letter versus mailing a registered letter with a return receipt request.

45. Can you sum all this up with an example?

We could, although the best and most practical example—the Internet—requires additional information that has not yet been presented. For example, the Internet is based on the TCP/IP protocol suite, part of which is discussed in Chapter 3. Also, we have not yet presented detailed discussions of some of the OSI layers, such as the data link and network layers, which are presented in Chapters 5 and 7, respectively. Since you asked, we will give you an example but without a lot of elaboration.

46. Go for it. I promise not to ask for additional information. I just want to try to piece all of this together conceptually.

OK. Let's assume we want to send a message across the Internet. Before doing so, though, we should first provide you with some information about addressing methodology. Three different addresses are needed to send a message from one node to another. The first address is the *hardware address*, which uniquely identifies each node. Hardware addresses are provided by the data link layer. The second address is the *network address*, which identifies the network to which a node is connected. In TCP/IP, this is called an *Internet address* or an *IP* (for Internet Protocol) *address*. Network addresses operate at the network layer. Each network node that is part of the Internet has a unique IP address. (*Note*: IP addresses do more than simply identify the network. See Chapter 3 for additional information about IP.) The third address is called the *port address*, which uniquely identifies a specific user application such as e-mail. All network applications have corresponding identifiers called *port numbers*. To send a message from one node to another, a message is first created at the application layer. It undergoes whatever formatting is required as it descends the layers. When the message reaches the network layer, a network address is assigned to the message. This network address identifies the specific network to which the destination host is connected. Depending on the protocol, this service is either connectionless or connection-oriented. For example, TELNET and SMTP are connection-oriented services. The network layer determines the path the message must follow to reach the destination node. It also encapsulates packets into IP datagrams and passes them to the data link layer. At the data link layer, the destination node's hardware address is added to the packet. This address uniquely identifies the location of the destination node within the destination network. The data link layer, among other tasks, also formats the packet into *frames*, which are like packets but exist at a lower level and checks, the integrity of each frame (see Chapter 5). Frames are then passed to the physical layer, which places them on the medium for transmission.

47. OK. Enough of OSI. How does TCP/IP compare to OSI?

TCP/IP's development preceded the OSI model by several years. Both had similar design goals, however, to fill a need for interoperability among heterogeneous computer systems. Unlike OSI, TCP/IP was never intended to be an international standard. It was developed to satisfy the need to interconnect various United States Department of Defense projects, including computer networks, and to allow for the addition of dissimilar machines to the networks in a systematic, standardized manner.

48. Does TCP/IP also have seven layers like OSI?

No. As a pre-OSI protocol architecture, it was not designed specifically with layers the way the OSI model was designed and it does not fit neatly into the seven layers of the OSI model. However, we can envision TCP/IP's layers as similar to the OSI layers since many of TCP/IP's functions are similar to those of the OSI model.

49. So what are the layers?

There is no universal agreement on the description of TCP/IP as a layered model. It is frequently described as either a four- or five-layered model depending on an author's perspective. For our purposes, we elect to describe TCP/IP as a four-layered architecture, which is shown in Figure 2.14. Note that a five-layered TCP/IP model maintains OSI's physical and data link layers as separate levels instead of combining them into a single layer as shown in Figure 2.14. In this scenario, TCP/IP's first layer is called the physical layer and its second layer is called the network access layer.

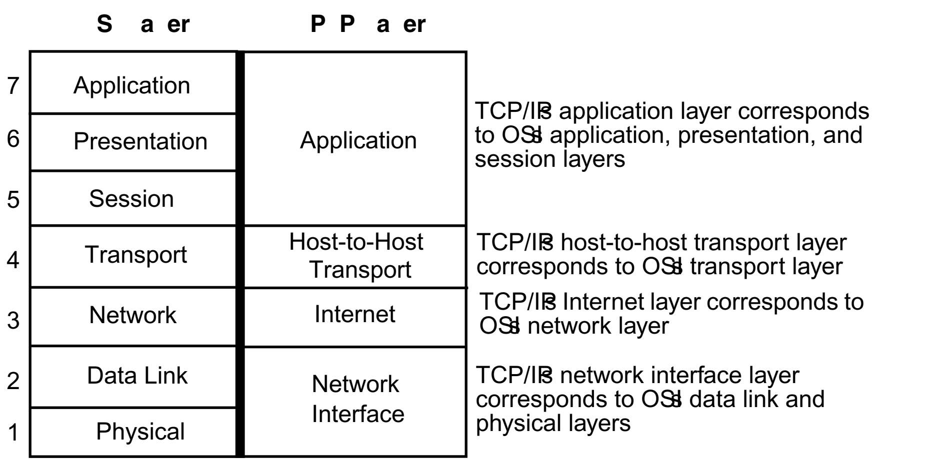


FIGURE 2.14 A comparison of the OSI and TCP/IP layers.

50. Are TCP/IP's layers similar to the corresponding OSI layers in terms of the functions and services they provide?

Yes. The TCP/IP *application layer* serves as the communication interface for users by providing specific application services to the user such as remote terminal login (i.e., virtual terminal), file transfer, and e-mail. Corresponding application protocols include TELNET, FTP, and SMTP. The TCP/IP *host-to-host transport layer* (known simply as the *transport layer*) is responsible for end-to-end data delivery. This layer is defined by two protocols: the *Transmission Control Protocol* (TCP) and *User Datagram Protocol* (UDP). A brief description of each follows. A summary description of the TCP/IP model is also given in Figures 2.15 and 2.16.

- **TCP.** This is the TCP of TCP/IP. It is a connection-oriented protocol that performs several functions including: providing for reliable transmission of data by furnishing end-to-end error detection and correction; guaranteeing that data are transferred across a network accurately and in the proper sequence; retransmitting any data not received by the destination node; and guaranteeing against data duplication between sending and receiving nodes. Application protocols using TCP include Telnet, File Transfer Protocol (FTP), Simple Mail Transport Protocol (SMTP), and Post Office Protocol (POP).
- **UDP.** This is a connectionless protocol providing an unreliable datagram service. It does not furnish any end-to-end error detection or correction, and it does not retransmit any data it did not receive. UDP requires very little overhead since it does not provide any protection against datagram loss or duplication. Application protocols based on UDP include the Trivial File Transfer Protocol (TFTP), Network File System (NFS), the Simple Network Management Protocol (SNMP), the Bootstrap Protocol (BOOTP), and Domain Name Service (DNS).

Application

¥ Similar to OSI application layer.
¥ Serves as communication interface by providing specific application services.
¥ Examples include e-mail, virtual terminal, le transfer, WWW.

ran port

¥ Defined by two protocols:

er atagram Protocol P

¥ Is a connectionless protocol.
¥ Provides unreliable datagram service (no end-to-end error detection or correction).
¥ Does not retransmit any unreceived data.
¥ Requires little overhead.
¥ Application protocols include Trivial File Transfer Protocol (TFTP), NFS, Simple Network Management Protocol (SNMP), Bootstrap Protocol (BOOTP), and Domain Name Service (DNS).

ran mi ion ontrol Protocol P

¥ This is the TCP of TCP/IP.
¥ Is a connection-oriented protocol.
¥ Provides reliable data transmission via end-to-end error detection and correction.
¥ Guarantees data are transferred across a network accurately and in proper order.
¥ Retransmits any data not received by destination node.
¥ Guarantees against data duplication between sending and receiving nodes.
¥ Application protocols include Telnet, FTP, SMTP, and POP.

nternet

¥ Heart and soul is Internet Protocol (IP) the IP of TCP/IP.
¥ Transfers user messages from source host to destination host.
¥ Is a connectionless datagram service.
¥ Route selection is based on some metric.
¥ Uses Internet or IP addresses as a road map to locate a host within the Internet.
¥ Relies on routers or switches (dedicated nodes that connect two or more dissimilar networks).
¥ Integral part is Internet Control Message Protocol (ICMP), which uses an IP datagram to carry messages about state of communications environment.

et or nter ace

¥ Connects a host to the local network hardware.
¥ Makes a connection to the physical medium.
¥ Uses a specific protocol for accessing the medium.
¥ Places data into frames.
¥ Effectively performs all functions of the first two layers of the OSI model.

IGURE 2.15 Summary of the TCP/IP layers and functions.

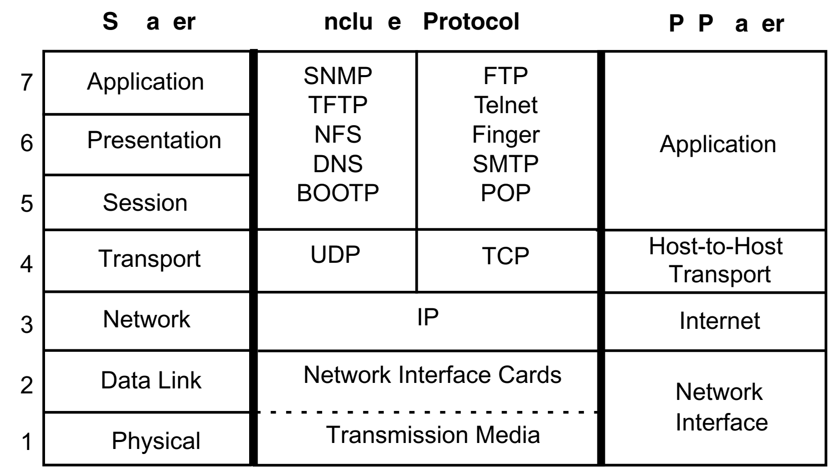


FIGURE 2.16 Another comparison of the OSI and TCP/IP layers. Source: Adapted from Miller, 1992.

The TCP/IP *Internet layer* (also called the *network layer*) transfers user messages from a source host to a destination host. The heart and soul of this layer is the *Internet Protocol*, which is the IP of TCP/IP. IP is a connectionless datagram service responsible for routing packets between nodes. In short, IP receives data bits from the lower layer, assembles the bits into packets (IP datagrams), and selects the "best" route based on some *metric*. (A metric is a description of the "cost" of a route used by routing hardware and software to select the best possible route.)

The TCP/IP *network interface layer* connects a host to the local network hardware. Its functions include making a connection to the physical medium, using a specific protocol for accessing the medium, and segmenting data into frames. It effectively performs all of the functions of the first two layers of the OSI model.

51. What is OSI's role given that the Internet is based on TCP/IP?

At one point during the early 1990s, it was believed by many that the OSI protocols were going to become "the" network standard for everyone. Even the U.S. Government got into the act by establishing GOSIP (Government OSI Profile), which mandated all government organizations purchase OSI-compliant networking products beginning in 1992. In 1995, however, GOSIP was modified to include TCP/IP as an acceptable protocol suite for GOSIP compliance. Today, OSI protocols are in use, but their presence pales in comparison to that of their TCP/IP counterparts. Nonetheless, the OSI model has had a lasting impact on networks, including TCP/IP. The model continues to provide a detailed standard for describing a network. It is from this perspective that the network design community continues to regard the OSI model as a theoretical framework for the development of networks and their architecture.

52. If TCP/IP is "in" and predates OSI, why bother studying OSI?

Recall from Chapter 1 our discussion on protocols. We stated that protocols are necessary to ensure that communications are understood. Analogous to this, the OSI model provides us with a common communication "protocol." It enables us to discuss network principles and concepts with other networking professionals and be understood. For example, if a vendor claims that its latest switch operates at layer 4 or if a co-worker states that network errors are occurring at layer 2, you will understand what each person is talking about because they are referencing the OSI model.

END-OF-CHAPTER COMMENTARY

On this note we conclude our discussion of network topologies, architectures, and the OSI and TCP/IP models. We will expand upon this material in subsequent chapters. For example, in Chapter 3, we give a detailed presentation about the Internet and TCP/IP protocols, including protocols that operate at layers 3 and higher. We also expand the concept of the OSI layers in later chapters. Chapter 4 is dedicated to the physical layer; Chapter 5 presents a discussion on the data link layer; network hardware components that operate at either layer 1 or 2 are presented in Chapter 6; and Chapter 7 addresses concepts relating to the network layer. Other chapters also expand on specific layer 2 and layer 3 protocols.

Chapter 3

The Internet and TCP/IP

In the previous two chapters, we briefly examined the general concept of an internet (lowercase i), contrasted it with the Internet, and introduced Internet-related terms such as intranet and extranet. We also presented a few TCP/IP-based application protocols such as the Simple Mail Transfer Protocol (SMTP) for e-mail, the File Transfer Protocol (FTP) for file transfers, and the Hypertext Transfer Protocol (HTTP) for Web applications. In this chapter, we expand our discussion of the Internet and its related protocols. We begin by defining the Internet from various perspectives. We then provide information about its history and contrast the Internet's early days to its current state. We also present other Internet-related initiatives and provide a brief discussion of its administration and governance structure. We next introduce the TCP/IP protocol suite. We trace its history and provide detailed information about several TCP/IP application-layer, transport-layer, and network-layer protocols. An outline of the major topics we discuss follows:

- Definition of the Internet (Question 1)
- Internet History (Questions 2–10)
- The Current Commodity Internet (Questions 11–12)
- Other Internet Initiatives: vBNS/vBNS+, Internet2, Next Generation Internet (Question 13)
- Internet Administration, Governance, and Standards (Questions 14–16)
- History of TCP/IP (Questions 17–22)
- TCP/IP Application Layer Protocols: SMTP, MIME, POP, IMAP, TELNET, FTP, HTTP (Questions 23–56)
- TCP/IP Transport Layer Protocols: UDP and TCP (Questions 57–71)
- TCP/IP Network Layer Protocol: IP (Questions 72–73)
- IPv4 Addresses and Subnetting (Questions 74–83)
- IP Address Assignments (Questions 84–85)
- IP Name Resolution (Questions 86–88)
- IPv6 (Questions 89–94)
- Internet Services, Resources, and Security (Questions 95–96)

1. Please begin by defining the Internet.

OK. We'll try. Defining the Internet today is a bit more problematic, though, than it was several years ago. Its definition varies from person to person. For example, during its early years, the Internet was defined as a collection of computer networks based on a specific set of network standards, namely, TCP/IP. This was the definition we gave in Chapter 1. Other users, however, whose focus might be on the information they have acquired or the people with whom they have communicated, might define the Internet as a global collection of diverse resources, or as an electronic community of people. In fact, compared to the Internet's early days, some people have commented that the Internet has been transformed from a community of networks to a network of communities. Still others, whose only experience with the Internet is using the World Wide Web, might say the Internet and World Wide Web are synonymous and hence the Internet is the World Wide Web. Consequently, defining the Internet is a function of perspective. Regardless of the definition or perspective, the Internet interconnects individual, autonomous computer networks and enables them to function and appear as a single, global network.

2. How did the Internet get started?

The Internet's roots can be traced back to 1957 when the United States formed the Advanced Research Projects Agency (ARPA) within the Department of Defense (DoD). The formation of ARPA was the United States' response to the former Soviet Union's launch of Sputnik, the first artificial earth satellite. ARPA's mission was to establish the United States as the world's leading country in defense- and military-applicable science and technology. ARPA, which later became known as Defense ARPA (DARPA), established in 1969 an early internetwork called ARPANET, the Advanced Research Projects Agency Network. The builder of ARPANET was a company named Bolt, Baranek, and Newman, which later became known as BBN Communications. Originally, the Internet meant ARPANET, and access to ARPANET was restricted to the military, defense contractors, and university personnel involved in defense research. ARPANET technology was based on packet-switching, and in 1969, with the connection of its first four nodes—Stanford Research Institute (SRI), University of California at Santa Barbara (UCSB), University of California at Los Angeles (UCLA), and University of Utah—ARPANET heralded the era of packet-switching networking.

3. I recall that the university I attended had a BITNET connection. Was this similar to ARPANET?

Not quite. BITNET, which stood for *Because It's Time Network*, was a low-speed and inexpensive academic network consisting of interconnected IBM mainframes. BITNET was one of several cooperative, decentralized computer networks that formed in the late 1970s and early 1980s on college and university campuses to serve the academic community. Using a proprietary IBM-based network protocol, BITNET connectivity was via 9600 bps leased circuits and was based on the store-and-forward principle. Networking services available via BITNET included file transfer, e-mail, and an IBM application called *remote job entry* (RJE). In an RJE environment, small processors located at remote sites were used to transfer "jobs" to and from a main computer that served as the "master" to these smaller processors. This scheme is based on a paradigm known as *master/slave*. BITNET eventually merged with another early academic network called the *Computer Science Network* (CSNET) to form the Corporation for Research and Educational Networking (CREN).

4. Was CSNET like the ARPANET?

CSNET was similar to the ARPANET. It was a large internetwork developed to provide connectivity to the nation's computer science community. The development of CSNET was grounded in the restricted use of the ARPANET. Owned by the Department of Defense, ARPANET's use was prohibited by anyone outside the defense community. In an effort to increase collaboration among the nation's computer scientists, the National Science Foundation (NSF) funded CSNET. Recognizing that the most popular ARPANET service was e-mail, the developers of CSNET initially thought an e-mail-only based network could be developed to connect academic and research institutions that did not have access to ARPANET. CSNET eventually evolved into a *metanetwork*—it consisted of several different physical networks logically designed to serve one community. Connectivity to CSNET was via a centralized machine called CSNET-RELAY, and connectivity to CSNET-RELAY was via other networks. These included a public packet switching network called X.25NET, a dialup network called Phonenet, and the ARPANET. CSNET provided its users with Internet-type services such as e-mail, member registry, and domain name service (DNS). Other services such as file transfer and remote logins were also available on some parts of CSNET. CSNET, BITNET, and CREN have since disbanded now that Internet access is easily attainable.

5. Friends of mine talk about a network called UUCP. What was this network like?

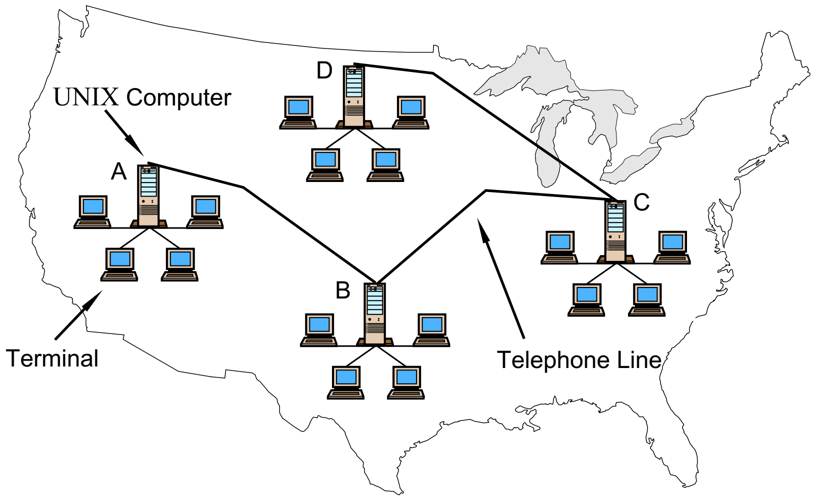
They probably also had a FidoNet connection. The UUCP network is a global network of interconnected UNIX machines. Standard telephone lines serve as the medium for connectivity, and the network is based on the store-and-forward principle (Figure 3.1). Using a suite of programs known as UUCP (UNIX-to-UNIX Copy), users can exchange e-mail and network news (also called Usenet news). Given its minimal requirements (UNIX machine, modem, and telephone connection) and relatively inexpensive nature, the UUCP network grew quickly throughout the world and is still in existence today.

6. OK. So what happened next? How did ARPANET become the Internet?

Around 1983, the academic and research science community convinced Congress that the United States had to meet the Japanese supercomputing challenge.

7. Time out. What was the Japanese supercomputing challenge?

The Japanese government committed itself to a national goal of developing a computer capable of displaying common sense, possessing a general knowledge about how the world works, having insight into human nature, having a vocabulary of 10,000 words, speaking



IGURE 3.1 An example of a nationwide UUCP network. A message from host A to host D ould first be transferred to host B, then to host C, and finally to host D.

and understanding English and Japanese, all by the end of the 1980s. In the United States, Japan's national goal was perceived as analogous to the United States' national goal of the 1960s of putting a man on the moon and returning him safely to earth by the end of that decade. Suffice it to say the Japanese supercomputing challenge was taken very seriously in the United States.

8. I see. OK. Continue with what you were saying before I interrupted you.

In responding to the Japanese challenge, Congress authorized the National Science Foundation (NSF) to fund the construction and operation of U.S. supercomputer centers. By the end of 1985, six such centers were established throughout the country. These centers also were connected to the ARPANET, which had by then (1984) split into two separate networks—ARPANET, for nonmilitary and research purposes, and MILNET, an unclassified military network. The supercomputer center network and the ARPANET were interconnected at Carnegie Mellon University (CMU) in Pittsburgh, Pennsylvania. Thus, network traffic originating at any of the supercomputer centers and destined for the ARPANET (or vice versa) was first sent to CMU and then transferred to the local ARPANET node (Figure 3.2).

With a supercomputer center network in place, the next issue to address was to provide researchers with direct and convenient electronic access to these centers from researchers' home institutions. To meet this challenge, NSF began funding in 1986 the development of a national "backbone" network. Eventually, a three-level or tiered network evolved consisting of a backbone network, several regional or mid-level networks, and local area networks of colleges and universities (Figure 3.3). The regional networks were organized geographically either by state (e.g., NYSERnet—New York State Educational Research

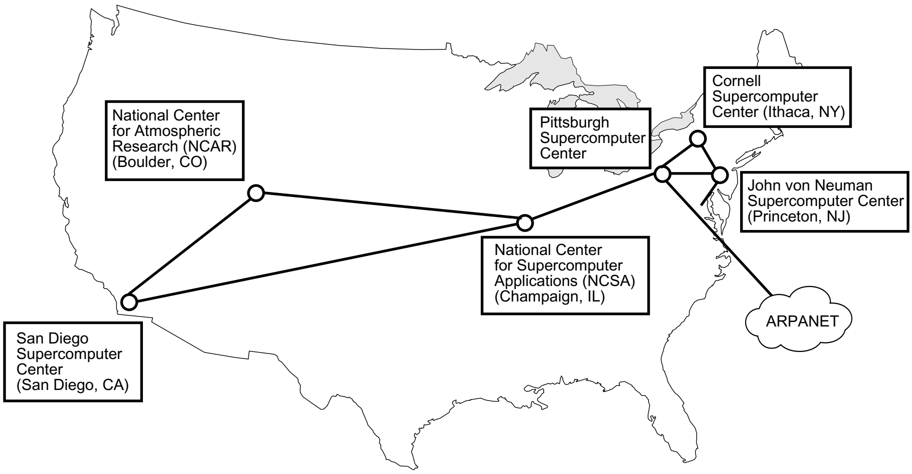


FIGURE 3.2 The National Science Foundation's supercomputer network. This network interconnected the six NSF-funded supercomputer centers. The network also was interconnected to the ARPANET through a node at Carnegie-Mellon University in Pittsburgh.

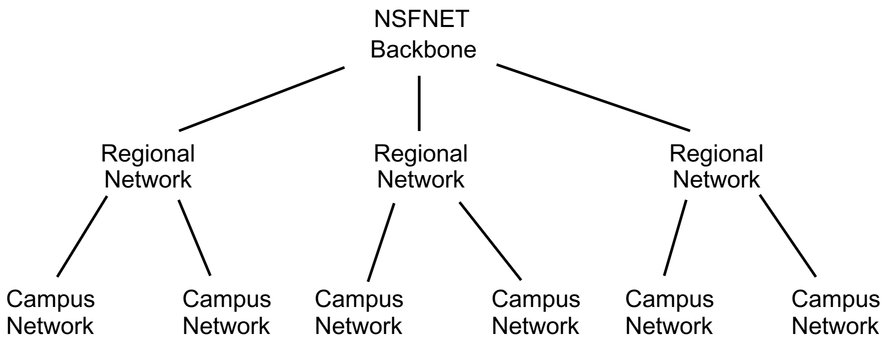


FIGURE 3.3 The NSFNET hierarchy.

Network—serviced New York State) or region (e.g., SURAnet—Southeastern Universities Research Association Network—serviced the southeastern part of the country). Thus, the backbone provided connectivity to the regional networks, which in turn provided connectivity to campus LANs (Figure 3.4). The first two levels of this network (the national backbone and the regional networks) became known as the *National Science Foundation Network* (NSFNET) and was a model for interconnecting independent and autonomous networks. Initially, the NSFNET backbone was based on 56-kbps leased circuits. You can think of these circuits as being equivalent to the data transmission rates capable of today's

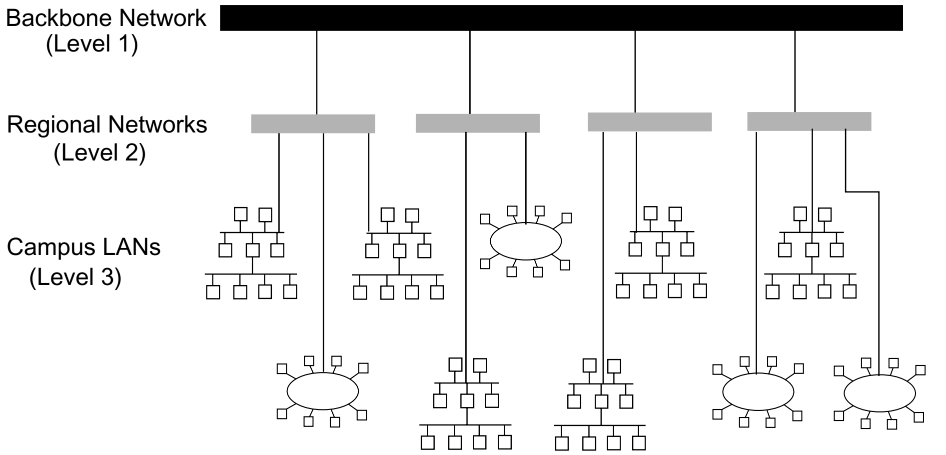


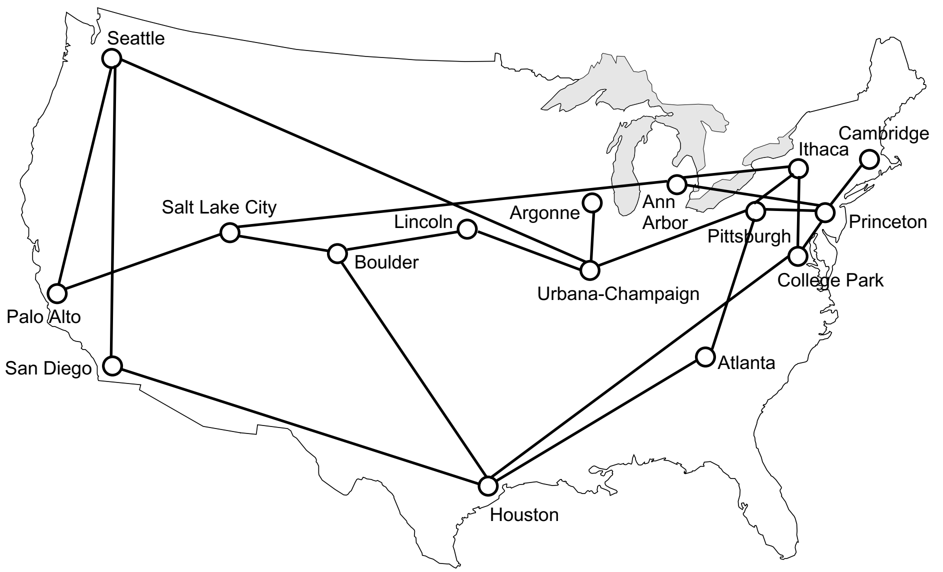
FIGURE 3.4 Example of the three levels of networks that evolved from the National Science Foundation's networking initiative. The first two levels made up NSFNET.

56K modems. They weren't very fast. Back then, however, 56K was considered by most to be very fast compared to 300 bits per second or 9600 bits per second. Furthermore, there were not that many users and the applications were mostly text-based and hence not bandwidth intensive. From 1989 to 1992, the backbone was reconfigured twice to handle increasing traffic loads. This reconfiguration included adding new circuits, deleting others, and increasing the backbone's bandwidth—first to T1 (1.5 Mbps) and then to T3 (45 Mbps). The T3 backbone is shown in Figure 3.5.

During the 1980s, other government organizations such as the Department of Energy (DOE) and the National Aeronautics and Space Administration (NASA) also began developing their own private networks. These networks interconnected the NSFNET. Eventually, all of these government-sponsored networks, including NSFNET, ARPANET, MILNET, and SPAN (the Space Physics Analysis Network), became known as the Internet. Although the Internet was comprised of many different networks, the NSFNET was perceived as *the* Internet by many people. This perception was grounded in the NSFNET's national presence and by its open door policy to any research or educational organization. In fact, through its networking infrastructure program, the NSF provided funding to any college or university that wanted to connect its LAN to the NSFNET. A modified version of this program was later extended to K–12 schools. Funds were used to purchase hardware and pay for high-speed line charges related to a school's NSFNET connection.

9. How did the Internet evolve from a restricted research and academic network to one that is available to the general population?

As NSFNET became more popular, the business community quickly took notice and realized there was money to be made from it. Connectivity to the NSFNET backbone,



IGURE 3.5 The T3 NSFNET backbone, circa 1992–1993.

however, was governed by NSF's Acceptable Use Policy, which—although usually ignored and often broadly interpreted—restricted the use of the NSFNET backbone to educational or research activities. This meant that the NSFNET could not be used for commercial purposes. In response to this policy the *Commercial Internet Exchange* (CIX, pronounced "kicks") was born in the early 1990s to meet the emerging connectivity needs of the commercial market. CIX was a subscription organization that consisted of a consortium of commercial and nonprofit regional network providers that started offering Internet service independent of the NSFNET backbone and without NSF's restriction on traffic type. It established a national backbone comprised of different telephone carriers (e.g., Sprint and MCI, which later became known as WorldCom) and interconnected the NSFNET backbone at selected points. Thus, commercial Internet traffic might start at a local organization, travel through various regional providers, get routed to a CIX node, and then get transferred to another CIX provider without ever crossing the NSFNET backbone.

During the early to mid-1990s, a commercial Internet began to take shape, consisting of "20 percent bottom-line" people instead of research or educational visionaries. During this period private Internet service provider (ISP) businesses were started by entrepreneurs who recognized the growing demand for Internet access by both the general public and the business community. ISPs ranged from small "mom and pop" operations that provided connectivity to a specific locale, to regional or state providers, to national and international providers. The source of connectivity for these ISPs was the backbones developed by the major long distance telephone carriers—Sprint, AT&T, and MCI (i.e., WorldCom)—all of whom operated their own private national networks independent of NSF access restrictions.

10. What happened to ARPANET and NSFNET?

In 1990, ARPANET was decommissioned, and the level of commercialization of the Internet led the National Science Foundation to remove the NSFNET backbone from active service on April 30, 1995. This latter action, however, did not occur without considerable forethought. For example, in May 1993, two years prior to NSFNET's retirement, NSF solicited proposals to design a new infrastructure capable of serving the needs of not only government, research, and educational organizations, but also those of the commercial user and general public. Furthermore, NSF maintained its commitment to the educational and research communities that comprised NSFNET by subsidizing the regional network providers' connections to a commercial network service provider. This support ended in 1999. Most of the regional providers selected MCI (WorldCom) as their primary carrier; the remaining few chose Sprint or ANS, which was purchased by America Online in 1995. Today, network traffic such as e-mail, Web browsing, and file transfers is carried by commercial providers; the former NSFNET regional networks now receive connectivity through commercial network service providers; and NSFNET's three-tiered hierarchical national backbone has been supplanted with several independent backbones that interconnect at specially designated exchange points (more on this later) where ISPs meet and distribute traffic (Figure 3.6).

Summarizing its history, over the past 30 years the Internet has evolved from a U.S. Department of Defense network research project (ARPANET) to an NSF-subsidized educational and research data communications medium for university, government, and research personnel (NSFNET) to a commercial, global network linking tens of millions of consumers, businesses, schools, and other organizations. Today, nearly every country in the world has some form of Internet connection. Initially, the ARPANET was the Internet. NSF then entered the picture and the Internet was perceived as a collection of various networks anchored by NSFNET. Today, both ARPANET and NSFNET have given way to a commercially flavored Internet that is transforming the Internet from its roots as a collection of networks to a global network of communities. If interested, you can download from ftp://ftp.cs.wisc.edu/connectivity_table/ various early network connectivity maps.

11. What is the current design of the Internet?

As noted earlier and illustrated in Figure 3.6, the Internet's infrastructure comprises numerous backbones operated by national, regional, and local Internet service providers. The major national backbones are high-speed networks owned by telecommunications organizations such as Sprint, AT&T, and WorldCom as well as by major cable companies such as TimeWarner, which recently merged with America Online (AOL), and satellite providers. Regional backbones are generally owned by regional Bell operating companies (RBOCs) such as BellSouth and Verizon (formerly Bell Atlantic) or by newly formed competitive local exchange carriers (CLECs). Local backbones are typically operated by small, independent Internet service providers (ISPs) or by CLECs. The major backbone providers (sometimes called Tier 1 ISPs) provide access to regional and local ISPs; many regional providers also provide access to local ISPs.

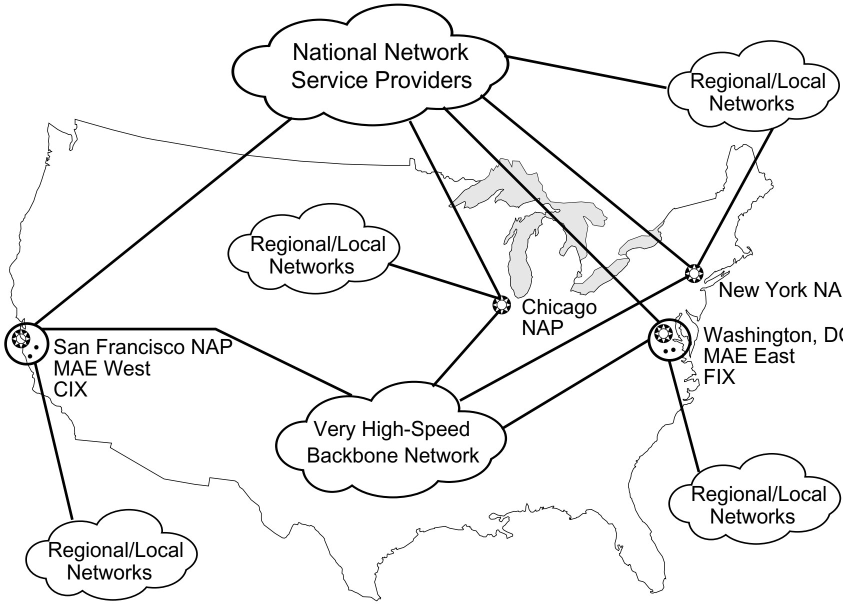


FIGURE 3.6 The new national network infrastructure consists of a "private" very high-speed backbone network service funded by NSF, several independent commercial backbones operated by network service providers such as Sprint, MCI WorldCom, and AT&T, and various interconnection centers where traffic from network providers is exchanged. Network service providers also interconnect the former NSFNET regional networks and thousands of smaller networks operated by independent Internet service providers (ISPs), which provide businesses, schools, and individual users access to the Internet.

12. Are all of these backbones independent of one another? If so, then how do my e-mail messages get to their correct destinations if I use a different provider (and hence, a different backbone) than the recipients of my messages?

The national backbones are indeed pretty much independent of one another. Given the multitude of independent backbones, the Internet relies on network access points (NAPs) and a routing arbiter for interregional connectivity. *Network access points* are special high-speed Internet exchange points (IXPs) or "switching stations" where different network backbone providers meet and exchange traffic on each other's backbone. This is where the individual networks comprising the Internet join together. NAPs conceptually represent the "Inter" part of the Internet. For example, if you have an Internet account from an ISP whose network connects to AT&T's backbone and your next-door neighbor receives Internet service from an ISP whose network connects to BellSouth's backbone, then an e-mail message sent from you to your neighbor (or vice versa) is exchanged at a NAP. To facilitate this traffic exchange, a *routing arbiter* maintains special servers that contain routing information databases of network routes (i.e., electronic road maps). Thus, the routing arbiter enables traffic entering a NAP from one backbone to be routed to the correct destination backbone. The concept of routes and routing is discussed later in Chapter 7.

Initially, the National Science Foundation funded four NAPs in the United States, one each in New York, Chicago, Washington, DC, and San Francisco. These NAPs are operated respectively by Sprint, Ameritech Advanced Data Services (AADS), Metropolitan Fiber Systems (MFS), and Pacific Bell. Augmenting the four official NAPS are additional de facto NAPs, IXPs, and private and public exchange points. These include seven metropolitan area exchange (MAE) points, which are located in Vienna, VA (MAE-East), San Jose, CA (MAE-West), Dallas (MAE-Dallas), Houston (MAE-Houston), Los Angeles (MAE-LA), Chicago (MAE-Chicago), and New York (MAE-NY). There is also the commercial Internet exchange (CIX) in Santa Clara, CA, as well as two federal IXPs, FIX-East in College Park, MD, and FIX-West in Ames Research Center in Mountain View, CA. Finally, another NAP, the Florida MIX (Multimedia Internet Exchange), was established in south Florida and operated by BellSouth in 2001.

Most national service providers maintain connections to the four official NAPs and to most of the MAEs. Unlike the NSF-funded NAPs, where traffic exchanges flow freely among all connected backbones, however, other IXPs usually require *peering agreements* among the ISPs that connect to their sites. Thus, two backbones that connect to MAE-LA, for example, do not exchange traffic with each other unless they have a peering arrangement. Establishing a peering agreement generally involves negotiation and technical cooperation to enable the two ISP networks to exchange traffic. Finally, outside the United States, NAPs are located in several parts of the world, including London, Amsterdam, Paris, Tokyo, and Hong Kong.

Additional information about Internet exchange points is available at the Web sites http:// www.mfst.com/MAE/ doc/MAE.doc and http://www.ispworld.com. Another excellent site is Mapnet, which provides an interactive tool for visualizing the Internet's infrastructure of multiple backbones simultaneously. It is located at http://www.caida.org/tools/visualization/mapnet/backbones. Once at this site, you can view the Internet's backbone infrastructure from the world, USA, Asian, and European perspectives. You can also select the type of backbone to view—commercial or research.

13. With the NSF out of the picture, how is network-related research conducted? I can't imagine those academic researchers were too pleased when the Internet went commercial. I bet this was tantamount to Dylan going electric.

Well, maybe for some, but we don't think a commercialized Internet was viewed in the same light as Dylan going electric. The latter was totally unexpected and represented a revolution in the music industry. The former was more of an evolution and hence was expected. Getting back to your question, though, the NSF is not really out of the picture. In addition to the current "commodity" Internet, several additional Internet backbone initiatives have been developed. These include the very high performance Backbone Network Service + (vBNS+), Internet2, the Next Generation Internet (NGI), and the National Information Infrastructure (NII). A brief review of each initiative along with a corresponding Web site address where more information is available follows.

vBNS+ The *very high performance Backbone Network Service +* originated as an NSF-funded research and educational network. The contract to develop and operate this network was awarded by NSF to MCI WorldCom in 1995. When initially developed, the network was called the very high performance Backbone Network Service (without the +) and provided researchers with an appropriate testbed-research environment for deploying and evaluating new high-speed internetworking technologies. The original vBNS also maintained a strict usage policy: Access to the vBNS was restricted to only those organizations that received NSF awards under NSF's High Performance Computing and Communications (HPCC) program. This enabled researchers to develop and test new network technologies on a national scale without the frequent congestion and other inherent problems of the public Internet.

On April 1, 2000, vBNS officially became vBNS+. The transition from vBNS to vBNS+ was more than a name change, though; vBNS+ represents a new network. Unlike the original vBNS 622 Mbps (OC-12) ATM (see Chapter 14) backbone, the vBNS+ backbone technology is packet over SONET (see Chapter 7). Furthermore, vBNS+ comprises multiple links running at 2.4 Gbps (OC-48) speed. For an average size packet of 300 bytes, the throughput rate of these links (i.e., the available bandwidth for the packet's payload) is 2072 Mbps. vBNS+ also provides access to the public Internet—something its predecessor did not—and does not have a restricted access policy. Today, vBNS+ service is open to the entire U.S. higher education community, which enables university faculty and researchers across the country to have access to many of the high-performance networking applications that were only available to NSF HPCC program awardees through vBNS. The vBNS+ network also extends to Europe and Asia with point of presences (POPs) located in London, Amsterdam, Frankfurt, Paris, Tokyo, and Hong Kong. National POPs are located in New York, Washington, DC, Boston, Atlanta, Houston, Los Angeles, San Francisco, Seattle, Chicago, Cleveland, Denver, and Memphis. Copies of the current and future vBNS+ backbones are available at http://www.vbnsplus.net.

vBNS+ offers digital video services over IP, intranet and extranet services, virtual private network (VPN) services (discussed in Chapter 7), IPv6 native service (discussed later in the chapter), and the obvious high-bandwidth throughput with negligible loss (packet loss is less than 0.001%). Ongoing research and testing activities supported by vBNS+ include IPv6-enabled wireless networking, IP telephony, high-bandwidth multicast, and wire speed security filtering. Finally, vBNS+ is a support service for Internet2 and Next Generation Internet (NGI) initiatives (see below).

Internet2 *Internet2*, or *I2,* is a not-for-profit consortium led by the University Corporation for Advanced Internet Development (UCAID). Over 180 U.S. universities are involved with the project and are working in partnership with government organizations and private sector firms. I2's mission is to (a) develop and deploy advanced network technologies and applications that support the research endeavors of I2-member colleges and universities and (b) to transfer this technology to the Internet community. Some of I2's research applications include remote scientific modeling and instrument control, high performance distributed computation, and large-scale database navigation. I2's educationally-related projects include high-resolution networked virtual reality that can be used for multisite collaboration, training, and distance learning. University members are also experimenting with real-time medical image diagnosis as well as comprehensive downloadable audio and video libraries. A sampling of some specific projects follows.

- Washington State's K–20 Education Network provides electronic access to a professional development video library for the state's educators.

- Missouri's MOREnet enables fourth-grade students to access online historical information available in rich multimedia formats from Presidential libraries and public television.
- A partnership comprising Northwestern University's International Center for Advanced Internet Research (iCAIR), C-Span, and IBM will support high quality C-SPAN programming over Internet2 networks enabling C-SPAN viewers to interact with video content.
- Gallaudet University and Georgetown University are collaborating on a project called Asynchronous learning via American Sign Language that will support full-motion video of sign language and finger spelling with synchronized audio, text, and graphics.

Internet2 members use one of two national backbones to conduct their research and test their applications. One backbone is the vBNS/vBNS+ network, which we discussed previously. The other is the Abilene Network, which is a high-performance network developed and operated under a partnership of Internet2, Cisco Systems, Nortel Networks, Qwest Communications, and Indiana University (see http://www.internet2.edu/abilene). The backbones, along with key member sites, are interconnected to one another via gigaPOPs, which are high-speed points of presence. Although implicit in its name, Internet2 is not a separate network and will not replace the current Internet. It is simply a collaborative project or activity involving academia, industry, and government. Additional information can be found at http://www.internet2.edu.

NGI The *Next Generation Internet* initiative, announced by former President Bill Clinton in 1996, is a research and development (R&D) program for designing and testing advanced network technologies and applications. This initiative's mission is also to forge collaborative partnerships between the private and public sectors. As stated earlier, the vBNS/vBNS+ network serves as the medium for NGI. Unlike Internet2, which is university led, NGI is federally led. Nevertheless, the two initiatives are engaged in parallel and complementary work. Additional information about NGI can be found at http://www.ngi.gov.

NII The *National Information Infrastructure* is a federal policy initiative to facilitate and accelerate the development and utilization of the nation's information infrastructure. The perception of the NII is one of a "seamless web" of telecommunications networks consisting of computers, specialized databases, radios, telephones, televisions, and satellites. The NII is expected to provide consumers with convenient and instantaneous access to nearly any kind of information ranging from research results to medical and educational material to entertainment.

14. Yikes! This was more than I wanted to know, but thanks for the info. Let's get back to the Internet itself. Who administers it?

The governance and administration of the Internet is overseen by various organizations composed mostly of volunteers from the global Internet community. These organizations consist of the Internet Society (ISOC), the Internet Architecture Board, the Internet Engineering Task Force, and the Internet Research Task Force. The administrative organizational structure of the Internet is shown in Figure 3.7.

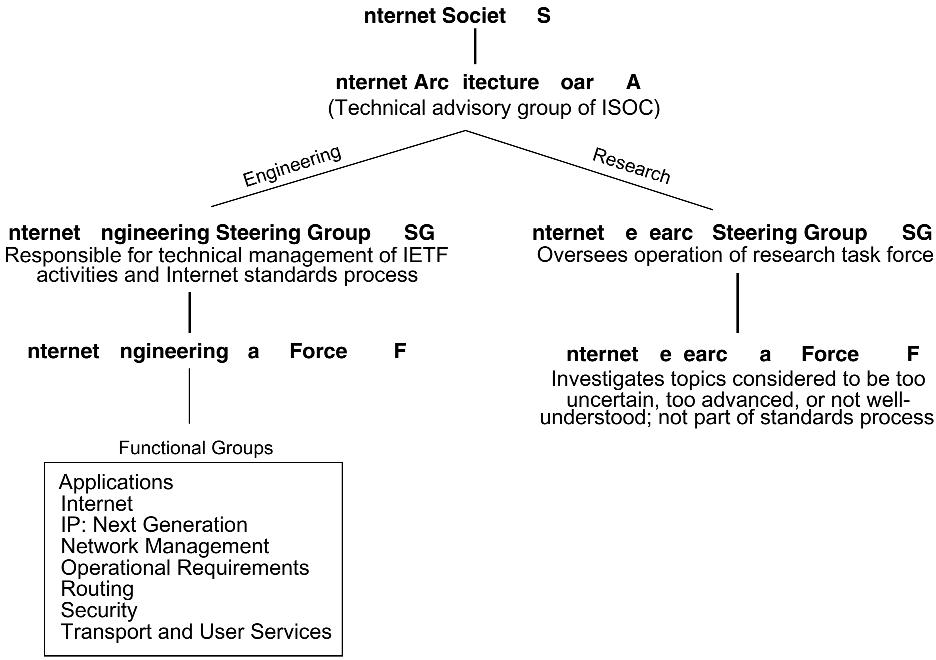


FIGURE 3.7 The administrative organizational structure of the Internet.

The *Internet Architecture Board* (IAB), formerly known as the Internet Activities Board, is responsible for the overall planning and designing of the Internet. Some of its responsibilities include setting Internet standards, managing the publication of RFC documents (discussed later), and resolving technical issues. Assigned to the IAB are the *Internet Engineering Task Force* (IETF) and the *Internet Research Task Force* (IRTF). IETF is primarily concerned with addressing short- or medium-term Internet engineering issues. For example, IETF was responsible for examining the design and implementation of a new Internet Protocol, IPv6 (discussed later), the Open Shortest Path First (OSPF) routing scheme (see Chapter 7), and other initiatives such as enabling TCP/IP to support multimedia more efficiently. IRTF works on long-term research projects. An example of its work is the e-mail privacy issue. Both task forces also have steering committees that prioritize and coordinate their respective activities. IETF's steering committee is called the Internet Engineering Steering Group (IESG), and IRTF's steering committee is called the Internet Research Steering Group (IRSG). The ultimate authority of the Internet is the *Internet Society* (ISOC), an international organization comprised of volunteers who promote the Internet as a medium for global communication and collaboration. ISOC publishes a newsletter called *Internet Society News*, which provides its readers with information surrounding the administration and evolution of the Internet. More information about these organizations can be obtained from their Web sites: ISOC (http://www.isoc.org); IAB (http://www.iab.org/iab); IETF (http://www.ietf.org); and IRTF (www.irtf.org),

15. How are Internet standards developed?

Internet standards are initially developed by IETF, reviewed by IESG, and ultimately approved by the IAB. Throughout the standards development process, Request for Comments (RFCs) are prepared by the RFC editor. RFCs, which address various aspects of a particular protocol under consideration, generally contain technical information about a protocol, but they can provide nontechnical information as well. Although the International Organization for Standardization (ISO; see Chapter 2) promotes approved Internet standards internationally and may also take information from RFCs in developing their own ISO approved standards, RFCs never become international standards.

16. What are RFCs?

Nearly all of the information about the Internet's history and its protocols is contained in RFCs. They are the working notes of the Internet research and development community. They provide network researchers and designers with a medium for documenting and sharing new ideas, network protocol concepts, and other technically related information. They contain meeting notes from Internet organizations, describe various Internet protocols and experiments, and detail standards specifications. RFCs presently comprise two subseries: for your information (FYI) documents and standards (STDs) documents. FYIs provide information about Internet-related topics but in a less technical manner; STDs identify RFCs that specify Internet standards. Consequently, all Internet standards are published as RFCs, but not all RFCs are Internet standards. In addition to RFCs are Internet drafts, which are working documents of the IETF. Internet drafts enable the Internet community to read and comment on proposed Internet-related documents before they are officially published as an RFC. Internet drafts are considered temporary documents and have a shelf life of only 6 months. To facilitate the dissemination process, and to maintain a spirit of openness, RFCs and FYIs are available on-line at http://www.rfceditor.org. There are presently over 3100 RFCs ranging from the serious to the not so serious. Two less serious RFCs written during the Christmas season are RFC 968, "'Twas the Night Before Start-up," written by Vint Cerf in1985 (Cerf is widely known as one of the Fathers of the Internet) and RFC 1882, "The 12 Days of Technology Before Christmas," written in 1995 by Bill Hancock, one of the authors of this book. For more information about the Internet standards process, see RFC 2026, "The Internet Standards Process—Revision 3."

17. You said earlier that the Internet is based on the TCP/IP protocol suite. What is TCP/IP's history? I heard that it was developed to withstand a nuclear attack. Is this true?

Not quite. TCP/IP, which stands for *Transmission Control Protocol/Internet Protocol*, is a formal protocol suite that is primarily based on two subprotocols: TCP, an OSI layer-4 protocol, and IP, an OSI layer-3 protocol. TCP/IP's history is tied to the development of the ARPANET, which initially was based on a protocol called the *Network Control Protocol* (NCP). ARPANET's original design was grounded in two fundamental principles: The physical network was assumed not to be completely reliable, and network protocols could not be dependent on any proprietary hardware or software. The presumption of a completely unreliable network might seem a little odd at first. However, as a Department of Defense project in the 1950s, the ARPANET was definitely a part of the Cold War and hence there was an accepted reality that the physical network could be disrupted by a catastrophic event. This mind set included the notion of a "second-strike" capability. That is, if we were "hit" first, then we need to be able to withstand the attack by having the capability to launch our own strike. This spurred TCP/IP's development. The idea was to enable different packet networks to be interconnected so the host computers did not have to know anything about the intermediate networks linking them together.

18. Who actually developed the TCP/IP protocols?

Helping in the development of TCP/IP were Vint Cerf and Robert Kahn. In the early 1970s, both Cerf and Kahn, as part of an ARPA internetworking research program, developed the idea of gateways and wrote the first specification for the basic TCP/IP protocols now used in the Internet.

19. How did TCP/IP become the network protocol of the Internet?

The nonproprietary principle, coupled with the success of the early ARPANET, led to TCP/IP becoming available on a wide variety of hardware and software platforms. By 1982, ARPA established TCP/IP as the protocol suite for ARPANET, and the Department of Defense declared them standards for military use. This led to one of the first definitions of an internet as a connected set of networks, specifically those using TCP/IP, and "Internet" as connected TCP/IP internets. The idea behind the Internet was the seamless linking of many different kinds of packet-switched networks. This was facilitated by the robustness of TCP/IP, which enabled data communications across analog lines, packet radios, satellite links, Ethernet networks, and others.

As the ARPANET grew in the 1980s, so did computer networking. The popularity of computer networks was helped in part by the proliferation of individual computers and workstations—users wanted to connect their systems together. Recognizing the potential of a large marketplace, networking's popularity quickly led to the development of several proprietary networking protocols. This in turn also led to problems of interoperability. Within a closed, homogeneous networking environment (e.g., DECnet or SNA), interoperability was not an issue because all networked devices spoke the same language. This was not the case, however, in a heterogeneous or mixed-vendor environment.

Around this same time, the University of California at Berkeley's Computer Science Department was enhancing the original version of the UNIX operating system. Called BSD UNIX, one of its new features was the incorporation of the TCP/IP protocol suite. This software was freely available and soon became quite popular at universities throughout the country. Given that TCP/IP was bundled with UNIX, and that TCP/IP was being used successfully in a real-time network (ARPANET), the National Science Foundation mandated that all NSF-funded supercomputer centers and computer networks that comprised NSFNET use TCP/IP as their network communications protocol. NSF's mandate essentially established TCP/IP as a de facto standard.

20. Why is TCP/IP so dominant? Wasn't OSI supposed to be "the" network protocol?

Yes. In the early to mid-1990s, the networking literature was replete with articles that extolled various advantages and virtues of OSI compared to TCP/IP. Many network administrators, in fact, developed strategies to eventually migrate their TCP/IP-based networks to OSI. Some even professed that everyone's network should only support a single protocol: OSI. In spite of all the hoopla and hyperbole, OSI never emerged as "the" network protocol, particularly in the United States. Several reasons (all speculative, of course) why this happened include the following:

- *Standards Development:* TCP/IP and OSI differ in the way their standards are developed and tested. As a formal international standards organization, ISO possesses considerable inertia. The process of developing standards is painstaking. From the initial development of OSI, ISO has tried to do everything at once, and from the top down. In stark contrast to this approach, TCP/IP supports an open process for standards-making participation by its end users. The development of new or modification of existing TCP/IP protocols is also done on an as-needed basis. Furthermore, research, development, and testing of new or modified protocols can be performed on a production network, the procedure is in the open via RFC documents, and distribution of TCP/IP standards is free. (OSI protocols are copyrighted and carry a nominal purchasing fee.)
- *Snob Factor:* The underlying policy of TCP/IP was directed at connecting hosts primarily within the United States—specifically academic, research, government, and military organizations. OSI, on the other hand, was the product of an international standards body (ISO). Consequently, many European users perceived TCP/IP as a parochial *de facto* standard specific to the United States and wanted to embrace OSI. In the United States, though, users did not want to accept anything "different" and stayed true to TCP/IP.
- *Versatility and Robustness:* Compared to OSI, TCP/IP is simple and dependable, it has a proven track history (more than 25 years), it is nonproprietary, its developers have a pragmatic approach to its enhancement, and it meets the networking needs of a diverse population including researchers, educators, and business personnel. Some people—including Vint Cerf, who professes "IP over everything"—think of TCP/IP as the universal language of networking.
- *The Internet:* TCP/IP is inextricably linked to the Internet.

21. How does TCP/IP compare to the OSI layers?

See Figures 2.14, 2.15, and 2.16 for a comparison between the two models. It might also be beneficial for you to review the text that accompanies these figures.

22. Is it possible for you to explain some of the protocols used at these layers?

Yes. Let's work down the stack and discuss the application layer first (OSI layers 5–7), followed by the host-to-host transport layer (OSI layer 4), and then the Internet layer (OSI layer 3). TCP/IP's network interface layer (OSI layers 1–2) are the subject of later chapters so we won't present them at this time.

23. Sounds good to me. What can you tell me about the application layer protocols?

There are many different TCP/IP-based application-layer protocols. Some of the most frequently used ones, and the ones you are probably most familiar with, include the Simple Mail Transfer Protocol (SMTP), the Multipurpose Internet Mail Extensions (MIME), and the Post Office Protocol (POP) for e-mail, the TELNET protocol for virtual terminal connections, the File Transfer Protocol (FTP) for file transfers, and the Hypertext Transfer Protocol (HTTP) for Web applications.

Before we begin discussing them, you might want to review the material we presented in Chapter 1 that distinguished between an application and application protocol. Recall that there is a difference between a network application like e-mail and the application-layer protocol that defines it (e.g., SMTP).

24. Okay. Let's start with electronic mail.

From a general perspective, electronic mail (e-mail) refers to the concept of creating, sending, receiving, and storing messages or documents electronically. Nearly every computer system has a program that serves as an interface for e-mail service. This interface provides users with a utility to, among others, compose, read, save, forward, and print mail messages. In addition to providing a user interface, a local system's e-mail service also supports background processes that govern how incoming and outgoing e-mail messages are stored, how users are presented with incoming e-mail, and how often delivery of outgoing messages is attempted. A generic diagram of how local e-mail service operates is shown in Figure 3.8.

25. How does SMTP work relative to this general description?

Of all the activities depicted in Figure 3.8, the only one that is not performed by the local e-mail service is message delivery. The method by which e-mail messages are transferred from one host to another across a network is defined by a mail application protocol, and in the Internet, the standard mail protocol is the *Simple Mail Transfer Protocol* (SMTP), which is defined in RFC 821. SMTP operates in the following manner:

> When a user sends an e-mail message, the sending host, through its mail service, places a copy of the message in a special location known as the mail queue. The sending host then attempts to establish a special mail connection to the receiving host as indicated by the mail address. A connection is established when the receiving host sends an acknowledgment to the sending host that it is ready to accept mail. Assuming a mail connection is successful, the sending host transfers a copy of the mail message to the receiving host's mail queue. When both sending and receiving hosts confirm the transfer was successful, the message is removed from the sending host's mail queue and the receiving host's mail service moves the message from the mail queue to the recipient's mailbox. If the local host has no additional messages to send, the protocol then allows for the two systems to interchange roles—that is, the sending host becomes the receiving host and the receiving host becomes

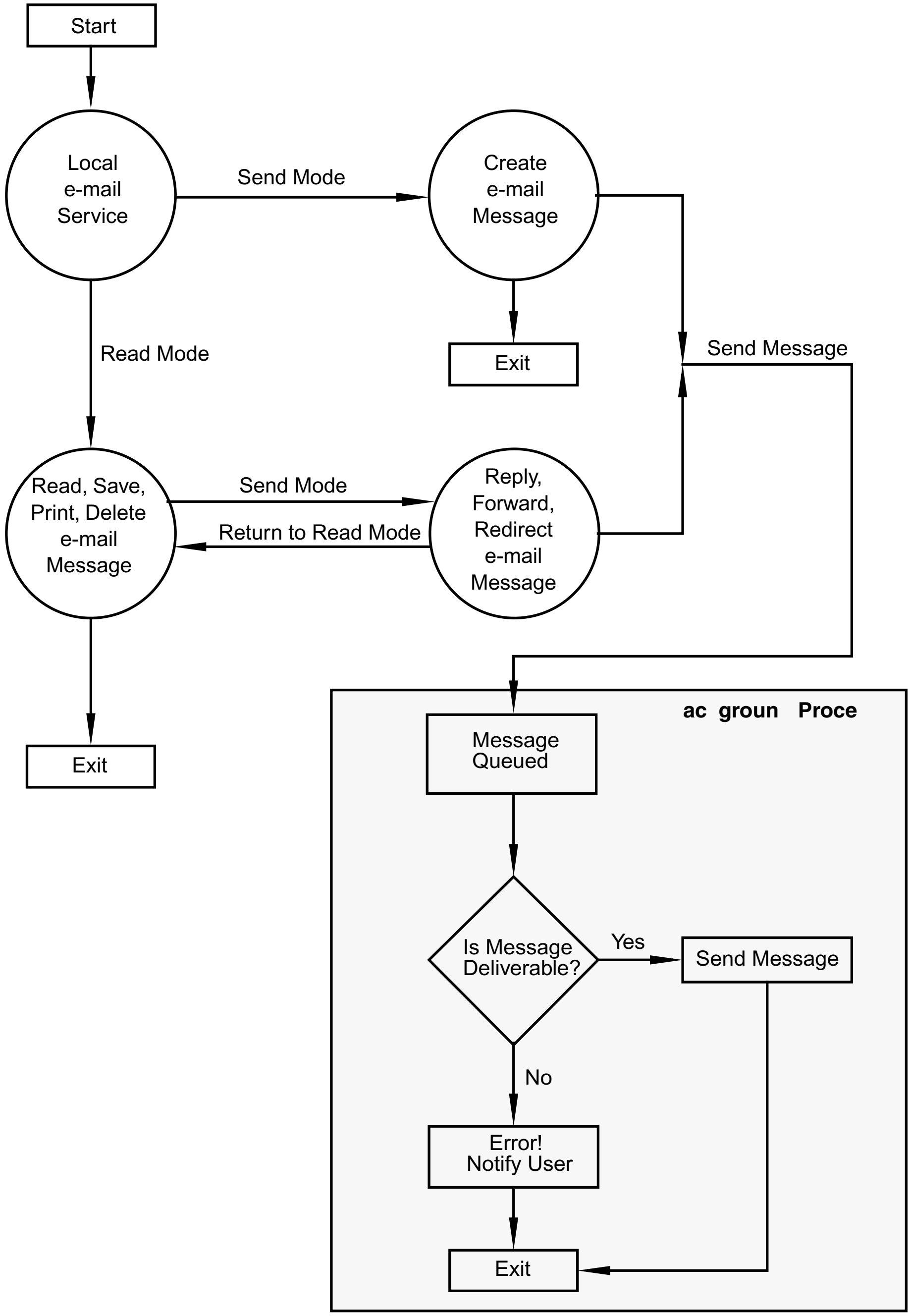


FIGURE 3.8 Generic flow of a typical e-mail service residing on a local host.

> the sending host. This allows mail to flow in the opposite direction. Thus, the host that was originally receiving mail can now send any mail stored in its queue destined for the host that was originally sending. Once all mail is delivered between the two systems, a command to terminate the connection is issued and if agreed by both hosts, the mail connection is closed.

Note that SMTP focuses only on transferring messages across a link from one host to another. The protocol has nothing to do with any other mail-related activities. These are provided by the local e-mail service. Further note that SMTP is TCP-based and hence provides a reliable mail delivery service since mail messages remain on the sending host until they are transferred successfully to the receiving host. For example, if the transfer of a mail message between hosts is not successful (e.g., a connection to the receiving host wasn't possible or a connection failed during the transfer of mail), then the message remains in the local mail queue. This local mail queue is periodically checked for undelivered mail by the system's mail program, and if any is found, a new mail connection is attempted. If mail is undelivered after a specified period of time (e.g., 5 days), it is ultimately returned to the sender with a note of explanation.

26. How are mail messages exchanged between two hosts?

The transfer of mail messages between two hosts via SMTP is performed using a specified set of commands defined by the protocol. The actual operation of the protocol begins with the initiating host (i.e., the SMTP sender) establishing a TCP connection to the destination host (i.e., the SMTP receiver). We will review how a TCP connection is established later when we discuss the transport layer. Once a connection is established, the SMTP sender and receiver engage in a "conversation" that involves a succession of commands and responses exchanged between sender and receiver. For each command issued by the sender, exactly one reply is provided by the receiver. A description of SMTP commands is given in Table 3.1; a description of the reply codes issued by a receiver in response to a command is given in Table 3.2; and Figure 3.9 contains an example of a typical exchange. Note that the UNIX command *mqueue* is used to list the contents of the local system's mail queue. Furthermore, an exchange similar to that shown in Figure 3.9 can be replicated by establishing a TELNET session to port 25 of a receiving host.

27. Can you give me an example of a real e-mail message and explain its parts to me?

We'll try. All e-mail messages that are exchanged via SMTP follow the same format, which is defined in RFC 822. This format consists of a header block, followed by a blank line, followed by the body of the mail message (Figure 3.10). The header block comprises individual lines that include a field name, followed by a colon, followed by specific information related to the field (called the field body). For example, the header line

Sender: jb@mailserver.bt.com

contains the field name, Sender, and the field body, jb@mailserver.bt.com, which is the e-mail address of the sender. The field name and body are delimited by a colon. A description of typical RFC 822 header fields is given in Table 3.3, and an example of an RFC 822-compliant message that uses some of these fields is shown in Figure 3.11.

TABLE 3.1 SMTP Commands

Name	Example of Use	Description
HELO	HELO mailserver.bt.com <CRLF>[a]	Identifies SMTP sender to SMTP receiver
MAIL	MAIL FROM:<jb@bt.com> <CRLF>	Begins mail exchange by identifying originator
RCPT	RCPT TO:<mickie1@ba.org<CRLF>	Identifies mail recipient
DATA	DATA<CRLF>	Signifies that what follows is mail message
SEND	SEND FROM:<jb@bt.com> <CRLF>	Send mail message to terminal
SOML	SOML FROM:<jb@bt.com> <CRLF>	"Send or Mail"; Send mail message to terminal *or* mailbox
SAML	SAML FROM:<jb@bt.com> <CRLF>	"Send and Mail"; Send mail message to terminal *and* mailbox
RSET	RSET <CRLF>	Specifies current mail transaction is to be aborted
VRFY	VRFY <user><CRLF>	Requests SMTP receiver to confirm that "user" is a valid user
EXPN	EXPN <mail-list><CRLF>	Requests SMTP receiver to expand given "mail-list" if valid
HELP	HELP [SMTP Command]<CRLF>	SMTP help command
NOOP	NOOP<CRLF>	No operation; generates OK from receiver
QUIT	QUIT<CRLF>	Ends current SMTP session by closing the TCP connection
TURN	TURN<CRLF>	Reverses role of SMTP receiver and sender

[a]<CRLF> means Carriage Return, Line Feed

28. Sometimes when I get an attachment with my e-mail I see something about MIME. What is this and how does it relate to SMTP?

MIME stands for *Multipurpose Internet Mail Extension* and is defined in RFC 1521. It was developed to address one of SMTP's deficiencies. You see, RFC 822 was designed for sending messages containing lines of ASCII text only. This avoids the problems that are inherent in supporting other character sets. By restricting messages to consist of only ASCII text, it is not necessary, for example, to determine a binary representation for sending binary files or to have to translate between a standard character set and the character set of the local host. The combination of a standard header format and support for only ASCII text makes it relatively easy to process and send mail messages across heterogeneous systems. Unfortunately, this simple formatting feature limits its functionality. In today's world of multimedia, users want to be able to exchange nontext files such as graphic, video, or audio in addition to plaintext files. This is where MIME enters the picture.

MIME is an extension of the current SMTP-based mail system; it is not a replacement. Specifically, MIME extends the concept of what we call e-mail by providing support for different data types and for complex message bodies. The MIME specification includes

TABLE 3.2 SMTP Reply Codes (Extracted from RFC 821)

Code	Description
211	System Status, or system help reply
214	Help message (Information on how to use the receiver or the meaning of a particular non-standard command; this reply is useful only to the human user)
220	<domain> Service ready
221	<domain> Service closing transmission channel
250	Requested mail action okay, completed
251	User not local; will forward to <forward-path>
354	Start mail input; end with <CRLF>.<CRLF>
421	<domain> Service not available, closing transmission channel (This may be a reply to any command if the service knows it must shut down)
450	Requested mail action not taken: mailbox unavailable (e.g., mailbox busy)
451	Requested action aborted: local error in processing
452	Requested action not taken: insufficient system storage
500	Syntax error, command unrecognized (This may include errors such as command line too long)
501	Syntax error in parameters or arguments
502	Command not implemented
503	Bad sequence of commands
504	Command parameter not implemented
550	Requested action not taken: mailbox unavailable (e.g., mailbox not found, no access)
551	User not local; please try <forward-path>
552	Requested mail action aborted: exceeded storage allocation
553	Requested action not taken: mailbox name not allowed (e.g., mailbox syntax incorrect)
554	Transaction failed

new message header fields, definitions for new content formats, and definitions for transfer encodings. The new header fields provide information about the message body; the new content formats identify the type of content the message body contains (e.g., text, audio, video); and the transfer encodings enable data to be re-encoded into a seven-bit short-line format so they can be delivered using a seven-bit mail transport protocol such as SMTP, which restricts mail messages to lines no longer than 1000 characters. Thus, when you receive a message with a nontext attachment, for example, your MIME-enhanced SMTP e-mail program reconstructs the nontext data into their original form, which can then be read by an appropriate application. Table 3.4 contains a description of MIME header fields, Table 3.5 contains a description of MIME content types, and Table 3.6 contains a description of MIME transfer encodings.

SMTP Sender (S): mailserver.bt.com
SMTP Receiver (R): ba.org

	S:	(mailserver.bt.com establishes a connection to ba.org)
	R:	220 ba.org ESMTP Sendmail 8.9.3/8.9.1; Tue, 16 Oct 2001 12:51:49 -0400 (EDT)
	S:	**HELO mailserver.bt.com**
	R:	250 ba.org Hello mailserver.bt.com, pleased to meet you
	S:	**MAIL FROM:<jb@mailserver.bt.com>**
	R:	250 <jb@mailserver.bt.com>... Sender ok
	S:	**VRFY <mickie1>**
	R:	252 Cannot VRFY user; try RCPT to attempt delivery (or try finger)
	S:	**RCPT TO:<mickie1>**
	R:	250 <mickie1>... Recipient ok
	S:	**DATA**
	R:	354 Enter mail, end with "." on a line by itself
	S:	**Hi,** **Just a quick message to demonstrate some of the SMTP commands.** **.**
	R:	250 MAA25178 Message accepted for delivery
	S:	**QUIT**
	R:	221 s1.cfe.fit.edu delivering mail

Description

A connection to the receiving host is made by sending the host.

The sending host identi es itself to the receiving host.

The sending host indicates to receiving host that it has mail from user jb.

The sending host attempts to verify that the recipient, mickie1, on the receiving host is a valid user but the receiving host responds that it cannot execute a VRFY command.

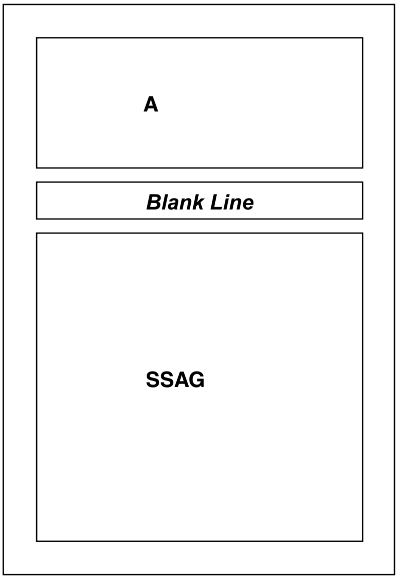
The sending host identi es to the receiving host that its mail is intended for mickie1; the receiving host acknowledges that mickie1 is a valid recipient on its system.

The sending host issues a DATA command, which implies that all subsequent text is to be treated as a mail message. The receiving host responds accordingly and instructs sending host to end the mail message by placing a period on a line by itself.

The sending host enters the mail message.

The sending host issues a QUIT command, which causes the receiving host to deliver the mail message and terminate the connection.

FIGURE 3.9 Sample SMTP command/response exchange between an SMTP sender and receiver.



IGURE 3.10 RFC 822 mail message format.

ea er

```
From: jb@mailserver.bt.com Thu Oct 25 08:54:10 1990
Return-Path: <jb@mailserver.bt.com>
Received: from mailserver.bt.com by ba.org (4.1/SMI-DDN) id AA07077;
Thu, 25 Oct 90 08:54:07 EDT
Date: Thu, 25 Oct 90 09:12:25 EDT
From: jb@mailserver.bt.com (Janie Busch)
Message-Id: <9010251312.AA04733@mailserver.bt.com>
To: mickie1@ba.org
Subject: Re: E-mail Address
```

Blank Line

o o e age

```
Mickie,
Thanks for sending me the address.
Ciao Bello.

JB
```

IGURE 3.11 Sample RFC 822 mail message.

TABLE 3.3 Frequently Used RFC 822 Header Fields

Field	Description
Date	The date the message was sent
From	The name and address of the sender
To	The name and address of the recipient
Subject	A statement that describes the content of the message
Return-Path	Identifies a path back to the originator of the message (added by mail transport service)
Reply-To	Identifies the address where replies should be sent if different than the *From* line address
In-Reply-To	Identifies the message to which the current message is replying
Cc	Address to which copy of the message is being sent
Message-ID	Message identification
Received from	The name of the SMTP sending host
Received by	The name of the SMTP receiving host
Received via	The physical path on which the message arrived
Received with	The mail protocol the message used for delivery
Received id	The receiver message id

TABLE 3.4 MIME Header Fields Defined in RFC 1521

Field	Description
MIME-Version	Provides a version number that declares a message to be MIME-compliant to a MIME specification. Version 1.0 implies conformity to RFC 1521 and RFC1522.
Content-Type	Specifies the type and subtype of data contained in message body. (See Table 3.5.)
Content-Transfer-Encoding	Specifies the encoding used for data transport.
Content-ID	Identifies MIME entities in various contexts. (Optional)
Content-Description	Text-based description of the data contained in the message body. (Optional)

29. I think I'm beginning to get the picture. Where does POP fit into this scheme?

In the early days of the Internet, mail messages were stored in users' mailboxes, which were located on a centralized host. Typically, these hosts were running the UNIX operating system, and access to e-mail meant logging into the e-mail server—either directly via a hard-wired terminal or remotely via Telnet—and using the server's mail utility. Interacting with a server's mail utility was not a pleasant experience for many users because UNIX mail service was a text-based program that usually required special commands (called *tilde*

TABLE 3.5 MIME Content Types Defined in RFC 1521

Type	Subtype	Description
Text (Data composed of textual information using different character sets)	• Plain	• Plain, unformatted text. This is the primary Content-Type for Internet e-mail.
Multipart (Data composed of multiple, independent sections)	• Mixed	• Used when the body parts are independent (i.e., they may contain data of any content type) but need to be placed together in a specific order.
	• Alternative	• Indicates that the message body contains the same data represented in different formats. The formats are ordered by increasing faithfulness to original content.
	• Digest	• Indicates that the document data consists of Content Type = Message/RFC 822.
	• Parallel	• Similar to Mixed except there is no specific data order and hence may be viewed simultaneously
Message (Data is an encapsulated mail message)	• rfc822	• Syntax of encapsulated message is RFC 822 compliant.
	• Partial	• Enables large messages to be fragmented for delivery.
	• External-Body	• Message body contains a pointer that points to an external source containing a large message.
Application (Data does not fit any other category)	• Octet-stream	• Indicates that message body contains binary data.
	• Postscript	• Indicates message body is Adobe Postscript data.
Image (Data composed of a still image)	• JPEG	• Indicates data in JPEG format using JFIF encoding.
	• GIF	• Indicates data in GIF format.
Audio (Message composed of voice data)	• Basic	• Message body is audio encoded using 8-bit ISDN mu-law (PCM). Assumes a sample rate of 8000 Hz and a single channel.
Video (Data composed of moving graphic image)	• MPEG	• Indicates message body consists of video data coded according to the MPEG standard

commands) to facilitate the manipulation of mail messages. For example, in the AT&T System V mail program, ~v invoked the vi editor, ~p displayed the contents of the current mail message, and ~r *filename* inserted the contents of the file, *filename*, into the current mail message. Although programs like *pine* helped tame the interface problem, interacting with the system's mail service was still unpleasant for many.

TABLE 3.6 MIME Transfer Encodings Defined in RFC 1521

Encoding	Description
7bit	Data represented as 7-bit U.S. ASCII; no encoding is done.
8bit	Data contains "short lines" but also might include non-ASCII characters, which contain octets with the high-order bit set; no encoding is done.
binary	Data contains non-ASCII characters and line size most likely not RFC821-compliant; no encoding is done.
quoted-printable	Used to encode data that consists mostly of ASCII characters. Non-ASCII characters are encoded using one of several rules defined in RFC 1521. Since most of the data being encoded in ASCII, the encoded representation is still recognizable by humans.
base64	Encoding scheme that represents three octets of data as four 6-bit characters. This scheme is similar to the one used for Pretty Good Privacy (PGP).
x-token	Any proprietary encoding scheme developed by a software developer.

In addition to the interface problem, there was the mail storage problem. As desktop computing grew, more and more users began accessing their local e-mail server from their desktop PCs using Telnet. This was not convenient on two fronts. First, users were unable to take advantage of their PC's native operating system features when interacting with the mail program—when it came to e-mail, users' PCs had to be employed as dumb terminals. Second, users' mail messages were stored on a remote server and not directly on their PCs. What users wanted was the ability to transfer the contents of their mailboxes from the mail server to their desktop PC and to use the full features of their PC's native operating system to interact with e-mail. It should be noted that there was nothing stopping users from creating an e-mail message locally on a PC using a word processing program like Word. The problem was sending this message to the mail server for delivery to the message's recipient and to transfer the contents of the users' mailboxes back to the PC. This two-way transfer could be done using a file transfer protocol such as Kermit, but such protocols are terminal-based and not network-based. The *Post Office Protocol* (POP) was designed to effect this two-way transfer in a simple and convenient manner.

POP is very similar to SMTP and uses an SMTP-like command set. Two versions of POP are available. POP2 is defined in RFC 937 and uses well-known port number 109; POP3 is defined in RFC 1939 and uses port 110. PC-based e-mail programs such as Eudora provide users with an interface that takes full advantage of a PC's native operating system and uses the client version of the POP protocol for e-mail transfers be-tween the PC and the designated mail server, which runs the POP server version. Although the POP2 and POP3 command sets are different and incompatible, both protocols operate in a similar manner. Following is the basic operation of the POP3 protocol:

> The server invokes the POP3 service by listening on TCP port 110. When a client wants to use the service, it establishes a TCP connection with the server. When the connection is first established, the server sends a greeting that identifies the POP3 process. The current session

then enters into an authorization state in which the client identifies itself to the server by sending the user's username and password for the mail account located on the server. Assuming valid authorization, the session then enters into a transaction state in which the client directs the server to manipulate the user's mailbox relative to the client e-mail program's configurations. These can include (among others) retrieving all unread mail, deleting messages, and sending queued messages.

As is the case with SMTP, each POP command issued by the client generates a response by the server. Thus, the client and POP server exchange commands and responses, respectively, until the connection is closed or aborted. In POP3, only two response types are defined: +OK and –ERR. The POP3 command set is shown in Table 3.7.

30. Let me try to summarize my understanding of these three protocols:

- **SMTP is the protocol the Internet uses for sending mail messages across the Internet between two hosts.**
- **MIME is an extension of SMTP that makes it possible for nontext data such as graphics, audio, or video to be sent electronically using SMTP.**
- **POP is a protocol for retrieving mail from a mail server.**

How did I do?

Not bad. To really drive home the distinction, check out the configurations of your e-mail program. For example:

- In Eudora, the "Checking Mail" component found under "Settings" provides a place for you to enter the name of your SMTP server for sending mail; the "Checking Mail" component requires you to select between two mail protocols: POP and IMAP; and the "Attachments" component permits you to select MIME as the encoding method for sending attachments and to receive MIME digests when receiving attachments.
- In Microsoft's Internet Explorer, under "Preferences," the e-mail component has a place for you to specify your SMTP host for sending mail and to select the protocol (POP or IMAP) for receiving mail.
- In Netscape's Communicator, under "Preferences" the "Mail Servers" component of "Mail & Newsgroups" enables you to specify an incoming mail server as well as an outgoing SMTP server. Furthermore, once you "add" an incoming mail server, you can then select its protocol, POP or IMAP.

31. I got it! One last e-mail question. What's IMAP?

IMAP, which stands for *Internet Message Access Protocol*, is similar to POP in that it enables a client e-mail application such as Eudora or those that are part of your Web browser to access mail messages from an SMTP mail server. The primary difference between the POP and IMAP protocols is that POP is best used when all of your e-mail messages are retrieved by a single computer. For example, if you work from home and use the one computer for all of your e-mail activities, then POP is the preferred protocol for

TABLE 3.7 POP3 Commands

Name	Syntax	Description
USER	USER *username* <CRLF>[a]	Indicates the user's account name on the mail server
PASS	PASS *password* <CRLF>	Indicates the user's password on the mail server
STAT	STAT <CRLF>	Report of the number of unread messages
LIST	LIST [*m*] <CRLF>	List the size of all messages or of only message number *m*
RETR	RETR *m* <CRLF>	Retrieve message number *m*
DELE	DELE *m* <CRLF>	Delete message number *m*
NOOP	NOOP <CRLF>	No operation; generates +OK from receiver
RSET	RSET <CRLF>	Unmark all messages marked as deleted
TOP	TOP *m n* <CRLF>	Retrieve the first *n* lines of message *m* (e.g., TOP 6 10 retrieves the first 10 lines of message number 6)
UIDL	UIDL [*m*] <CRLF>	Provides unique-id information about message *m*
APOP	APOP *name digest* <CRLF>	Alternative POP authentication method that does not involve sending a clear password over the network during frequent, intermittent POP sessions.
QUIT	QUIT <CRLF>	• If entered from authorization state, current session terminates (TCP connection is closed). • If entered from the transaction state, current session enters update state and all messages marked for deletion are deleted; any mailbox locks are then released and the TCP connection is closed.

[a] <CRLF> means Carriage Return, Line Feed

retrieving mail. IMAP, on the other hand, was designed to enable you to access mail messages from more than one computer. For example, e-mail stored on an IMAP server can be accessed or manipulated from your home PC, from a workstation at your office, and from a notebook computer while traveling. Furthermore, unlike POP, IMAP enables you to do this without transferring messages or files back and forth among these computers. This functionality is needed by people, particularly business people, who rely on e-mail as part of their daily activities and use different computers when conducting activities.

32. I am still a little confused. Can you try again to explain the differences between the two protocols?

Sure. Think of it this way: POP is an *offline* protocol. In a typical POP session, you download all new e-mail messages from your Internet service provider's mail server to your home computer. These messages are also deleted immediately (usually) from your ISP's server after they are downloaded. This is generally an acceptable procedure if your home computer is the only computer you use.

Suppose, though, that your ISP is your employer and that you use a computer at work and another computer at home for e-mail messaging. With POP, when you retrieve your e-mail messages at work, they are now stored on your office computer. This means that when you get home you do not have access to them on your home computer. If you now check your e-mail from home, the messages are stored on your home computer and not readily accessible by your office computer. In other words, your e-mail messages are local to the computer to which you downloaded them. Get the picture?

33. Yes I do. I can relate to this very easily. I usually have to transfer my messages from one machine to another. It's a pain because I never know which mailbox is current and I sometimes have to forward or redirect messages to myself. So IMAP resolves this problem?

Yes. IMAP is both an offline (like POP) and *online* protocol. As an online protocol, IMAP enables you to manipulate your mailbox that resides on your e-mail server as if it were local to your computer. Instead of downloading and deleting them as is the case with POP, you *manipulate* your messages, which maintain their residence on the IMAP server. You can mark messages with flags such as "seen," "answered," and "deleted," you can define your own flags, and you can set up mail folders and store your messages within these folders. You can also search these folders for specific messages. All of these operations or manipulations are done by your local computer. For example, let's assume that you check your e-mail at the office just before you are ready to leave for home and receive several messages that require your attention. Instead of staying at the office and answering them, you can flag them and go home. Now, later that evening, when you check your e-mail from your home computer, these messages will be received again. Thus, IMAP permits multiple computers to access the same e-mail account easily, readily, and efficiently.

Incidentally, you might be interested in knowing that IMAP was developed in 1986 at Stanford University. There is also an IMAP companion protocol, the *Application Configuration Access Protocol* (ACAP), which was developed at Carnegie Mellon University.

34. Okay. Let's move on to the TELNET protocol.

TELNET is a general purpose remote login protocol included in the TCP/IP protocol suite; it is defined in RFC 854. TELNET provides *virtual terminal service*, which enables a user who has a direct connection on a local host to log into a remote host and interact with this remote host as if the user were directly connected to it. Once a remote login session is established, the TELNET protocol provides the necessary mechanism for keystrokes entered on the local host to be passed directly to the remote host. TELNET uses TCP for reliable data transport and is accessible via the *telnet* application program, which is a built-in function on UNIX- and Windows-based (i.e., Windows 95/98/2000) machines. Third-party telnet application programs such as NCSA Telnet for Macintosh are also available.

35. How does TELNET work?

Without going into too much detail (we want to stick to our basic philosophy of providing an overview of networking), here's an overview of the process. The TELNET protocol defines an imaginary, bidirectional character device called a *network virtual terminal*

(NVT), which is designed to maintain terminal characteristic information about the terminals located at each end of a TELNET connection. Through the NVT, a standard network-wide representation of a terminal is provided so that both client and server have an agreed upon convention of how a terminal appears over the network. When a TELNET connection is first established, each end is assumed to originate and terminate at the NVT, and both client and server map their local device characteristics and conventions to the NVT. The TELNET protocol also permits the negotiation of additional services beyond the minimal terminal services provided by the NVT. Thus, if a client and server want to use a different set of conventions for a particular connection, then they can negotiate specific options to implement. For example, each end of a TELNET connection might consist of devices that support more sophisticated terminal characteristics. In this instance, the two terminals might want to change the character set they use for this connection. In another example, hosts might want to invoke a data format change from ASCII to binary. Regardless of what changes are made, TELNET's option negotiation service provides client and server with the ability to enhance their communications.

36. I operate a UNIX workstation and have used something called *rlogin* in addition to TELNET to log into a remote host. How different is TELNET from rlogin?

As a remote login protocol, TELNET is not as sophisticated as other protocols that provide similar capability. One such protocol is the UNIX-based remote login facility, *rlogin*, which exports most of a user's local environment to the remote host. This enables the remote terminal type to be equivalent to the local terminal environment (as defined by the environmental variable TERM), and if windows are being used, window size is also maintained. Given its simple nature, a TELNET session might not pass the local computing environment to the remote host. This means that when a remote connection is established, local terminal settings most likely will not be maintained at the other end of the connection. This can result in terminal emulation problems, which can lead to a high degree of user frustration, particularly during an editing session. Fortunately, most telnet application programs provide some form of keyboard mapping that minimizes this problem. Along this same line, the TELNET protocol also defines a standard representation of five control functions. These include: Interrupt Process, which suspends, interrupts, aborts, or terminates a process; Abort Output, which stops output being generated by a process from being sent to a user's terminal; Are You There, which enables a user to receive visible confirmation that the remote system is still "up"; Erase Character, which provides the equivalent of a delete key; and Erase Line, which erases the current line of input. Finally, some remote hosts are IBM-based devices that do not support ASCII terminal controllers. A virtual terminal connection to such sites requires special software that emulates an IBM 3270 terminal. An application called tn3270 is designed for such connections.

37. Using TELNET can be a pain sometimes because it always is so slow.

You're right. When using the TELNET protocol for virtual terminal service across the Internet, you need to exercise patience. In the presence of network congestion or high remote system activity, it is not uncommon for response times to be in the tens of seconds. It is also common for TELNET connections to simply time out, which effectively closes the TCP connection. The "Are You There" control function is often helpful to confirm that a connection is still active. Frustrated TELNET users can probably appreciate the following adage:

- *If you can see it and touch it, it's physical.* (This is a direct connection.)
- *If you can see it but can't touch it, it's virtual.* (This is a TELNET connection.)
- *If you can't see it and can't touch it, it's gone.* (The TELNET connection times out.)

38. How different is FTP from TELNET?

The *File Transfer Protocol* (FTP) is the standard application protocol of the TCP/IP protocol suite that is used for exchanging files between two systems. So one difference is that FTP is used for file transfers and TELNET is used for remote logins.

39. Okay. But isn't an FTP session really a remote login?

Yes it is. In fact, part of the FTP protocol is based on the TELNET protocol. You see, an FTP session actually involves two separate connections. One connection supports the data transfer process itself; the second connection handles various control processes relative to the actual FTP session. To keep things simple, FTP uses TELNET's basic NVT to exchange data across this control connection. (No option negotiation is allowed.)

40. How does this work?

A conceptual model of these two processes and connections is shown in Figure 3.12. An FTP server process listens on *well-known port* 21 for an initial connection request from an FTP client. Once a connection is established, the control process creates a separate TCP connection for a particular data transfer. Since the same port number cannot be used, the server uses well-known port 20. The client randomly assigns and reports the *port number* it will use for the data transfer connection to the server via the control connection.

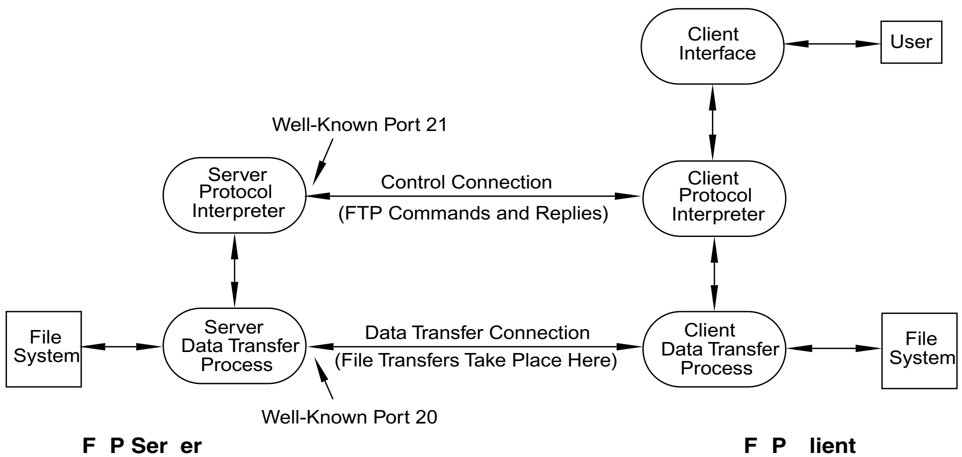


FIGURE 3.12 Conceptual model of the FTP process. Source: Adapted from RFC 959.

TABLE 3.8 Commonly Used FTP Commands

Command Syntax	Description
get *remotefile* [*localfile*]	Transfers one file from the remote host to the local host (i.e., we *get* a file). If used with the optional argument, *localfile*, then the file is stored on the receiving host with the name, *localfile*. If the optional argument is not specified, then the transferred file is given the same name on the receiving host as it has on the sending host.
mget *remotefile-1 ... remotefile-n*	Transfers multiple files from remote host to local host.
put *localfile* [*remotefile*]	Transfers one file from the local host to the remote host (i.e., we *put* a file). If used with the optional argument, *remotefile*, then the file is stored on the receiving host with the name, *remotefile*. If the optional argument is not specified, then the transferred file is given the same name on the receiving host as it has on the sending host.
mput *localfile-1 ... localfile-n*	Transfers multiple files from local host to remote host.
cd	The *change directory* command; used to change the remote directory from its current location to a new location.
lcd	The *local change directory* command; used to change the local directory from its current location to a new location.
dir	The *directory* command; used to list the contents of the current remote directory.
pwd	The *print working directory* command; used to display the current working directory of the remote host.
mkdir	The *make directory* command; used to create a new directory on the remote host.

TABLE 3.9 Commonly Used FTP Options

Option	Description
ascii	This option sets the *representation type* of subsequent transferred files to *network ASCII*. Use this option to transfer ASCII formatted files.
binary	This option sets the *representation type* of subsequent transferred files to *image*. Use this option to transfer binary formatted files.
case	By enabling this option, remote computer files whose names are in uppercase letters are written to the local directory using lowercase letters.
prompt	This option toggles interactive prompting, which occurs when multiple files are being transferred using *mget* or *mput*. If *prompt* is enabled when *mput* or *mget* is executed, then prior to transferring each file of a file list, users are prompted to confirm that they want to transfer a file.

A separate data transfer connection (and corresponding data transfer process) is established for each file transfer. There is only one control connection, however, for an FTP session. Thus, once the control connection is closed, the FTP session is terminated, and any data transfer processes that are active also get terminated by the respective hosts.

Assuming valid authorization, the user is placed in "command mode," from which all FTP commands and options are enabled or disabled. Commands and corresponding replies are transferred across the control connection. Many Internet servers, called *anonymous FTP* servers, run a restricted FTP program that permits access to their systems using the login name *anonymous* and the user's e-mail address or local login name as the password.

A list of FTP commands and options is generated by entering "help" or a question mark at the FTP command prompt. A list of commonly used FTP commands is given in Table 3.8, and a list of commonly used FTP options is given in Table 3.9. Unless otherwise specified, all transferred files are stored in the current working directory of the receiving host.

41. What's a well-known port?

We briefly mentioned the concept of port numbers at the end of Chapter 2 when we presented an example of connectionless and connection-oriented service. We will discuss port addresses in more detail later in this chapter when we discuss TCP/IP's transport layer. For now, though, you may think of a *port address* or a *port number* as a location that uniquely identifies a specific user application such as e-mail. All network applications are uniquely identified by their port numbers. For example, SMTP's port number is 25 and TELNET uses port number 23.

42. What program do I use for FTP?

The user interface to the FTP protocol is provided by the *ftp* application program, which is available on UNIX- and Windows-based (i.e., Windows 95/98/2000) machines. Third-party ftp application programs such as Fetch for the Macintosh are also available. Since FTP uses TELNET for its control connection, a user must first establish a valid login (i.e., valid username and password) on the remote system before a file transfer can be effected. Finally, the protocol can also be invoked directly from your Web browser. For example, instead of entering an address of the familiar form "http://," you begin the Web address with "ftp://." More on this later.

43. Why wait? I've had my fill of e-mail TELNET, and FTP. Let's talk about HTTP.

OK. The *Hypertext Transfer Protocol* (HTTP) is the TCP/IP application-layer protocol on which the World Wide Web (WWW) is based. First implemented in 1990, HTTP is considered a request–response protocol. An HTTP client program establishes a connection to an HTTP server program and requests a specific service from the server; the HTTP server program accepts this connection and responds to the client's request. In short, HTTP provides the specifications for how Web clients such as a browser request Web pages from Web servers as well as how Web servers transfer documents to Web clients. A Web client's request message is made through a *user agent*, which is a Web browser, an editor, a search engine, or any other end user tool. A response message is provided by the server after it has received and interpreted the request message.

44. So the entire format of a Web page is defined by HTTP?

Not quite. The format of how Web pages are transmitted across the Internet is defined by HTTP. Both request and response messages for HTTP version 1.1 are defined in RFC 2616 and follow the generic format of RFC 822. They consist of a start line, a header field that comprises zero or more headers, a blank line to signify the end of the header field, and a message body that contains the resource content (called an *entity*). Request header fields are used by clients to pass additional information about themselves and about a particular request to a server. Table 3.10 contains a list and description of request header fields. Response header fields are used by servers to pass additional information about themselves to clients as well as information about further access to a requested resource. Table 3.11 contains a list and description of response header fields.

TABLE 3.10 HTTP Request Header Fields—Used by Clients to Pass Additional Information About Themselves and About a Particular Request to a Server

Field	Description
Accept	Specifies the media types (e.g., audio, image, text) the client will accept as a response by the server. If this field is absent, then it is assumed that the client will accept all media types.
Accept-Charset	Specifies the character sets the client will accept as a response by the server.
Accept-Encoding	Similar to the Accept field except it restricts the content-codings (e.g., gzip, compress, deflate) the client will accept as a response by the server for the current request.
Accept-Language	Similar to the Accept field except it restricts the set of client-preferred natural languages.
Authorization	Used to authenticate a client to a server; consists of the user agent's *credentials* that are relative to the requested resource.
Expect	Identifies the specific behaviors the client requires from the server when it responds to the current request.
From	Provides the server with the user agent's human user's Internet e-mail address.
Host	Specifies the Internet host and port number of the resource being requested.
If-Match	A conditional list of entity tags that the server uses to verify which entities the client has already received. If the client's listed tag matches a server's entity tag, then the server responds with that entity.
If-Modified-Since	A conditional time clause that directs the server not to return the requested resource if the resource has not been modified since the specified time.
If-None-Match	A conditional list of entity tags that indicates to the server which entities the client presently has that are not current.

(continued)

TABLE 3.10 HTTP Request Header Fields—Used by Clients to Pass Additional Information About Themselves and About a Particular Request to a Server (continued)

Field	Description
If-Range	Indicates to the server that if the specified entity is unchanged, then the server is to respond with only missing entities; otherwise, an entire new entity should be returned. Should be used with the Range header.
If-Unmodified-Since	A conditional time clause that directs the server to perform the requested operation if the requested resource has not been modified since the specified time.
Max-Forwards	Limits the number of times a request can be forwarded en route to the origin server.
Proxy-Authorization	Used to identify a client to a proxy server that requires authentication.
Range	Byte ranges that specify the sequence of bytes that represent the length of an entity; enables a client to request specific parts of an entity as designated by the entity's byte range.
Referer	Allows a client to specify a resource's URI from which the Request-URI was obtained; enables a server to generate a list of resource back-links.
TE	Specifies the extension transfer-codings the client will accept as a response by the server, and whether or not it is willing to accept trailer fields in a chunked transfer-coding.
User-Agent	Provides the server with information about the user agent originating the request.

In addition to request and response header fields, all request and response messages contain general header fields. These are listed and described in Table 3.12. If a message contains a message body, which is used to transfer a specific resource or entity, then entity header fields are used to define the contents. Entity header fields are listed and described in Table 3.13. Finally, response messages also contain a three-digit status code, which the server returns to provide information about its attempt to understand and satisfy a particular client request. Table 3.14 contains a list and description of these result codes.

45. Can you give me an example?

Sure. First recall that request messages follow the generic format of e-mail messages as defined in RFC 822 and shown in Figure 3.10. With this in mind, request messages comprise a header block followed by a blank line (denoted as CRLF for carriage return/line feed) followed by a message body. The header block consists of a request line and header lines. The general format is as follows:

```
Request        = Request-Line
                  (( general-header
                     response-header
                     entity-header )  CRLF
                  CRLF
                  [ message-body ]
```

TABLE 3.11 HTTP Response Header Fields—Used by Servers to Pass Additional Information About Themselves to Clients and to Provide Information About Further Access to a Requested Resource

Field	Description
Accept-Ranges	Informs client of the server's acceptance of range requests for a resource. An accept-range of "none" indicates that the client is not to attempt a range request.
Age	Provides a time estimate in seconds of how long it has been since a particular response was generated at the origin server.
ETag	Provides the current value of the entity tag for the requested resource.
Location	Identifies the location where the current request must go to either be completed or to identify a new resource; used to redirect the recipient to the appropriate location if different from the Request-URI.
Proxy-Authenticate	Provides information about the authentication scheme and parameters applicable to the proxy for the current Request-URI; used with a Proxy Authentication Required (status code 407) response to inform the requesting client that a proxy authentication is required.
Retry-After	Indicates how long service is expected to be unavailable to the requesting client; used with a Service Unavailable (status code 503) response.
Server	Provides information about the software the origin server used to address the request.
Vary	Identifies the request-header fields that determine whether or not a cache may use the response to reply to a subsequent request without re-validation.
WWW-Authenticate	Consists of at least one challenge that indicates the authentication scheme(s) and parameters applicable to the Request-URI; included in all Unauthorized (status code 401) response messages.

The Request-Line has the following format:

Request-Line = *Method* <space> *Request-URI* <space> *HTTP-version* CRLF

The *Method* component specifies what is to be performed on the requested resource, which is identified by the *Request-URI* component. The *HTTP-version* is simply the version number of the HTTP protocol being used. Sample *Methods* and corresponding descriptions follow.

- OPTIONS refers to the communication options that are available relative to the requested resource.
- GET requests that whatever information is identified by the requested resource (*Request-URI*) be retrieved.
- HEAD performs the same function as GET except that the server does not include a message body in its response to the request.
- POST requests that the origin server accept the contents of the request as a new subordinate of the resource specified by the *Request-URI* component in the

TABLE 3.12 HTTP General Header Fields

Field	Description
Cache-Control	Provides specific directives that must be followed by all caching mechanisms throughout the length of a connection (called the request/response chain). The directives are designed to prevent caches from adversely interfering with a request or response.
Connection	Allows the sender to specify options that are desired for a given connection.
Date	Specifies the date and time the message was originated.
Pragma	Provides implementation-specific directives that might apply to any recipient along the request/response chain.
Trailer	Indicates that the given header fields are present in the trailer of a message encoded with chunked transfer-coding.
Transfer-Encoding	Indicates the type of transformation applied to the message body so the message can be transmitted safely between sender and recipient.
Upgrade	Specifies any additional protocols the client supports and would like to use; informs the server of the client's protocol capabilities and preferences in the event the server finds it appropriate to change protocols.
Via	Identifies the intermediate protocols and recipients that a request or response is subjected to en route between the user agent and server, or between the origin server and client; used by gateways and proxies, and is analogous to a mail message's Receive field defined in RFC 822.
Warning	Contains supplemental information about a message that might not be reflected in the message itself.

request line. A POST is necessary when existing resources are be annotated, when messages are to be posted to a newsgroup or mailing list (or similar), or when data are to be processed (e.g., when you complete a Web form).

- PUT requests that the contents of the request be stored under the resource identified by the *Request-URI* component.
- DELETE requests that the origin server delete the resource specified by the *Request-URI* component.
- TRACE is for testing or diagnostic purposes.
- CONNECT is for proxy connections (discussed later).

Following is a simple HTTP request message from a client to a server:

```
GET http://www.fit.edu/AcadRes/graduate/gradpol/index.html HTTP/1.1
Host: www.fit.edu
Accept-Language: en-us, en
Connection: close <CRFL>
<CRLF>
```

TABLE 3.13 Entity Header Fields—Defines the Contents of an HTTP Message Body

Field	Description
Allow	Specifies the "methods" supported by the resource identified in the Request-URI. Examples of methods include: GET, which is a request to retrieve the resource and return it in the entity body; HEAD, which is identical to GET except the server returns only the response headers as it would with GET and not the message body; and DELETE, which requests that the origin server delete the specified resource.
Content-Encoding	Indicates any additional content codings that have been applied to the entity body as well as the decoding mechanisms necessary to obtain the media type referenced by the Content-Type header field. Content-Encoding enables a document to be compressed without losing its underlying media type identity.
Content-Language	Describes the natural language(s) of the intended audience for the enclosed entity.
Content-Length	Specifies the size (in number of octets) of the entity body.
Content-Location	Identifies an entity's accessible location if different from the requested resource's URI.
Content-MD5	Used to provide an end-to-end message integrity check of the entity body; the contents is an MD5 digest that has been applied to the entire entity body. Defined in RFC 1864.
Content-Range	Used to identify exactly where a partial entity is to be placed within a full entity body; provides the specific byte ranges of the partial entity relative to the full entity.
Content-Type	Specifies the media type of the entity body.
Expires	Specifies the expiration date and time of a response; used to determine when the response is considered stale. An Expire value that is one year in the future from the time the response is sent is interpreted as a response that "never expires."
Last-Modified	Specifies the date and time at which the origin server believes the resource was last modified.

The first line is the Request-Line and requests that the specified document be retrieved. The next three lines are header lines. The second line is a request header (Table 3.10) and specifies the Internet host of the requested resource. The third line is another request header and indicates that the user prefers U.S. English, but will accept any form of English. The fourth line is a general header (Table 3.13) and indicates that the connection will be closed after the completion of the response. The last line specifies a blank line that separates the header from the body message.

TABLE 3.14 HTTP Status Codes

1xx Codes: Informational—Request received; continuing process	
100 Continue	An interim response used to inform the client that the initial part of the request was received and was not rejected by the server; the client should continue with its request.
101 Switching Protocols	Server acknowledges client's request (indicated in the Upgrade general header field) to change application protocols for the current connection and will switch to the protocol(s) defined in the response's Upgrade header field.
2xx Codes: Successful—Action was successfully received, understood, and accepted	
200 OK	Request has succeeded.
201 Created	The request was completed and a new resource was created, which is referenced by the returned URI(s).
202 Accepted	The request was accepted for processing but processing has not been completed. (*Note*: It is possible that the current request might be disallowed once processing occurs.)
203 Non-Authoritative Information	The returned information contained in the entity header fields is not authoritative; it was acquired from a local or third party copy instead of from the origin server.
204 No Content	The server has completed the request but there is no information to return, hence the response message does not contain an entity body.
205 Reset Content	The server has completed the request and the user agent should reset the document view that initiated the request.
206 Partial Content	The server has fulfilled the partial GET request for the resource.
3xx Codes: Redirection—Further action is required by user agent to complete the current request	
300 Multiple Choices	The requested resource is not uniquely represented; information about each resource representation, including specific locations, is being provided so the user agent can select its preferred representation.
301 Moved Permanently	The requested resource has been assigned a new permanent URI; future references to this resource should use the returned URI.
302 Found	The requested resource resides temporarily under a different URI. future references to this resource should continue to use the requested URI.
303 See Other	The response to the current request is available under a different URI and should be retrieved using a GET method on that resource.
304 Not Modified	The client performed a conditional GET request and although access was allowed, the document was not modified.
305 Use Proxy	The requested resource is only accessible via a proxy, identified by the URI in the response header's Location field.
306 Unused	Used in a previous version of the specification; not currently used and is reserved.
307 Temporary Redirect	The requested resource resides temporarily under a different URI; future references to this resource should continue to use the requested URI.

(continued)

TABLE 3.14 HTTP Status Codes (continued)

4xx Codes: Client Error—Request contains bad syntax or cannot be completed	
400 Bad Request	The request contained malformed syntax and could not be understood by the server.
401 Unauthorized	The request requires user authentication.
402 Payment Required	Reserved for future use.
403 Forbidden	The server understood the request but refuses to act on it.
404 Not Found	The server could not find anything matching the requested URI; commonly used when the server does not want to reveal why the request was refused, or when no other response is available. (Sometimes referred to as *link rot* or *dead link*.)
405 Method Not Allowed	The specified method is not allowed for the requested resource.
406 Not Acceptable	The requested resource can only generate response entities with content characteristics that are different from those provided in the request message's Accept header field.
407 Proxy Authentication Required	The client must first authenticate itself with the proxy; similar to code 401.
408 Request Timeout	The client's request was not produced within the time the server was prepared to wait.
409 Conflict	The request was not completed due to a conflict with the current state of the resource.
410 Gone	The requested resource is not available at the server and no forwarding information is known.
411 Length Required	The server refuses to accept the request without a defined content length.
412 Precondition Failed	One or more of the preconditions specified in the request header fields returned a false value when tested by the server.
413 Request Entity Too Large	The server refuses to process a request because the requested entity is larger than the server is willing or able to process.
414 Request URI Too Long	The server refuses to service the request because the requested URI is longer than the server is willing to interpret.
415 Unsupported Media Type	The server refuses to service the request because the requested entity is formatted using an unsupported format.
416 Requested Range Not Satisfiable	The request was not processed because it included an invalid range relative to the requested resource.
417 Expectation Failed	The request was not processed because the server could not satisfy the information given in the request message's Expect header field.

(continued)

46. I'm a little confused. I've set up Web pages before and this does not look anything like what I've done. What gives?

You're probably confusing HTTP with HTML, which stands for *Hypertext Markup Language*. Recall that HTTP is a TCP/IP protocol that defines how Web clients such as browsers request Web pages from servers and how servers transfer Web pages to clients. HTML, on the other hand, is a specially coded "presentation" language that defines, among others, how text and art appear on a page. When you create a Web page, you specify how

TABLE 3.14 HTTP Status Codes (continued)

5xx Codes: Server Error—Server fails to fulfill what appears to be a valid request	
500 Internal Server Error	The server had an unexpected condition that prevented it from fulfilling the request.
501 Not Implemented	The server does not support the functionality required to fulfill the request.
502 Bad Gateway	The server, while acting as a gateway or proxy, received an invalid response from the upstream server it accessed in attempting to fulfill the request.
503 Service Unavailable	The server presently cannot process the request; this is presumed to be a temporary condition caused by any number of factors including a system overload or scheduled maintenance.
504 Gateway Timeout	The server, while acting as a gateway or proxy, did not receive a timely response from an upstream or auxiliary server it needed to access to fulfill the request.
505 HTTP Version Not Supported	The server does not support, or refuses to support, the HTTP protocol version specified in the request message.

specific information you want to provide should "look." This has nothing to do with how clients and servers request and receive resources.

47. I get it! I forgot that there is a difference between what I see and what is happening behind the scenes. Is the HTTP response header similar to the request header?

Yes. That's correct. The general format is

```
Response        = Status-Line
                  (( general-header
                     response-header
                     entity-header )   CRLF
                  CRLF
                  [ message-body ]
```

The Status-Line has the following format:

Status-Line = *HTTP-version* <space> *Status-Code* <space> *Reason-Phrase* CRLF

The *Status-Code* is a 3-digit integer result code that indicates the final outcome of a request. These codes and their interpretations are summarized in Table 3.14. The *Reason-Phrase* is the interpretation column of Table 3.14 and intended for the human user. Clients are not required to examine or display these phrases. The general-header and entity-header are similar to those found in a client request message (Tables 3.12 and 3.13), and the response-header part is based on the information contained in Table 3.11). Finally, the message-body component contains the requested resource. Once again, note the similarity in format to an e-mail message. There is a header block, followed by a blank line, followed by the message-body. In the case of a Web server response message, the message-body is the actual data requested by the Web client.

48. I've seen some of these codes before. For example, I sometimes see 404.

Ah. The old "dead link" response. Yes. This is always returned whenever the specified resource that is being requested cannot be located. It is the default status code when the server does not want to reveal why a particular request was refused or when no other response is available.

49. In all of this discussion about HTTP thus far, you never mentioned URL but instead use the term "resource." Why is this?

When used in the context of the Web, a resource generically refers to any Internet-accessible object. Examples include text files, graphical images, photographs, audio files, e-mail messages, newsgroup articles, and video clips. In short, anything that can be represented digitally and stored on an Internet-connected device is considered a resource. Specific resources are identified throughout the Web using a uniform resource identifier (URI), which contains the name, location, or any other defining attribute of the specified resource.

50. Wait a minute here. Don't you mean URL and not URI?

No. Most readers are definitely familiar with a *universal resource locator* (URL). However, this is not the same as a uniform resource identifier or URI, which conceptually represents a universal set of names and addresses of all resources regardless of their location. The URI concept is important because it allows all object names, regardless of their locations or attributes, to be treated the same.

51. So what is a URL then?

A URL is a specific example of a URI. URLs are used to identify a resource's location by specifying the access method needed to acquire the resource. The general form of a URL consists of two parts delimited by a colon; that is:

scheme:scheme-specific data

The first part, called the *scheme*, consists of a specific protocol that is used to access the resource; the second part is a function of the particular scheme (i.e., protocol) selected. As an example, consider the HTTP URL, which uses the HTTP protocol to identify a resource's location. The general form of this URL is

http://host:port/url-path

where *host* is the fully qualified domain name of an Internet-connected node, *port* is the port number to which the TCP connection is made (the HTTP well-known port number is 80), and *url-path* provides the details of how the specified resource is accessed relative to the scheme being used. Thus, both the syntax and interpretation of *url-path* are a function of the scheme. Note that in the general form of the HTTP URL, the scheme is the access method, namely, the HTTP protocol, and the scheme-specific data begin with the double slash (//). Thus, the given URL

http://www.fit.edu

informs us that the resource we seek is accessible via the HTTP protocol. Since this protocol is the foundation of the Web, we know that the resource is located at a Web site whose address is designated by the *host* portion of the URL, namely, www.fit.edu. Moreover, since *port* is not given, we also know that the user agent makes a connection to the origin server on port 80. Finally, since no paths are listed, we know that the object we seek is a top-level resource, most likely the home page for the designated site.

52. What other protocols does the Web support?

RFC 1738, which provides the primary specifications for URLs, lists several different IP-based access protocol methods, including ftp, http, gopher, nntp, and telnet. Table 3.15 provides a description of these URL schemes. It should be noted that the general concept of URI continues to evolve and that new URLs are expected to be introduced. For example, RFC 2806 provides information about URLs for telephone calls and describes the syntax for the schemes of telephone, fax, and modem. Additionally, RFC 2732 provides the conventions used for formatting IPv6 addresses, which are discussed later in the chapter, within URLs.

An alternative to URLs is the *Common Names Resolution Protocol* (CNRP), approved by the Internet Engineering Task Force (IETF) in September, 2000. CNRP enables users to access Web sites and send e-mail using more human-friendly names instead of strings of characters, slashes, dashes, and dots. For additional information about CNRP, see http://www.ietf.org/html.charters/cnrp-charter.html.

53. I have two more questions relative to HTTP. First, you mentioned "origin server" a few times. What is this?

An *origin server* is simply any server that contains a given resource. Most HTTP communication requests apply to a specific resource that resides on an origin server.

54. Okay. Lastly, how are HTTP connections actually established?

There are two primary ways in which HTTP connections can be established. The first and simplest type involves a single connection between the client and server, enabling the user agent (such as a browser) and server to communicate directly via an established virtual circuit. In this scenario, the user agent transmits its request messages directly to the server, and the server transmits its response messages directly to the user agent. Given that no intermediary connections exist, it is presumed that the requested resource also resides on the server. Thus, the server in this example is also the origin server since it contains the requested resource.

The second type of connection involves the presence of intermediate devices in the request–response chain. HTTP defines three types of intermediate devices: proxy, gateway, and tunnel.

- A *proxy* is an intermediate application program that acts as both a client and server; it is used as a proxy to the actual application. In the HTTP specifications, a proxy is thought of as a *forwarding agent*. It receives requests for a URI, rewrites all or part of the original request message, and then forwards this reformatted message to the server specified in the URI. A common use of a proxy is when a client is behind a firewall. In this scenario, a connection is established

TABLE 3.15 Universal Resource Locator (URL) Schemes Defined in RFC 1738

Scheme	Description	General Form and Syntax
ftp	The FTP URL. Designates files and directories that reside on Internet hosts and are accessible via the FTP protocol.	ftp://*user*:*password*@*host*:*port*/*url-path* • *user* is optional. If not specified, but one is requested by the FTP server, then "anonymous" is given. • *password* is optional. If not specified, but one is requested by FTP server, then user's e-mail address is given. • The *url-path* is optional. If given, it is interpreted as a series of FTP commands and has the following general form: *cwd1*/*cwd2*/. . ./*cwdN*/*name*;type=*typecode* — each *cwd* entity represents an argument to FTP's "change working directory" command — possible typecodes are a, i, and d — ;type=*typecode* may be omitted — *cwdx* and *name* may be empty • *port* defaults to 21 if omitted
http	The HTTP URL. Designates resources that reside on Internet hosts and are accessible by the HTTP protocol.	http://*host*:*port*/*path*?*searchpart* • No username or password is permitted • *path* is an HTTP selector • *searchpart* is a query string • *path* is optional and can be omitted • ?*searchpart* is optional and can be omitted • *port* defaults to 80 if omitted
gopher	The Gopher URL. Designates resources that reside on Internet hosts and are accessible by the Gopher protocol.	gopher://*host*:*port*/*gopher-path* • *gopher-path* is any of the following: *gophertype*
mailto	The mailto URL. Designates an Internet e-mail address. Does not directly access any specific resource.	mailto:*rfc822-addr-spec* • *rfc822-addr-spec* refers to the address specification defined in RFC 822.
news	The news URL. Designates USENET newsgroups or articles as specified in RFC 1036.	news:*newsgroup-name* news:*message-id* • *newsgroup-name* is the name of a newsgroup (e.g., alt.binaries.pictures); * is used to access all newsgroups • *message-id* is the ID number of a specific article

(continued)

TABLE 3.15 Universal Resource Locator (URL) Schemes Defined in RFC 1738 (continued)

Scheme	Description	General Form and Syntax
nntp	The NNTP URL. An alternative protocol for accessing USENET news.	nntp://*host*:*port*/*newsgroup-name*/*article-number* • *newsgroup-name* is the name of a newsgroup • *article-number* is the ID number of a specific article • *port* defaults to 119 if omitted
telnet	The TELNET URL. Designates an interactive service (not a resource) that is accessible by the TELNET protocol.	telnet://*user*:*password*@*host*:*port*/ • *port* defaults to 23 if omitted
wais	The wais URL. Designates WAIS databases, searches, or individual documents available from a WAIS database. (*Note*: The WAIS protocol is defined in RFC 1625.)	wais://*host*:*port*/*database* wais://*host*:*port*/*database*?*search* wais://*host*:*port*/*database*/*wtype*/*wpath* • The first form designates a searchable WAIS database • The second form designates a specific search • The third form designates a specific document to be retrieved from a WAIS database • *database* is the name of the database being queried • *wtype* identifies the object type • *wpath* specifies the WAIS document id • *port* defaults to 210 if omitted
file	The file URL. Designates a file that is accessible to a specific host and not universally accessible over the Internet.	file://*host*/*path* • *path* is a hierarchical directory path of the form *directory*/*directory*/. . ./*name*
prospero	The prospero URL. Designates resources that are accessed via the Prospero Directory Service.	prospero://*host*:*port*/*hsoname*;*field*=*value* • *hsoname* is host-specific object name in the Prospero protocol • *port* defaults to 1525 if omitted

between the client and proxy and the proxy and origin server. Now, client requests and corresponding server responses are submitted to the proxy, which in turn passes them to the server and client, respectively.

- A *gateway* is a server that acts as an intermediary to another server. A gateway operates differently than a proxy because a gateway receives client requests as if it were the origin server. A common application of a gateway is when an access protocol other than http is used. For example, when a user submits a Web request to retrieve an updated version of its Web browser software, the requested resource generally is located on an FTP server. Thus, transparent to the user, the initial request is made to a gateway server, which contacts the appropriate FTP server to acquire the requested resource. The gateway server then translates this resource back into HTTP form for delivery to the client.

- A *tunnel* is an intermediary program that acts as a relay point between two connections. Unlike a proxy or gateway, a tunnel does not operate on HTTP requests or responses. A tunnel is used when a client/server connection needs to pass through an intermediary. A common example is when a secure connection is needed for a specific HTTP transaction. In this scenario, the client and server first establish an authenticated connection to a tunnel, which is then maintained throughout the HTTP session. The tunnel process terminates when both ends of the relayed connections are closed. Tunneling is commonly used in virtual private networks (VPNs), which are discussed in Chapter 7, to encapsulate encrypted data into an IP packet so that data can be transported across the public Internet in a secure manner. Protocols designed to effect this include the Point-to-Point Tunneling Protocol (PPTP), the Layer 2 Tunneling Protocol (L2TP), and the Internet Security (IPSec) protocol. These protocols are discussed in Chapter 16.

55. I realize I said I only had two remaining questions, which you answered. However, I do have one more question. Actually, it's more of a request than a question. Could you conclude this discussion of HTTP with a description of how a Web page is transferred from server to client?

Certainly. Let's assume that we are browsing Florida Institute of Technology's Web site and find ourselves at the school's Office of Graduate Programs Web page, which is found at the URL http://www.fit.edu/AcadRes/graduate. Let's further assume that we are interested in acquiring information about graduate policies. When we place the browser's cursor over the Graduate Policies "icon," note that at the bottom of the browser the URL http://www.fit.edu/AcadRes/graduate/gradpol/index.html appears. This is the URL for the base HTML file that contains the information we seek. Following is a summary of what happens when we click on "Graduate Policies.

1. Your Web client program (i.e., your browser) initiates a network connection to the Web server "www.fit.edu." We will discuss how such a connection is actually initiated later in the chapter. For now, though, just realize that your client must first establish a network connection to the target Web server before any user data are transferred between the two nodes. This is because HTTP uses TCP as its transport protocol (OSI layer 4) and TCP is a connection-oriented protocol. If you're quick enough (or if the network is slow enough), you will see at the bottom of your browser the message "Connecting to ww.fit.edu."

2. When the connection is established, the client sends an HTTP *request message* similar to the example we gave in Question 45. The request message includes the Request-Line "GET http://www.fit.edu/AcadRes/graduate/gradpol/index.html HTTP/1.1," which requests that the specified resource be retrieved. Once again, if you are quick enough (or have a slow network connection), you will see at the bottom of your browser the message, "Sending request for AcadRes/graduate/gradpol/index.html."

3. The Web server www.fit.edu receives the request message and retrieves the requested resource ("gradpol/index.html") from its disk or RAM. The server then encapsulates this resource in an HTTP *response message* (see Question 47), and transmits the message to the client.

4. Your client receives the response message, recognizes that the encapsulated resource is an HTML file, and extracts it from the response message. The client then reads this file, interprets its contents, and displays the result of this interpretation to you.
5. The server and client eventually terminate the TCP connection.

Hopefully this illustration clears up some things for you.

56. Yes it does. One last (and I mean it this time) question. Suppose after I receive this resource, I then want to acquire another resource from the same server. Will my client have to re-establish a new TCP connection?

It depends. When HTTP was initially designed, separate TCP connections had to be established to acquire each URL. This was extremely inefficient, resulting in increased server loads and overall network congestion. The current implementation of HTTP now uses the concept of *persistent connections* as its default. With persistent HTTP connections, fewer TCP connections need to be opened and closed. Thus, in our illustration, if a *nonpersistent connection* were established, the server would close its side of the TCP connection immediately after it transmitted its response message and the client would close its side of the connection after it received the message. This implies that if the client requested a new resource from this same server, a new TCP connection would have to be established. With a persistent connection, though, the initial connection remains intact until a predetermined timeout interval. This enables the same client and server to submit and receive subsequent requests and responses across the same connection.

Persistent connections also can be implemented with or without a procedure called *pipelining*, which permits clients to make multiple requests without waiting for each response. Without pipelining, a client cannot make a new request until the previous one is finished (i.e., received). With pipelining, though, a client can issue successive requests, which are operated on concurrently, so that a new request can be made before the previous one is received. The default implementation of HTTP 1.1 uses persistent connections with pipelining. This reduces overall network congestion and latency, and new HTTP features or versions can be implemented more easily.

57. As promised, no more questions about HTTP or application-layer protocols. Let's now discuss the transport layer.

Okay, but it will be most helpful if you first go back to Chapter 2 and review Figures 2.15 and 2.16. As can be seen from these figures, the TCP/IP protocol suite defines two different transport layer protocols: the *User Datagram Protocol* (UDP) and the *Transport Control Protocol* (TCP). Additional information about UDP and TCP can also be found in the RFCs at http://www.rfceditor.org.

58. I went back to Chapter 2 and looked at Figures 2.15 and 2.16. I am little confused about TCP versus UDP and don't really understand when each one is used. How do you decide whether to use TCP or UDP?

You don't. This determination is made by the application program. For example, in Figure 2.16, examples of programs that use TCP include FTP, TELNET, FINGER, SMTP, and POP. Similarly, examples of programs that use UDP include SNMP, TFTP, NFS, DNS, and BOOTP. What is important to understand here is that TCP/IP has two different transport layer protocols—TCP and UDP. TCP is connection-oriented and provides for reliable network transmission. UDP is connectionless and provides unreliable network transmission. This implies that UDP cannot recover from lost packets. Thus, the application must detect lost data and retransmit them. UDP also has no ability to perform error or flow control. This makes UDP faster than TCP in performance when the network is not congested because it carries less overhead than TCP. However, when the network is congested, UDP-based applications will most likely result in session timeouts and poor performance.

59. Could you show me how these two protocols differ in their implementations?

Yes. Prior to doing so, though, we first extend the overview we initially introduced in Chapter 2 as part of our discussion on connectionless and connection-oriented services. It is repeated here with greater emphasis on the concepts of port numbers and data encapsulation among TCP/IP's layers.

Recall that to send a message from one node to another across the Internet, three different addresses are needed: the MAC sublayer address (also know as the *hardware address*), the network address (in this case, the Internet address), and the port address. The hardware address uniquely identifies a node on a network, the IP address specifies the network the node is connected to, and the port address uniquely identifies the specific application protocol or process (e.g., e-mail) that produced the data message. (Hardware and network addresses were discussed in Chapters 5 and 7, respectively. Internet addresses are also discussed in greater detail later in this chapter.) This leaves *port addresses*, which we briefly mentioned earlier in our discussion of some application-layer protocols. Port addresses, which are also known as *well-known port numbers* or *well-known services*, identify the specific process or application a user accesses on a host. Port numbers are two bytes long and are standardized according to the application protocol. For example, the two popular e-mail protocols, the Simple Mail Transfer Protocol (SMTP) and the Post Office Protocol, Version 3 (POP3), are assigned port numbers 25 and 110, respectively. Other examples include TELNET (port 23), domain name server (port 53), FINGER (port 79), and HTTP (port 80).

Well-known port numbers are controlled by the Internet Assigned Numbers Authority (IANA). Furthermore, RFC 1700, "Assigned Numbers," contains a complete listing of all well-known ports used in the Internet, and the /etc/services file on UNIX systems contains a list of well-known ports as well. Whenever possible, the same port assignments are used with both UDP and TCP. Standardized port numbers originally ranged from 0–255, but were expanded to 0–1023 in 1994. A sample of well-known port numbers is provided in Table 3.16. RFC 1700 also contains a listing of registered port numbers that are not controlled by IANA. These numbers, which are in the range of 1024–65535, are available for use by the general user.

TABLE 3.16 A Sample of Well-Known Port Numbers (Extracted from RFC 1700)

Decimal/Protocol	Keyword	Description
0/TCP		Reserved
0/UDP		Reserved
5/TCP	RJE	Remote Job Entry
5/UDP	RJE	Remote Job Entry
7/TCP	ECHO	Echo
7/UDP	ECHO	Echo
9/TCP	DISCARD	Discard
9/UDP	DISCARD	Discard
11/TCP	DAYTIME	Daytime
11/UDP	DAYTIME	Daytime
17/TCP	QOTD	Quote of the Day
17/UDP	QOTD	Quote of the Day
20/TCP	FTP-DATA	File Transfer (Default Data)
20/UDP	FTP-DATA	File Transfer (Default Data)
21/TCP	FTP	File Transfer (Control)
21/UDP	FTP	File Transfer (Control)
23/TCP	TELNET	Telnet
23/UDP	TELNET	Telnet
25/TCP	SMTP	Simple Mail Transfer
25/UDP	SMTP	Simple Mail Transfer
42/TCP	NAMESERVER	Host Name Server
42/UDP	NAMESERVER	Host Name Server
49/TCP	LOGIN	Login Host Protocol
49/UDP	LOGIN	Login Host Protocol
53/TCP	DOMAIN	Domain Name Server
53/UDP	DOMAIN	Domain Name Server
58/TCP	XNS-MAIL	XNS Mail
58/UDP	XNS-MAIL	XNS Mail

(continued)

Port numbers are contained within the UDP or TCP header, which is generated at TCP/IP's transport layer and corresponds to layer 4 of the OSI model. Thus, a data message is first created on the sending machine at TCP/IP's application layer (OSI layers 5–7) by a specific application protocol. This data message is passed to TCP/IP's transport layer (OSI layer 4) and encapsulated into either a UDP or TCP header, which includes the port address that identifies the application or process that created the message. (Note that the application determines whether UDP or TCP will be used.) The resulting UDP or TCP data structure is then passed to TCP/IP's Internet layer (OSI layer 3) and encapsulated into

TABLE 3.16 A Sample of Well-Known Port Numbers (Extracted from RFC 1700) (continued)

Decimal/Protocol	Keyword	Description
67/TCP	BOOTPS	Bootstrap Protocol Server
67/UDP	BOOTPS	Bootstrap Protocol Server
68/TCP	BOOTPC	Bootstrap Protocol Client
68/UDP	BOOTPC	Bootstrap Protocol Client
69/TCP	TFTP	Trivial File Transfer
69/UDP	TFTP	Trivial File Transfer
79/TCP	FINGER	Finger
79/UDP	FINGER	Finger
80/TCP	WWW-HTTP	World Wide Web HTTP
80/UDP	WWW-HTTP	World Wide Web HTTP
88/TCP	KERBEROS	Kerberos
88/UDP	KERBEROS	Kerberos
110/TCP	POP3	Post Office Protocol—Version 3
110/UDP	POP3	Post Office Protocol—Version 3
119/TCP	NNTP	Network News Transfer Protocol
119/UDP	NNTP	Network News Transfer Protocol
531/TCP	CONFERENCE	Chat
531/UDP	CONFERENCE	Chat

an Internet Protocol (IP) header, which contains the network address that identifies the network and host running the specified application or process. Finally, the IP datagram is passed to TCP/IP's network interface layer (OSI layers 1–2), encapsulated into a data frame that contains the hardware address of the destination machine on the remote network, and then placed on the medium for transmission. A summary of this process is shown in Figure 3.13. UDP's and TCP's role in this transfer process is a function of the type of transport service being provided. UDP provides connectionless service; TCP provides connection-oriented service.

60. Start with UDP. How does it operate?

As stated above, UDP is a connectionless protocol that provides an unreliable datagram service. It does not furnish any end-to-end error detection or correction, it does not retransmit any data it did not receive, and it has no ability to perform error or flow control. This implies that all UDP-based application programs must bear the onus of providing a mechanism for error and flow control and for recovering from lost packets. In short, all data reliability and integrity issues fall to the application programs that use UDP. This makes UDP faster than TCP in performance when the network is not congested because it carries less overhead than TCP. However, when the network is congested, UDP-based applications will most likely result in session time outs and poor performance. Application protocols based on UDP include the Trivial File Transfer Protocol (TFTP), Network File System (NFS), the Simple Network Management Protocol (SNMP), the Bootstrap Proto-

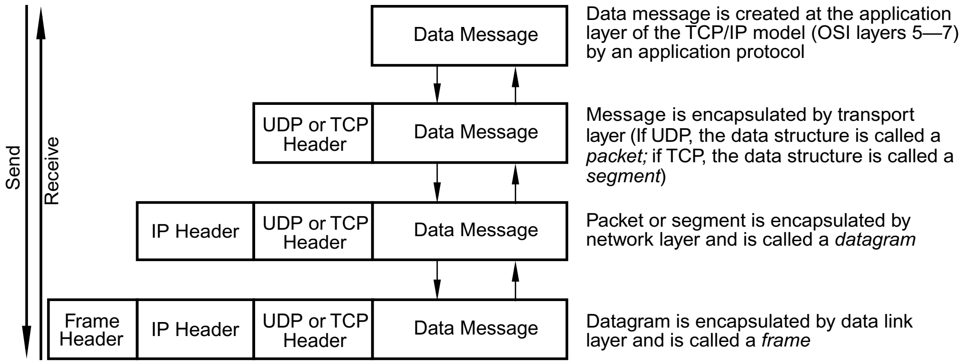


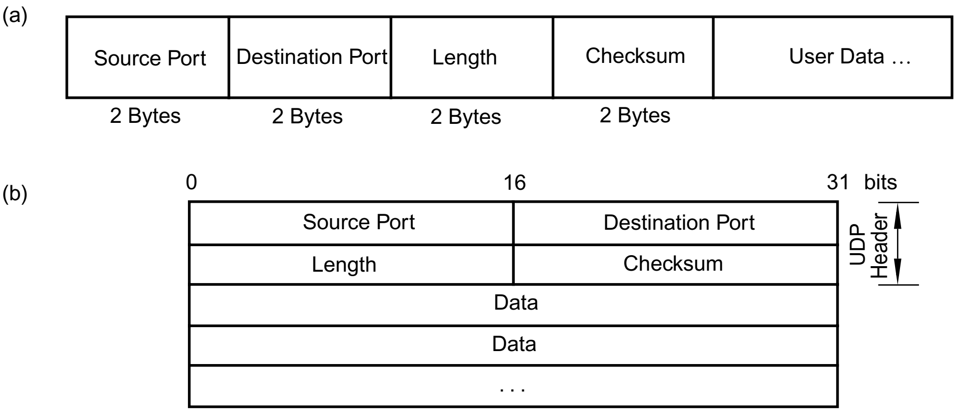
FIGURE 3.13 A conceptual view of data encapsulation. Each layer encapsulates (or deencapsulates) the data structure from the previous layer into its own independent data structure. The four layers shown here correspond to TCP/IP's four layers as shown in Figure 2.16. A data message is first created on the sending machine at TCP/IP's application layer. It is then passed to the transport layer where it is encapsulated with either a UDP or TCP header. If the application protocol is connectionless, then a UDP header is prepended and the new data structure is called a packet. If the application is connection-oriented, then a TCP header is prepended and the new data structure is called a segment. This transport layer header includes the source and destination port numbers corresponding to the application protocol. The transport layer data structure is then passed to the network layer where an IP header is prepended. The IP header includes the source and destination network addresses. This new data structure is now called a datagram. Finally, the datagram is passed to TCP/IP's network interface layer where a frame header is prepended. Included with this header are the source and destination hardware addresses. The resulting data frame is then passed to the physical layer for transmission. On the receiving machine, each layer's header is stripped off as the data structure works its way up the layers.

col (BOOTP), and Domain Name Service (DNS). Figure 3.14 contains a diagram of the UDP header format.

Note from Figure 3.14 that the UDP source and destination ports refer to a well-known port number. As we discussed earlier, these well-known port numbers correspond to a specific application. Port numbers are important because they enable the sending node to identify on which port the destination machine is running the application program that created the data message. For example, if a user on a sending node uses TFTP to transfer a file from a destination node, the sending machine "knows" that the destination machine will be running the TFTP application process on port 69. Further note that UDP (as does TCP) uses IP to send and receive datagrams, as shown in Figure 3.13. This is why UDP (and TCP) is layered above IP. It ensures that a UDP-based message gets encapsulated into an IP datagram for delivery to a destination node across the Internet.

61. How does this differ with TCP?

The Transport Control Protocol is the TCP of TCP/IP. It is a full-duplex, connection-oriented protocol that performs several functions, including providing for reliable data



Source Port: 16 bits in length and contains the well-known port number corresponding to the application protocol that created the data message. This field is optional and is filled with 0s if not used.

Destination Port: 16 bits in length and contains the well-known port number corresponding to the application protocol that created the data message.

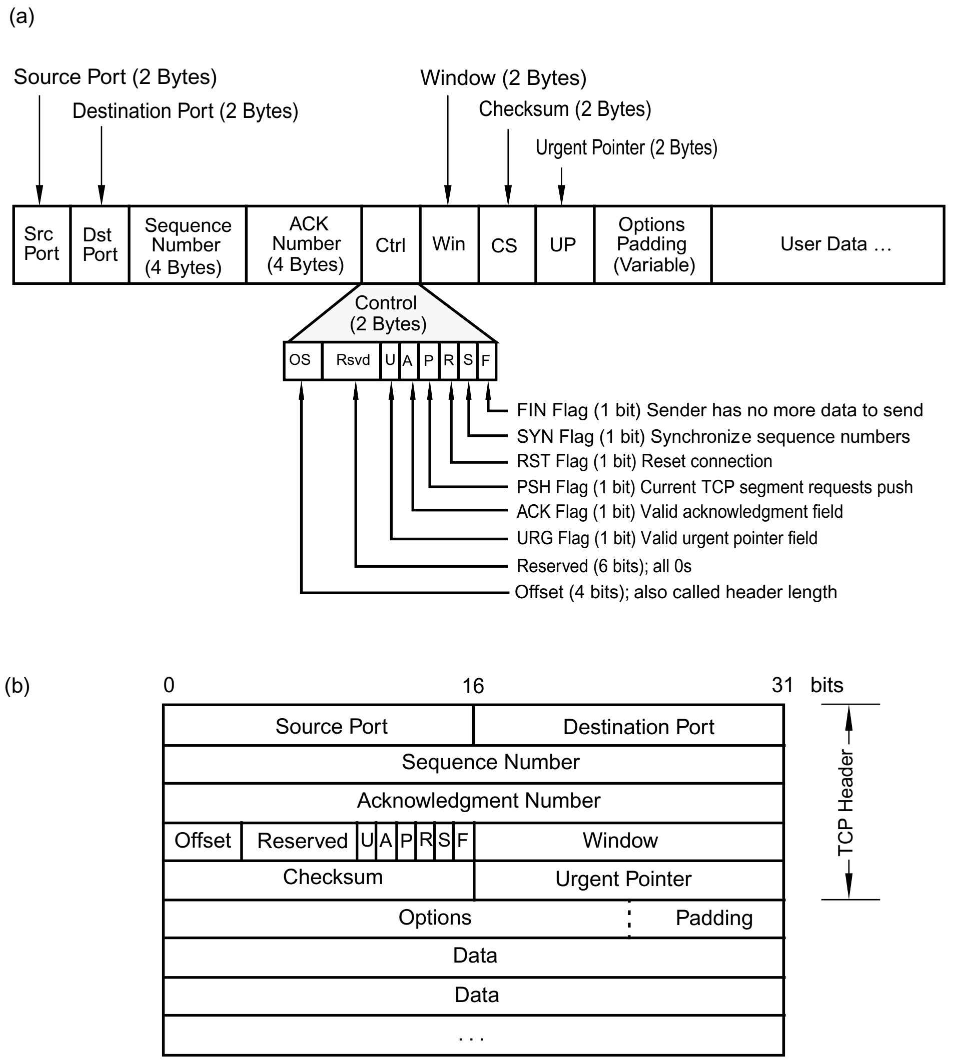
Length: This 16-bit field corresponds to the length of the UDP datagram. The minimum length of a UDP datagram is 8 bytes.

Checksum: This optional 16-bit field contains the checksum of the datagram. The calculation of the checksum includes the source and destination addresses and the protocol field from the IP header, which is shown in Figure 8.20. (This combination of UDP and IP header fields is called the UDP pseudoheader.) These fields are included in the checksum so that the datagram is delivered to the correct destination network and host. If the checksum field is not used, then it is filled with 0s.

User Data: This field contains actual user data created by an application protocol such as SMTP or POP3 for e-mail.

FIGURE 3.14 Format and contents of a UDP header. Part (b) is a block diagram frequently shown in the RFCs.

transmission by furnishing end-to-end error detection and correction, guaranteeing that data are transferred across a network accurately and in the proper sequence, retransmitting any data not received by the destination node, and guaranteeing against data duplication between sending and receiving nodes. Application protocols that generally use TCP include TELNET, File Transfer Protocol (FTP), Simple Mail Transport Protocol (SMTP), and Post Office Protocol (POP). The format and contents of the TCP header are shown in Figure 3.15. You will note that when compared to UDP's header (see Figure 3.14), TCP's header contains similar information carried by UDP. For example, the first four bytes of the header specify the source and destination ports. Thus, like UDP, TCP uses port numbers to identify and deliver data to the correct application.



IGURE 3.15 Format and contents of a TCP header. Part (b) is a block diagram view frequently hown in the RFCs.

62. OK. But how does TCP guarantee reliable delivery?

TCP guarantees reliable host-to-host data transfers via the following fields:

Sequence Number: This field carries the sequence number assigned to the first byte of data being transmitted. It is used so that the TCP process at the receiving host can keep the data segments it receives in the correct order. When a connection is first established between sending and receiving hosts, the SYN bit in the two-byte control field is set to inform the receiving host that the number given in the sequence number field is the initial sequence number for synchronization purposes. The next byte of data sent will then increment this number by 1. Generally, the initial sequence number is 0, and the first data byte is 1. Sequence numbers are needed because TCP was not designed to deliver data as independent packets. Instead, TCP delivers data as a continuous byte stream. Thus, sequence numbers keep track of the order in which byte streams are sent and received.

Acknowledgment Number: This field is used to verify the receipt of data. Acknowledgments are only provided by the receiving host when it receives a data segment with a valid checksum. This strategy is known as positive acknowledgment with retransmission (PAR). Thus, if the sender does not receive a positive acknowledgment from the receiver within a certain time frame, then it will assume the data segment did not arrive, or it arrived damaged, and retransmit the data. The number given in the acknowledgment number field by the receiving host is the sequence number of the next byte of data the host expects to receive. When the acknowledgment number field is used, the ACK bit in the two-byte control field is also set.

Control: This two-byte field provides control information.

—The four-bit *offset* specifies the length, measured in 32-bit multiples, of the current TCP segment header. This field is needed because TCP headers are of variable length (although they must be at least 20 bytes in length).

—The six-bit *reserved* field is set to all 0s and is reserved for future use.

—The six one-bit flags are used to specify the type of segment being transferred. For example, when set:

- The *URG* flag indicates that the segment contains urgent data and should be processed by the receiver as soon as possible. (See the urgent pointer field description that follows.)
- The *ACK* flag indicates that the segment contains data in the acknowledgment number field as discussed earlier.
- The *PSH* flag indicates that the current segment being "built" contains data that must be delivered immediately. A "push" event is initiated by an upper-layer process and is interpreted as, "Transfer all queued data to receiver immediately."
- The *RST* flag is used when an event arises that causes an ungraceful disconnect. In such instances, a host sends a TCP segment with the RST bit. The receiving host interprets this segment as an immediate termination alert and responds by aborting the segment. TCP will also inform the corresponding application of this event.
- The *SYN* flag indicates that the segment contains data in the sequence number field as discussed earlier.
- The *FIN* flag is used to terminate a TCP connection. When the host that initiates a termination segment receives a positive acknowledgment, the TCP connection is closed.

Window: The two-byte window specifies the maximum number of bytes the receiving host is capable of accepting. In short, it provides end-to-end flow control; the window field enables the receiver to control the flow of bytes from the sender. Thus, the amount of data the sender transmits can never exceed the window size established by the receiver. The receiver can change the window size anytime during an established connection as additional buffer space becomes available. A window size of 0 informs the sender to stop transmission until it receives a non-0 window size. Since TCP supports full-duplex, both sender and receiver may provide flow-control information via the window field.

Checksum: This two-byte field is used to verify that the transmitted segment is valid. TCP uses a pseudoheader similar to UDP to provide error control on the IP header, the TCP header, and user data. Like UDP, TCP prepends the pseudoheader to the segment, and the checksum calculation is performed over the entire data structure (pseudoheader plus TCP segment).

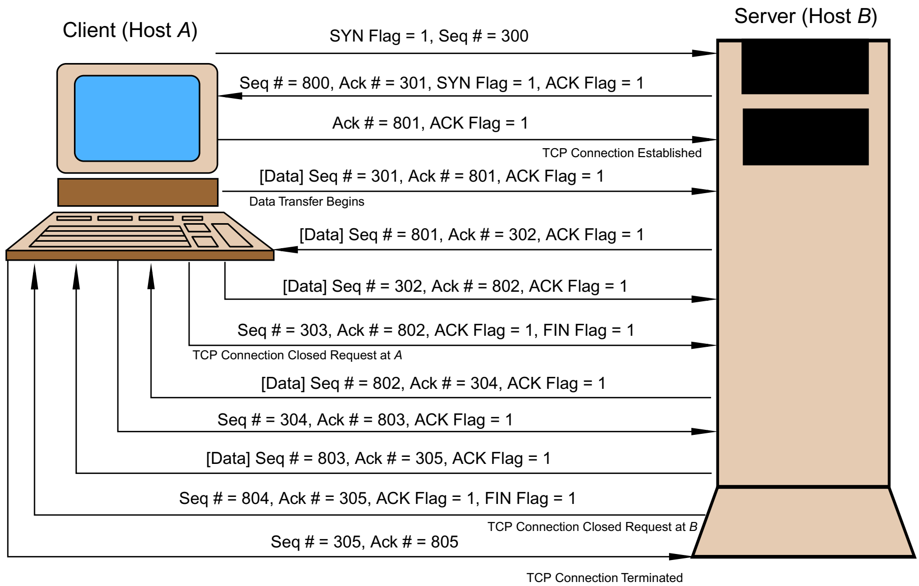
Urgent Pointer: This two-byte field is used in tandem with the one-bit URG flag as a signal to the receiving node that it needs to inform the corresponding application program that urgent data are being transmitted and need to be processed. An example of an urgent data instance is a hung TELNET session. To abort this session, the user on the local host would press the escape character, which is then transmitted to the remote host as urgent data. The urgent pointer field specifies where within the segment the urgent data end and the regular (i.e., nonurgent) data begin.

Options and Padding: This variable-byte field specifies any options a particular TCP process requires. A common option used is maximum segment size, which is used to restrict the size of a TCP segment. Padding is then used to fill the remaining bits of the field with 0s so that it ends on a 32-bit boundary.

63. How is a TCP connection established?

TCP is a connection-oriented protocol, which implies that a logical connection must first be established (i.e., "nailed up") between two nodes before any data are transferred. (Recall from Chapter 2 that this logical connection is called a virtual circuit.) To do this, TCP uses a process known as a *three-way handshake*, which involves an initiating host (i.e., a client) requesting that a link be established between it and a server. (Recall our earlier HTTP illustration of how a Web page gets transferred between a client and server.) In its simplest form, this handshake consists of two nodes exchanging three TCP segments. This is illustrated in the first three steps of Figure 3.16 and can be described in the following general terms:

- The client (node *A* in Figure 3.16) generates and transmits a TCP segment that has the one-bit synchronize (SYN) flag set to 1 and the four-byte sequence number field set to the initial sequence number (x) it intends to use for the current connection. The SYN bit, which is part of the TCP header's control field (see Figure 3.15), notifies the server that the client wants to establish a connection; the sequence number informs the server of the initial sequence number on which data segments transmitted by the client will be based. (Recall that sequence numbers enable a node to keep data segments in the correct order.)



A sends a synchronization segment to *B* indicating its desire to establish a connection and that its initial sequence number is 300. This means that the rst data segment *A* sends will be numbered 301.
B receives *A's* synchronization segment and sends *A* a synchronization and acknowledgment segment. Note that *B's* initial sequence number is 800, which means that *B's* rst data segment will be numbered 801.
A receives *B's* synchronization and acknowledgment segment and sends an acknowledgment segment. At this stage, a TCP connection is established between *A* and *B*.
A transmits data segment 301 to *B* and informs *B* that it is expecting segment number 801.
B receives *A's* segment and sends *A* data segment 801. This segment also acknowledges receipt of *A's* segment by informing *A* that it expects to receive segment 302.
A receives *B's* segment and sends *B* data segment 302, which also acknowledges receipt of *B's* segment.
A sends *B* a nish segment, which informs *B* that *A* is closing its side of the TCP connection.
B receives *A's* last two segments and sends *A* data segment 802. Note that this segment also acknowledges receipt of *A's* previous transmissions by setting the acknowledgment number to 304. At this stage, *A* cannot transmit any new data segments but continues to transmit acknowledgment segments.
A acknowledges *B's* latest transmission.
B receives *A's* acknowledgment segment and sends *A* data segment 803.
B sends *A* a nish segment, which informs *A* that *B* is closing its side of the TCP connection.
A receives *B's* latest transmissions and acknowledges them. At this stage, the link is terminated since neither *A* nor *B* has any more data to transmit.

IGURE 3.16 Pictorial description of a TCP connection. TCP's three-way handshaking process, hich is used to establish a TCP connection, is illustrated in steps 1–3, and link termination is illustrated in steps 7–12.

- Upon receipt of this first segment, the server records the sequence number and then responds to the client by generating and transmitting an acknowledgment segment, which has the SYN and acknowledgment (ACK) bits of the control field set to 1. Additionally, this segment's four-byte ACK number field is set to one more than the client's initial sequence number ($x + 1$), and the four-byte sequence number field is set to the server's initial sequence number (y). Note that the one-bit SYN flag = 1 because the synchronization process (i.e., the handshaking) is still ongoing and that the one-bit ACK flag = 1 because the ACK number field contains data. Further note that the ACK number ($x + 1$) informs the client that the server expects the next segment to be numbered ($x + 1$) and that the server's initial sequence number (y) informs the client of the initial sequence number on which data segments transmitted by the server will be based.
- Lastly, the client acknowledges receipt of the server's segment, and at this stage, a link is established.

64. Can you illustrate this procedure?

Yes. Consider the first three steps shown in Figure 3.16. In this example, host *A* initiates a TCP connection to host *B* by sending a synchronization segment with an initial sequence number of 300. Host *B* responds to this segment by sending a synchronization/acknowledgment segment. The acknowledgment number it uses is 301, which confirms receipt of *A*'s segment and indicates that *B* expects the next segment it receives from *A* to be numbered 301. This segment also contains *B*'s initial sequence number, 800. *A* now responds to *B* by sending an acknowledgment segment using an acknowledgment number of 801, which confirms receipt of *B*'s segment by indicating that *A* expects the next segment it receives from *B* to be numbered 801. At this stage, the TCP connection is established and data segments can now be exchanged between the two hosts. From this illustration, note the implicit understanding acknowledgment numbers provide the two nodes in their exchange of segments: By sending an acknowledgment number of $n + 1$, a node is not only informing its mate at the other end of the connection that it expects the next segment to be numbered $n + 1$, but it is also stating implicitly that it has successfully received all previous segments numbered n or less. This is the heart of TCP's reliability feature.

65. How come the client and server's initial sequence numbers are different?

The client and server's initial sequence numbers are administered locally and hence do not have to be the same.

66. When you illustrated HTTP earlier, you said that the TCP connection between the client and server is eventually closed. How is this accomplished?

A TCP connection is closed when a host has no more data to transmit. Since TCP is a full-duplex protocol, a TCP connection can be closed in one direction but remain open in the other direction. Thus, in Figure 3.16, host *A* can close its side of the connection but continue to receive data from *B*, host *B* can close its side of the connection but continue to receive data from *A*, or both *A* and *B* can close the connection simultaneously. For example, if HTTP were using a nonpersistent procedure, then immediately after the server transmitted a response message it would close its end of the connection, and after the client received the response message it would close its end of the connection. Regardless of which host initiates the close request, the procedure is the same and is similar to the three-way handshaking process used for establishing the connection except the one-bit finish (FIN) flag is set to 1 instead of the SYN flag.

67. How about an example?

OK. As an illustration of how a TCP connection is closed, examine steps 7–12 in Figure 3.16. At steps 6 and 7, the client transmits its last data segment (302) and then initiates a close request by sending a finish segment (303) that has FIN = 1. The server acknowledges these transmissions at step 8, and the client side of the connection is now terminated. Although *A* will no longer be sending *B* any more data segments, it still may continue receiving data from *B* and is capable of acknowledging receipt of these data segments. This is shown in steps 8–10. In step 11, *B* sends *A* a finish segment, which indicates that it no longer has any more data to transmit. Finally, in step 12, *A* acknowledges receipt of *B*'s finish segment, and the overall link is terminated. Note that the overall connection is not closed until the second FIN segment is acknowledged.

The entire process of establishing, transferring data between both ends of, and closing a TCP connection is readily apparent when using a TCP-based Web browser application such as Netscape. The user on the client machine (host *A* in Figure 3.16) requests data from a Web server (host *B*). The client application makes a call to the TCP software to establish a link to the server (steps 1–3 in Figure 3.16). Once the link is established, the client submits its request(s) and the server responds accordingly (steps 4–6). When the client and server have no more data to transfer, the application program informs the TCP software of this state and directs the link to be terminated (steps 7–12).

68. Could you expand on TCP's reliability aspect?

Sure. TCP's reliability feature is a function of its sequence and acknowledgment numbers. This was discussed earlier and illustrated in Figure 3.16. Each time a host receives a valid TCP segment, it sends an acknowledgment as part of each outgoing segment. If, on the other hand, a valid segment was not received, then no acknowledgment is sent. Thus, if the transmitting node fails to receive an acknowledgment for a previously transmitted segment within a specified time period, it will retransmit the segment. This is why TCP is regarded as a *positive acknowledgment with retransmission* (PAR) protocol. The use of sequence numbers also enables the receiving host to reassemble data segments in the correct order in the event that they were to arrive out of sequence.

69. What about flow control issues? How does TCP address this?

In addition to reliability, TCP also provides end-to-end flow control. This is done using the two-byte window field (see Figure 3.15). At each end of a link, TCP maintains send and receive buffers. The send buffer contains two sets of data: data waiting to be transmitted and data that have been transmitted but not yet acknowledged. This second data set represents data generated by the application program and ready for transmission. The receive buffer also contains two data sets: correctly ordered data that have been received but not yet processed by the application and data that arrived out of sequence, which require TCP to reorder in the correct sequence. Using the window field, a receiver advertises a window size that is less than or equal to the size of its available buffer space. Thus, on the receive side, the available buffer space must take into account the amount of data currently occupying the buffer at any given time.

On the send side, two things must be done. First, the sending entity must calculate a window size that will limit the amount of data it can send. This calculation, which is a function of the receiver's advertised window size and the difference between the amount of data sent and the amount of data acknowledged, is necessary because without it a sender can overrun a receiver's buffer space. For example, assume the receiver's advertised window is 1000 bytes. If the sender, at this moment in time, has currently transmitted 700 bytes of which 500 bytes have been acknowledged, then the sender's window size is 1000 – (700 – 500) = 800 bytes. This means that at this particular moment, the sender can continue sending segments as long as the total number of bytes it sends does not exceed 800 bytes of data. In addition to computing this value, the sending node must also ensure that its own buffer space does not get overrun by the application process writing data that are ready for transmission. Thus, the difference between the amount of data waiting to be transmitted and the amount of transmitted data that has been acknowledged must be less than or equal to the maximum size of the send buffer.

Note that if a sending node receives an advertised window size of 0, then it is prohibited from transmitting any more data. This will eventually lead to a full send buffer because the application process will continue to generate data ready for transmission. At this stage, TCP will block the application process, which will effectively stop the application from generating any more data. Only until a non-0 window size is advertised will the sender be able to resume data transmissions.

70. How can this happen if no more data are transmitted?

TCP has a built-in mechanism that permits the sending node to probe the receiver periodically with one-byte segments if it receives a 0-sized window. One of these segments will eventually evoke from the receiver a response that contains a non-0-sized window.

71. Is there anything else I should know about TCP?

That is hard for us to say because we are unaware of what level of knowledge you need about TCP. Nevertheless, we encourage you to read the RFCs related to TCP for additional information. Box 3.1 also provides supplementary information that might be useful.

72. OK. Let's focus on TCP/IP's network layer. What else can you tell me about it besides what was already presented in Chapter 2?

The network layer transfers user messages from a source host to a destination host. In TCP/IP, the heart and soul of this layer is the *Internet Protocol*, which is the IP of TCP/IP. IP receives data bits from the lower layer, assembles the bits into packets (called IP datagrams), and selects the "best" route based on some metric to route the packets between nodes. (*Note:* A metric is a description of the "cost" of a route used by routing hardware and software to select the best possible route. See Chapter 7 for more information.)

BOX 3.1: TCP Enhancements

Although TCP continues to function admirably with little change since its development (which is nearing 30 years), this does not mean that it does not require any modifications. TCP is still inadequate in certain situations. Fortunately, TCP was designed so that enhancements to the protocol can be made without major ramifications.

One of TCP's inadequacies is its window size. As we discussed earlier, TCP's window field specifies the maximum number of bytes a sender can transmit without stopping and waiting for an acknowledgment from the receiver. The original protocol was designed with a 16-bit window field, which implies that the maximum buffer size a receiver can advertise is $2^{16} - 1 = 65{,}535$ bytes. Note that this 16-bit restriction is not an issue when TCP is implemented on today's high-speed switched Ethernet LANs because latency is low, and hence, data are transmitted, received, and acknowledged with little delay. This is not the case, however, for WAN-based TCP implementations. Compared to LANs, WANs typically exhibit high latency. As a result, a 16-bit window size can adversely restrict data flow between two nodes. This restriction was removed by the introduction of the TCP option, *window scale*, which was initially defined in RFC 1072 and then later redefined in RFC 1323. The window scale option takes advantage of TCP's options field (see Figure 3.15). It enables a receiver to advertise a window size of up to 30 bits by specifying a 14-bit value in the options field. This 14-bit value specifies the number of bits the original window field should be "moved to the left." For example, if the original window field contains sixteen 1-bits, and the window scale option consists of the 14-bit field 00000000011111, then the receiver's maximum window size is interpreted as $2^{16+5} - 1 = 2^{21} - 1 = 2{,}097{,}151$ bytes. This option, which enables a receiver to increase its maximum buffer size up to one gigabyte, is negotiated as part of the three-way handshaking process. Current implementations of this option include Microsoft's Windows 98/2000 and Sun Microsystems' Solaris 7 and Solaris 8 operating systems.

A second TCP embellishment focuses on its time-stamp function. As noted earlier, TCP's PAR requires a sender to retransmit all unacknowledged data segments if it does not receive an acknowledgment from the receiver within a specified time frame. Once again, the LAN/WAN difference becomes apparent. On LANs, this time threshold can be short. On WANs, however, a short time threshold can result in an "early" time expiration; that is, the sender is given a "time's up" alert and retransmits a segment before the original segment actually arrives at its destination. Standardized algorithms for establishing appropriate time stamps are now specified in RFC 1323.

A third enhancement to the TCP protocol improves its use of sequence numbers. Recall from our earlier discussion that TCP identifies all segments by sequentially numbering them and that acknowledgments carry the sequence number of the segment the receiver next expects to receive. So if a receiver receives segments 23–89, it will send an acknowledgment with sequence number 90, which implies that it has correctly received all segments up to and including segment 89. The inadequacy in this scheme is the sequential nature in which segments are acknowledged. For example, assume a receiver receives segments 23–89 and 112–200. It must submit an acknowledgment with a sequence identifier of 90 since it did not receive segments 90–111. The original version of the protocol does

(continued)

BOX 3.1: TCP Enhancements (continued)

not accommodate for segment "gaps." This inadequacy is resolved by the *selective acknowledgment* option, which is specified in RFC 2018. If a gap exists in the receiver's buffers, it sends an acknowledgment that identifies the last cumulative data byte it received (e.g., 89) and then uses TCP's options field to specify all additional data segments it received after the missing segments (e.g., 112–200). The selective acknowledgment option is negotiated during the three-way handshaking process via the selective acknowledgments permitted function, which is also defined in RFC 2018.

IP is connectionless, which implies that every datagram must contain the address of the destination node. This address, called an Internet or IP address, is assigned to a node's network interface as part of the node's initial network configuration. An IP address uniquely identifies a host similar to the way a social security number uniquely identifies a person. It is used by the network layer as a road map to locate a host within the Internet by determining what path a datagram is to follow en route to its final destination. Datagrams destined for a host connected to the same local network as the sending host are delivered directly by the sending host. To transfer datagrams destined for a host connected to a remote network, however, IP relies on routers or switches (see Chapter 7), which connect two (or more) dissimilar networks to each other. In the context of the network layer, routers and switches are frequently referred to as gateways. Thus, an IP gateway routes packets between the networks to which it is connected. (An illustration of this concept is given in Figure 7.4.) As datagrams are routed through the Internet, each intermediate gateway maintains a routing table that contains entries of the location of the next gateway a datagram should be transferred to based on the destination address of the datagram. A UNIX and Microsoft NT command line utility that displays IP routing information for a local host is *netstat*. An example of this command's output is shown in Table 7.4.

When a datagram passes through an intermediate gateway en route to another network, it is called a hop. IP routing is usually accomplished via a simple hop-by-hop algorithm. If a packet doesn't incur a route through a router, then it hasn't incurred a hop. If, on the other hand, a packet transverses through two gateways in reaching its final destination, then we say the destination is two hops away. For example, in Figure 7.5, packets from H1 to H2 that follow the R1-R2 path require two hops since packets pass through two routers en route from source to destination. However, packets from H1 to H2 that follow the R1-R4-R5-R3-R2 path, require five hops. A UNIX program that depicts the gateways a packet transverses is called *traceroute*. (The corresponding Microsoft NT command is called *tracert*.) The output of a specific trace made from one web server to another is shown in Figure 7.6.

An important concept in routing a packet is that of "cost." Each link to another gateway has a pseudo-cost assigned to it by the network manager of the link. This pseudo-cost is used to compute the maximum cost allowed by a system to reach another system. Most routing algorithms used for IP routing use a least-cost/least-hops methodology. This means that the primary path a datagram takes is determined by the least-cost to transfer the packet from source to destination. If something on that path fails, then the next least-cost is used. If there are two or more paths with the same cost, then the least-hops is computed and the shortest hop path with the least-cost is taken. A side effect of this type of routing is used by the routing algorithm to determine that a system is not reachable. What happens in a gateway is the destination node's cost goes to infinity. Thus, by definition, the node is simply too expensive to reach and hence is unavailable to the network.

Although gateways are used to deliver packets from one network to another, IP does not guarantee that a packet will indeed be delivered to its destination. If an intermediate gateway, for example, contains incorrect or stale routing information, packets might get lost. IP does not take any action to retransmit undelivered packets. This is done by higher-level protocols, specifically TCP. Additionally, IP fragments and reassembles datagrams when necessary so they do not exceed the maximum packet length (called the maximum transmission unit, or MTU) a physical network is capable of supporting. If a packet's size is greater than a network's MTU, the packet is broken into smaller units (called fragmenting) and sent to the destination in the form of several separate datagrams. The complete packet is then reassembled at the destination node before it is passed to the higher levels. Reassembly of an IP datagram can only occur at the destination node and not at any of the intermediary nodes the datagrams transverses. This is because IP is a connectionless datagram service—datagrams can take different paths en route to their destination and hence an intermediary node might not receive all of the fragmented datagrams.

73. OK. In Figure 2.15, I see that IP includes something called ICMP. What is this?

An integral part of IP is the *Internet Control Message Protocol*, ICMP, which uses an IP datagram to carry messages about the communications environment of the Internet. Although ICMP is layered above IP, it is generally discussed and shown with IP (as it is in Figure 2.15) because of its relationship to IP. ICMP allows interconnected nodes to exchange messages to report flow-control problems, to report that a destination node is unreachable, to notify a host to use a different gateway to route packets, and to test the status of a link to a remote host. For example, if a gateway receives a datagram destined for a host that is unreachable, the gateway will send an ICMP "host unreachable" message to the originator. This message is triggered when the local router sends an *Address Resolution Protocol* (ARP) request to the target node requesting the node's MAC sublayer address. On an Ethernet network, the MAC address is the node's Ethernet address. (See Chapter 5.) The ARP request is a network broadcast that announces the target node's IP address and requests the node to return its MAC address. If the node does not reply within a specified period of time, then a "host unreachable" message is sent by the router to the source node. A "host unreachable" message signifies that the target host is not connected to the local network, the target host is a valid local node but is currently offline, or the local network is congested and the router's ARP request is timing out before it reaches the target host.

Another example of an ICMP message is "network unreachable." This message is sent by a router when it cannot reach the target network. This can be due to a downed link (e.g., a cable cut or disconnected port), an incorrectly configured network address mask (discussed later), or an incorrectly entered network address. An example of an ICMP-based application is the UNIX and Microsoft NT command *ping*, which allows one node to test

```
gallo@bb> ping -s zeno.fit.edu
PING zeno.fit.edu: 56 data bytes
64 bytes from zeno.fit.edu (163.118.5.4): icmp_seq=0. time=128. ms
64 bytes from zeno.fit.edu (163.118.5.4): icmp_seq=1. time=127. ms
64 bytes from zeno.fit.edu (163.118.5.4): icmp_seq=2. time=128. ms
64 bytes from zeno.fit.edu (163.118.5.4): icmp_seq=3. time=120. ms
64 bytes from zeno.fit.edu (163.118.5.4): icmp_seq=4. time=130. ms
^C
----4.5.118.163.in-addr.arpa PING Statistics----
5 packets transmitted, 5 packets received, 0% packet loss
round-trip (ms) min/avg/max = 120/126/130
```

FIGURE 3.17 Output of the *ping* program using the *s* option.

the communication path between it and a destination node. The output of this command in verbose mode is shown in Figure 3.17. Normally ping simply reports whether a destination node is "alive." In verbose mode, ping reports the roundtrip time of a packet between source and destination. Note how ping gives specific information about the condition of the network environment. For example, lengthy roundtrip times would indicate some sort of problem between the source and destination nodes, and the percent of packet loss is also beneficial in assessing the condition of a link.

Another protocol that operates at TCP/IP's network layer is the *Resource Reservation Protocol* (RSVP), which was recently developed by the Internet Engineering Task Force (IETF). RSVP can be thought of as an IP-based *quality of service* (QoS) protocol that provides a mechanism to control network latency for specific applications. (See Chapter 5 for additional information about QoS.) This is done by prioritizing data and allocating sufficient bandwidth for data transmission. QoS is inherent in technologies such as token ring and ATM, but is absent in Ethernet/802.3 and IP. With RSVP, though, Ethernet/802.3 or IP end nodes can reserve a specific amount of bandwidth from the network for a particular transmission. This feature is critical for transmitting data from time-sensitive applications such as real-time voice and video. For example, a videoconferencing application might receive a high-priority tag that requires a certain amount of bandwidth, a specific transmission rate, and maximum latency.

For RSVP to be effected across a WAN, every router that is along the path an RSVP data packet traverses must support RSVP. If not, then the application fails. Furthermore, if the WAN cannot support an RSVP request (e.g., there is insufficient bandwidth available), then the application will not run. These two issues pose serious challenges to running an RSVP application across the Internet. The IETF, however, is working diligently to address these issues. For example, IETF working groups have been established to integrate RSVP into the OSPF and BGP routing protocols. On another front, the IETF's Integrated Services Working Group is considering modifying the type of information contained in an IP packet header's type of service field to identify the level of service a packet should receive. Finally, IPv6, the next generation of IP that is discussed later in the chapter, has provisions for QoS via the 24-bit flow label field (see Figure 3.23). ATM's QoS offers a more elegant approach to transmitting time-sensitive data across a WAN than RSVP. However, the current level of work and attention IETF is giving to the issue of QoS makes the concept of IP QoS more of a reality than a contradiction in terms.

74. Although you have mentioned Internet addresses before, you really haven't discussed them in any detail. Can you do this now, please?

Yes. In fact, this is a good place to do it. Internet addresses (called IP addresses for short) are node addresses that identify their location within the Internet. IP addresses play an important role in the successful delivery of data across the Internet. If a node cannot be located, then data cannot be delivered to it. Currently, two versions of IP addresses are available: Version 4 (IPv4) and version 6 (IPv6).

75. Please describe IPv4 addresses.

Okay. IP Version 4 addresses (IPv4) consist of 32 bits (0 through 31) partitioned into four groups of eight bits each (called octets). Since it is difficult for us to read addresses in binary notation, IP addresses are expressed in decimal form; a decimal point (read as "dot") separates the octets. An example of an Internet address is 204.163.25.37. Each octet of an IP address is treated as an independent unit. Since octets comprise eight bits, the contents of each octet can contain anywhere from eight 0-bits to eight 1-bits. The decimal equivalent of an all 0-bit octet is 0, and the decimal equivalent of an all 1-bit octet is $(2^8 - 1) = 255$. This is illustrated in Figure 3.18(a) with the letter x being used in a general manner to represent bits. The bit pattern for the IP address 204.163.25.37 is shown in Figure 3.18(b).

76. What about IP address schemes? For example, what's the difference between a Class B and a Class C address?

IPv4 addresses are organized into one of five classes: A, B, C, D, or E. Classification is determined by the value of the first four bits (bits 0 through 3).

Class A Addresses. If bit 0 is 0, then the address is a Class A address and begins with a decimal number ranging from 0 to 127. (0 and 127 are reserved.) Thus, the bit pattern of the first octet ranges from 00000000 to 01111111. An example of a Class A address is 13.123.17.8.

Class B Addresses. If the first two bits are 10, then the address is a Class B address and begins with a decimal number ranging from 128 to 191. Thus, the bit pattern of the first octet ranges from 10000000 to 10111111. An example of a Class B address is 163.118.5.4.

Class C Addresses. If the first three bits are 110, then the address is a Class C address and begins with a decimal number ranging from 192 to 223. Thus, the bit pattern of the first octet ranges from 11000000 to 11011111. An example of a Class C address is 198.42.239.17.

Class D Addresses. If the first four bits are 1110, then the address is a Class D address and begins with a decimal number ranging from 224 to 239. Thus, the bit pattern of the first octet ranges from 11100000 to 11101111. Class D addresses are used for multicast addresses.

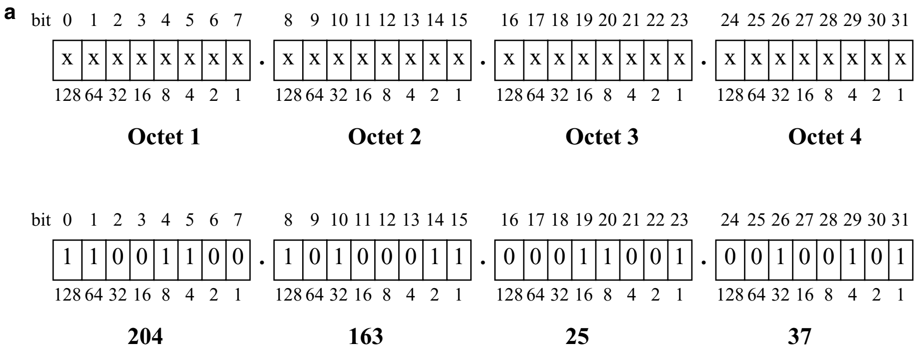


FIGURE 3.18 IP Version 4 addresses are 32 bit-length addresses partitioned into four octets. These octets are shown in (a) using the letter x to represent bits. In (b), the binary and decimal equivalents of the IP address 204.163.25.37 are shown.

Class E Addresses. If the first four bits are 1111, then the address is a Class E address and begins with a decimal number ranging from 240 to 255. All Class E addresses are reserved for future use.

In addition to address classes, IPv4 addresses also distinguish between the address of a network (called the *network identifier*) and the address of a host (called the *host identifier*) connected to a specific network. This is similar to a telephone number—an area code identifies the region, and the phone number identifies a specific location in that region. Class A addresses use bits 1 through 7 for the network ID and bits 8 through 31 for the host ID; Class B addresses use bits 2 through 15 for the network ID and bits 16 through 31 for the host ID; and Class C addresses use bits 3 through 23 for the network ID and bits 24 through 31 for the host ID. This is illustrated in Figure 3.19.

Finally, each address class allows a different maximum number of possible networks and hosts. These upper limits are based on the number of bits used for the network and host identifiers. For example, since Class A addresses use seven bits (bits 1 through 7) for the network identifier and 24 bits (bits 8 through 31) for the host identifier, there are a maximum of $2^7 = 128$ Class A networks, each capable of supporting $2^{24} = 16{,}777{,}216$ hosts. Similarly, Class B addresses have a maximum of $2^{14} = 16{,}384$ networks and $2^{16} = 65{,}536$ hosts; and Class C addresses support $2^{21} = 2{,}097{,}152$ networks, each capable of supporting $2^8 = 256$ hosts. Since IPv4 reserves the use of all 0s or all 1s for special addresses, each of the preceding calculations should be reduced by 2 to yield the true number of unique networks and hosts allowable for each network class.

77. I also hear people use the term "subnetting." What does this mean?

Subnetting refers to the partitioning of a network address space into separate autonomous subnetworks. To understand this concept, let's first set the stage by examining a "flat" Class C network, which does not use subnetting. A flat Class C address consists of

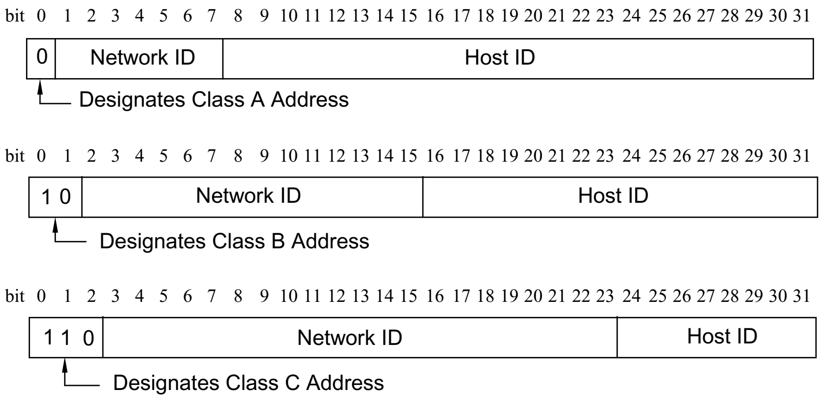


FIGURE 3.19 IPv4 addresses can also be interpreted from a network- and host-ID perspective. In a Class A address, bits 1–7 represent the network ID, and the remaining bits represent the host ID. In a Class B address, bits 2–15 stand for the network ID, and bits 16–31 represent the host ID. Finally, in a Class C address, bits 3–23 specify the network ID, and the last eight bits represent the host ID. In addition, the first three bits of an IPv4 address specify the address class. (Class D and Class E addresses are not shown but follow a similar pattern.)

256 host IDs. As discussed earlier, IP reserves two of these IDs for special purposes. These are always the first and last host IDs. The first ID is used to identify the network itself (although historically this is the "all 0s" broadcast address); it represents the overall network address. The last ID specifies the network's broadcast address. Thus, given the Class C network identifier 198.42.17, the address 198.42.17.0 is reserved for network identification; the address 198.42.17.255 is reserved for broadcasting; and the remaining 254 addresses, from 198.42.17.1 to 198.42.17.254, are used as host IDs. As a result, an unsubnetted Class C address consists of one network address, one broadcast address, and 254 unique host IDs.

When an IPv4 address is subnetted, the flat address space is partitioned into multiple subnetworks. Through subnetting, we can make efficient use of an IPv4 address because subnetting reduces wasted address space. For example, suppose an organization has been assigned the Class C network address 198.42.17.0 but only has 60 nodes to connect to the Internet. Without subnetting this address space, 194 host addresses are being unused. This is wasteful given the current state of IPv4 address depletion. With more and more dedicated network appliances such as e-mail stations and MP3 players, each device requires an IP address. A more efficient use of this network address space is to partition it into smaller sizes so that fewer host addresses are unused. In this case, we would partition the network into four subnetworks. So instead of having 256 possible host IDs, each subnetwork would have $256 \div 4 = 64$ host IDs. Since one host ID must be reserved for the subnetwork address and a second ID for the subnet's broadcast address, each subnetwork can only support a maximum of 62 host IDs. Nevertheless, for the given situation, this is quite adequate resulting in only 2 unused host IDs as opposed to 194. Subnetting is an efficient way of using the limited IPv4 address space.

78. How does subnetting work?

Key to subnetting is a network's *subnet mask*. When used with subnetting, a subnet mask specifies which bits are part of the network ID and which bits are part of the host ID. To illustrate this concept, consider once again the Class C network address 198.42.17.0. Without subnetting, this address has the corresponding mask of 255.255.255.0. Note that a mask is expressed in dotted notation similar to a network address and is also 32 bits in length. Note further that the binary representation of this mask consists of 24 consecutive 1-bits followed by 8 consecutive 0-bits:

11111111	**11111111**	**11111111**	**00000000**
255	255	255	0

A network address and its corresponding mask are denoted by writing the network address and the number of 1-bits contained in the mask and delimiting the two expressions by a slash symbol. Thus, in this example, the expression 198.42.17.0/24 denotes that the network address has a corresponding mask of 24 consecutive 1-bits, which implies that the mask's remaining octet consists of 8 consecutive 0-bits.

When a network address is combined with its corresponding subnet mask via a logical AND, the network ID part is preserved but the host ID becomes 0. In this way, the subnet mask is used to mask the network ID from the host ID. As an example, consider the Class C address 198.42.17.239. Given the subnet mask 255.255.255.0, we can extract the network ID by combining the two addresses via a logical AND, as is shown here: (*Note:* The logical AND of two bits is equal to 1 only when the two bits being combined are 1. In all other cases, the result is 0. Thus, 0 AND 0 = 0, 0 AND 1 = 0, 1 AND 0 = 0, and 1 AND 1 = 1.)

198.42.17.239:	11000110	00101010	00010001	11101111
255.255.255.0:	11111111	11111111	11111111	00000000
Result of AND:	11000110	00101010	00010001	00000000
	198	**42**	**17**	**0**

As a result, the subnet mask is able to correctly determine the network number that the destination node is connected to (i.e., network 198.42.17). Furthermore, by adding the host ID part of the address (239) to the corresponding bits of the mask (i.e., 11101111 + 00000000), the correct destination node is also identified. Thus, the IPv4 address 198.42.17.239 is correctly identified as node 239 on network 198.42.17.0.

79. Was any of this really necessary? Wouldn't the address itself, namely, 198.42.17.239, automatically provide this information?

Yes. You're right. However, as indicated earlier, IPv4 addresses are wasteful by default and hence are partitioned into multiple subnetworks. So if subnetting was in effect, the previous interpretation of 198.42.17.239 would be incorrect.

80. Okay. So how is subnetting implemented?

Subnetting is effected by "stealing" bits from the host ID part and joining them to the network ID part. When taking bits from the host ID, we always use the higher-order bits because they are adjacent to the network ID bits. For example, one subnetting scheme involves using one host ID bit as part of the network address. In this setting, the network ID part would contain 25 bits and the host ID part would contain 7 bits. This scheme results in two subnetworks, with each subnet having its own network and broadcast addresses. Furthermore, since only 7 bits remain in the host ID part, each subnetwork supports a maximum of $2^7 - 2 = 126$ host IDs. (Remember that one address is used to specify the subnetwork's address, and a second address is used as the subnetwork's broadcast address.)

81. Could you illustrate this using the same Class C address?

Certainly. If the Class C address 198.42.17 were partitioned using this scheme, then the two subnetworks and corresponding subnet host addresses are as follows:

Subnet	Subnet Address	Host ID Range	Broadcast Address
1	198.42.17.0	198.42.17.1 to 198.42.17.126	198.42.17.127
2	198.42.17.128	198.42.17.129 to 198.42.17.254	198.42.17.255

These subnets are denoted 198.42.17.0/25 and 198.42.17.128/25. Here, the /25 indicates that the given network address has a corresponding subnet mask of 25 consecutive bits, which implies that the mask's last octet is 10000000. The reason we need a mask with 25 consecutive 1-bits is because we used 1 bit from the host ID to create a 25-bit network ID. To extract the correct network ID, we have to AND it with the mask, which must have the correct number of consecutive 1-bits to correspond with the network ID's length.

Note that when subnetting is used, it is extremely critical to correctly identify the subnet to which a host is connected. As an example, consider once again the destination address 198.42.17.239 from our earlier discussion. Without specifying a corresponding subnet mask, the default mask is 255.255.255.0, and the address is interpreted as node 239 on network 198.42.17.0. However, this interpretation is incorrect in the current situation because the destination address really specifies node 111 on subnetwork 198.42.17.128. To make the correct interpretation, the address needs to be denoted as 198.42.17.239/25, which once again implies that a subnet mask consisting of 25 consecutive 1-bits followed by 7 consecutive 0-bits is used. The concept of subnetting and the use of subnet masks are demonstrated further in Box 3.2. If you do not want to bother with the arithmetic, check out http://www.vcsa.org/computer/subnet.html for /26, /27, /28, /29, and /30 Class C subnetting.

82. So that's how it works. I often wondered what that slash thing was after an Internet address. Is there anything else I need to know about subnetting? For example, can you subnet other classes like a Class B address?

Yes. Although our subnetting discussion focused on Class C addresses, Class A and Class B addresses can also be subnetted following a similar scheme. For example, the Class B address 163.118.0.0 by default specifies a flat network address space with a 16-bit network ID and a 16-bit host ID. Thus, this unsubnetted network address comprises a single network with 65,536 host IDs with a default subnet mask of 255.255.0.0. This same address can be subnetted into 256 autonomous subnets with each subnet supporting 256 host IDs by using 8 higher-order bits of the host ID as part of the network ID. By doing so, the network ID will consist of 24 bits, the host ID will consist of 8 bits, and the corresponding subnet mask will be 255.255.255.0. Hence, this scheme effectively subnets a Class B address into 256 Class C equivalent addresses. Once again, note how much more efficient this is than maintaining a single (flat) address space with 65,536 nodes.

83. Is it possible to subnet a subnet?

Yes it is. Subnetted networks themselves can also be subnetted. For example, a Class C address could first be partitioned into two subnets via a /25 mask. One of these subnets could then be further partitioned into two sub-subnets with a /26 mask, and the other subnetwork could be partitioned into three sub-subnets with a /27 mask. Similarly, a Class B address could be subnetted into 256 Class C equivalent addresses with a /24 mask, and then any one of these subnets could be further subnetted using an appropriate mask of varying lengths. This concept of partitioning a network address into unevenly sized subnets is called *variable length subnet masking*.

84. How are IP addresses assigned?

IP address assignments are handled in a distributed fashion via an Internet registry (IR) system, which is hierarchical in structure and involves several organizations. At the root of this hierarchy is the Internet Assigned Numbers Authority (IANA). The IANA allocates blocks of IP address space to regional Internet registries (RIR), which in turn allocate blocks of IP address space to their local Internet registries (LIR), who assign the addresses to end users. Regional registries are established under the authority of IANA and include the American Registry for Internet Numbers (ARIN), the Asian-Pacific Network Information Center (APNIC), and Réseaux IP Européens Network Coordination Centre (RIPE NCC). ARIN is the RIR for North America, South America, the Caribbean, and sub-Saharan Africa; APNIC is the RIR for countries in the Asian Pacific region (e.g., Japan, China, Thailand); and RIPE NCC is the RIR for Europe, the Middle East, and parts of Africa. In some cases, local registries are also Internet service providers (ISPs).

The way IP address assignments work is as follows: IANA allocates blocks of IP address space to regional Internet registries. The RIRs allocate blocks of IP address space to their local Internet registries. LIRs then assign addresses to either end users or ISPs. The assignment of IP addresses using this structure enables routing information for end users to be aggregated once it leaves a provider's routing domain. This reduces the number of route announcements and state changes throughout the Internet. (See Chapter 7 for more information about routing.)

85. A friend of mine who works at an ISP told me they receive blocks of IP addresses that use a slash notation similar to what we discussed earlier. Is this the same thing?

The slash concept we introduced earlier was presented in the context of subnetting a specific IP network address and referred to the number of consecutive bits that corresponded to the subnet mask. What your friend is referring to is another context for the slash notation, namely, for assigning and routing Internet addresses. Recall that IPv4 addresses are 32-bits

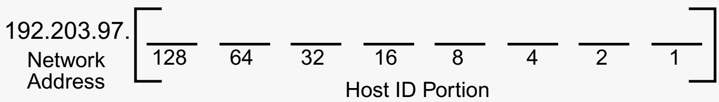
BOX 3.2: Subnet Masking

A subnet mask partitions an IPv4 address into two parts: the network ID and host IDs. The concept of subnet masking is as follows:

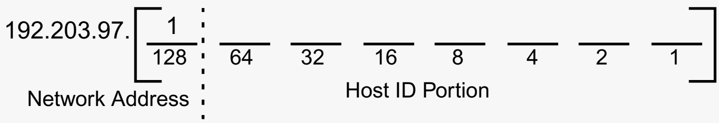
> *If a bit in an IPv4 address is used as part of the network ID, then the corresponding mask bit is set to 1. If a bit in an IP address is used as part of the host ID, then the corresponding mask is set to 0.*

For example, the Class C IPv4 address 192.203.97.0 with subnet mask 255.255.255.0 implies that the network address is 192.203.97, the host ID addresses correspond to this network range from 192.203.97.1 to 192.203.97.254, and the network broadcast address is 192.203.97.255.

To conserve the IPv4 address space, network addresses are subnetted by borrowing bits from the host ID portion of the network address. To illustrate this concept, consider the following Class C address, which has its last octet (the part that represents the host ID) expanded into its eight-bit equivalent with corresponding decimal values.



If we "steal" the highest-order bit of the host ID portion of this address (i.e., 128) to use as part of the network address, then the subnet mask becomes 255.255.255.128.



Thus, one bit is for subnetting and seven bits are for host IDs. This yields $2^1 = 2$ subnets and $2^7 = 128$ host IDs. Since each network requires one address to name the network and one for broadcasting, the actual number of host IDs per subnet is $2^7 - 2 = 126$.

	Subnet 1	**Subnet 2**
Network Address:	192.203.97.0	192.203.97.128
Host IDs:	192.203.97.1 to 192.203.97.126	192.203.97.129 to 192.203.97.254
Broadcast Address:	192.203.97.127	192.203.97.255

(*Note:* It should be observed that the "all 1s" subnet is not recommended in the RFCs and hence should be avoided. As a result, having a mask one-bit wide would allow only one subnet to be defined, subnet 0; subnet 128 would be illegal.)

(continued)

BOX 3.2: Subnet Masking (continued)

Similarly, if the two left-most bits of the host ID are used (128 + 64), the subnet mask is 255.255.255.192. This implies $2^2 = 4$ subnets, each with $2^6 - 2 = 62$ host IDs.

	Subnet 1	**Subnet 2**	**Subnet 3**	**Subnet 4**
Network Address:	192.203.97.0	192.203.97.64	192.203.97.128	192.203.97.192
Host IDs:	192.203.97.1 to 192.203.97.62	192.203.97.65 to 192.203.97.126	192.203.97.129 to 192.203.97.190	192.203.97.193 to 192.203.97.254
Broadcast Address:	192.203.97.63	192.203.97.127	192.203.97.191	192.203.97.255

Routing protocols need to support subnet masking in order to determine correct routing information. For example, consider the destination address 192.203.97.143 with the subnet mask 255.255.255.192. Let's examine an expanded view of these addresses.

$$\textbf{a} \quad 255.255.255.\left[\frac{1}{128}\ \frac{1}{64}\ \vdots\ \frac{0}{32}\ \frac{0}{16}\ \frac{0}{8}\ \frac{0}{4}\ \frac{0}{2}\ \frac{0}{1}\right]$$

$$\textbf{o t} \quad 192.203.97.\left[\frac{1}{128}\ \frac{0}{64}\ \vdots\ \frac{0}{32}\ \frac{0}{16}\ \frac{1}{8}\ \frac{1}{4}\ \frac{1}{2}\ \frac{1}{1}\right]$$

The correct subnet is determined by "masking" the bits in the host field from the 1-bits in the mask. This is done by combining these bits via a logical AND.

a	1 1 : 0 0 0 0 0 0
o t	1 0 : 0 0 1 1 1 1
e ult o logical A	1 0 : 0 0 0 0 0 0

Since $10000000_2 = 128_{10}$, the correct subnet is 128. That is, the host with address 192.203.97.143 is attached to the subnetwork whose address is 192.203.97.128. Furthermore, when the six bits of the host ID (001111) are added to the corresponding bits of the mask, the result is 001111_2, which is equal to 15_{10}. Thus, the destination address, 192.203.97.143 is correctly identified as host 15 on subnet 128 of network 192.203.97. If the routing protocol does not support subnet masking, then the destination address corresponds to host ID 143 on network 192.203.97.0, which is incorrect.

in length and are organized by classes, namely, A, B, and C. Class A addresses contain an eight-bit network ID and a 24-bit host ID; Class B addresses contain a 16-bit host ID and a 16-bit host ID; and Class C addresses contain a 24-bit network ID and an eight-bit host ID. When IP address assignments are made to local registries, the regional registries no longer use this concept of "classful addressing." Registries now use a *classless* addressing strategy, which enables the network ID of Class A, B, and C addresses to be any size instead of being restricted to 8, 16, or 24 bits. This strategy is formally called *Classless Interdomain Routing* (CIDR, pronounced "cider"). When assigning IPv4 addresses, quantities of address space are referred to by their prefix length (i.e., the number of leading bits that specify the network ID) and are denoted by a slash notation. Thus, /24 refers to a single conventional Class C IPv4 network because /24 implies that the network ID consists of 24 bits.

As an illustration, consider an organization that needs to assign IP address to 4000 hosts. Prior to CIDR, this organization would be assigned a Class B address because a Class C address would only provide $2^8 = 256 - 2 = 254$ unique host addresses. However, a Class B address supports $2^{16} = 65{,}536 - 2 = 65{,}534$ hosts, which means that more than 61,000 IP addresses will be unused and cannot be assigned to another organization. A CIDR strategy would allocate a block of addresses with a /20, which indicates that 20 bits are used for the network ID and $32 - 20 = 12$ bits for the host ID. More specifically, a /20 designation means that the organization is allocated $2^{12} = 4096$ IP addresses, which is equivalent to 16 Class C addresses. (*Note:* 16 is obtained as follows: $24 - 20 = 4$ and $2^4 = 16$. We subtract 20 from 24 because /24 denotes a "regular" Class C IPv4 network address.) Thus, the slash prefix in this context specifies how many "networks" are covered by the CIDR address.

CIDR was developed as a solution to the routing table explosion caused by the rapid growth of the Internet. As the Internet grew, the folks in charge of assigning Internet addresses began issuing multiple class C addresses to organizations that had more than 256 hosts to connect to the Internet. This meant that more individual networks needed to be announced for routing purposes and maintained by routers. CIDR addressed this issue by summarizing Class C network prefixes and using them as routing entries instead of the actual network addresses. For example, in a CIDR-less environment, an organization that is issued 64 Class C network addresses must have an entry for each address in its router's routing table. That's 64 entries. With CIDR, though, a special prefix (/18) is assigned to these 64 networks to indicate that they all belong to the organization's routing domain. Thus, only one routing entry is needed instead of 64.

86. OK. Thanks for the info. Whenever I use the Internet I never deal with IP addresses. I always use real names like www.att.com. Is there some sort of translation that takes place that's transparent to me?

Every host (well, almost every host) that is connected to the Internet has both an IP address and a corresponding logical name. This logical name, which is generically referred to as a *domain name*, is another type of addressing construct used for identifying Internet nodes. For example, the IP address that corresponds to www.att.com is 192.20.3.54. The translation from logical name to IP address, called *name resolution*, makes it easier for us to deal with Internet addresses. After all, we are mere humans and, as such, have a limited capacity for memorizing long strings of numbers, particularly when they are formatted as IP addresses, and especially if they are IPv6 addresses, which are 128 bits long. Without name resolution, Web site addresses or e-mail addresses could not employ names of organizations, descriptive titles, acronyms, and the like. They would all be numerically based. (Remember: You are nobody unless you are somebody@somewhere.com.)

87. Could you explain this translation process?

Yes, but before we explain the translation process, it might be helpful first to understand the concept of domain names. Domain names are organized in a hierarchical treelike fashion. At the top of the tree are top-level domains. When name resolution was first developed, there were seven three-letter descriptive top-level domains that represented general entities, plus two-letter country codes assigned to different countries throughout the world. The top-level domains include *com* for commercial organizations (e.g., for-profit companies); *edu* for educational organizations (currently restricted to 4-year colleges and universities; all other schools are part of the *us* country domain); *gov* for government organizations or agencies (currently restricted to U.S. federal government agencies); *int* for international treaty organizations; *mil* for U.S. military organizations; *net* for Internet service providers; and *org* for any miscellaneous organization including not-for-profit organizations. Several new top-level domain names were approved in late 2000, including *info* for general use, *biz* for business, *name* for individuals, *pro* for professionals, *museum* for museums, *coop* for business cooperatives, and *aero* for the aviation industry. The two-letter country codes are those specified in the ISO-3166 document. Examples include *us* for United States, *ca* for Canada, and *au* for Australia. Under each top-level domain are sublevels, and each lower level is considered a subdomain of its immediately preceding higher level.

When expressed as a logical Internet address, the domain name hierarchy is structured from right to left—the top level is at the right and the bottom level at the left—and consists of multiple levels separated by dots. For example, www.att.com is a three-level structure. Its top-level domain is com, att is a subdomain within com, and www is the name of a host within att. If the lowest level is the name of a host, then the entire structure is called a *fully qualified domain name*, or FQDN. As another illustration, consider the six-level domain name, pirate.pbhs.brevard.k12.fl.us. Its top-level domain is us; fl is a subdomain in us (it's for the state of Florida); k12 is a subdomain within fl (it represents public schools from kindergarten through 12th grade); brevard is a subdomain of k12 (it's the name of a specific county school district); pbhs is a subdomain of brevard (it's the name of a specific school); and pirate is the name of a host within pbhs. This is another fully qualified domain name (Figure 3.20).

To translate domain names to IP addresses, a domain name service is used. The translation process involves configuring host machines as domain name service (DNS) servers. DNS servers store specific address information about a particular domain or subdomain. They are located at each top-level domain and at various subdomains within a top-level domain. Special root servers are also used to process interdomain DNS requests. Each subdomain has at least one DNS server that is authoritative for that domain. This means that it contains accurate and complete domain name-IP address resolution records for all the connected hosts within that domain. For example, in Figure 3.21, let's assume that a user on host pirate wants to connect to the Web server raider, which is in the rhs.brevard.k12.fl.us do-main. The Web browser on pirate places a DNS query to the DNS server for brevard.k12.fl.us, which is authoritative for the brevard.k12.fl.us do-main. This DNS server looks up the information in its database and returns the address 204.128.64.2. Now suppose this same user wants to connect to the Web server www.att.com. A similar

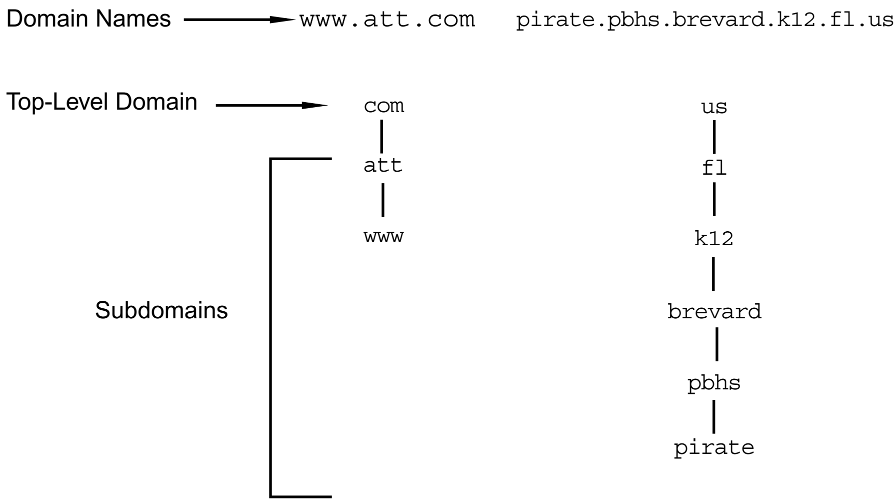


FIGURE 3.20 Domain names are structured in a right-to-left hierarchy. The top-level domain is the rightmost part of an address. All sublevels are subdomains of their immediately preceding higher-level domain. If a domain name includes the hostname as its lowest level, then the structure is called a fully qualified domain name.

query is made to the brevard.k12.fl.us DNS server. This time, though, it does not have any information about the att.com domain and hence cannot resolve the name. This DNS server then initiates a process of trying to get the name resolved. Each DNS server has the address of other name servers, including at least one root server. If a DNS server cannot resolve a name, it replies by specifying the name server that should be contacted next. Eventually, a root server gets involved. Root servers maintain information about all the authoritative name servers for each top-level domain. Thus, if the brevard name server cannot get www.att.com resolved by the higher-level name servers within its domain, a root server eventually provides it with the att.com DNS server's address. The brevard name server then contacts the att.com name server, which returns the address 192.20.3.54. The brevard name server then sends this information to the client process on pirate. The brevard server also stores this address locally in its cache along with other names it has resolved recently. This helps reduce the number of DNS lookups and makes the DNS process more efficient.

A UNIX and Microsoft NT program that is used to acquire the IP address of a domain name is *nslookup*. To find the IP address of a domain name, simply enter nslookup followed by the domain name. For example, "nslookup www.att.com" returns the address 192.20.3.54. This command can also be used for IP address resolution, which translates numerical IP addresses to corresponding domain names.

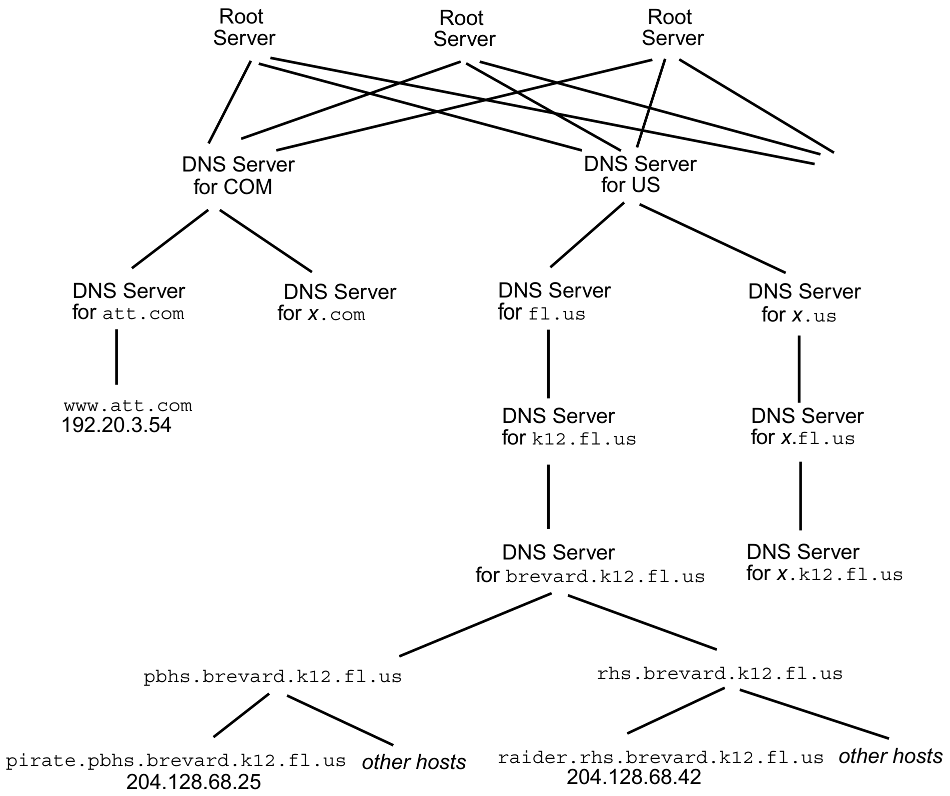


FIGURE 3.21 DNS servers provide name resolution service. Each subdomain has at least one name server that is authoritative for that domain. That is, it maintains complete and accurate information about all the hosts within its subdomain. Name servers also have the address of other servers they can contact in the event they cannot resolve a name locally. They also have the address of at least one root server, which provides interdomain name resolution capability.

88. Is DNS the only method of name resolution?

DNS is not the only method of name resolution. A simple but inefficient alternative is to maintain on every local machine a host file that contains the fully qualified domain names and their respective IP addresses of all the hosts you need to contact. Some operating systems enable you to use both DNS and host files. For example, a host might be configured to use host table lookup first and DNS second. This way, if a host cannot resolve a hostname via its local host file, it can then place a DNS query.

89. OK. Let's move on to IPv6. Why was it developed?

IPv6 was developed for the same reason other new Internet protocols have been developed or existing ones modified: growth. Due to the tremendous growth of the Internet, the number of available IPv4 addresses is shrinking. Furthermore, with the deployment of more networks, routing tables are overflowing and unable to handle the demand of maintaining information about every network. This depletion of IP addresses and routing table overflow is analogous to the telephone company running out of area codes and old telephone switches unable to place calls to certain numbers. This is why we now have area codes with middle digits other than 0 or 1, a new toll-free 888 exchange in addition to the traditional 800 exchange, and multiple area codes serving the same area. Although the original developers of TCP/IP in the early 1980s accounted admirably for the future growth of TCP/IP, no one could have predicted the exponential growth that is occurring within the Internet today.

Although the driving force behind this phenomenal growth can be linked to the computer market, this probably will not be the case in the near future. Indeed the computer market will continue growing; there are still some schools and small businesses not yet connected to the Internet. However, the future growth of the Internet will be fueled by several new markets including the following:

Personal Communication Devices. Fax machines, personal digital assistants (PDAs), telephones, and nomadic units that rely on wireless communications are very popular. Eventually, these devices will be network-addressable and hence require a unique IP address. Furthermore, some of these devices must be capable of automatically configuring (and reconfiguring) themselves with correct addressing information when moved from one remote location to another.

Networked Entertainment. Video-on-demand, stereos, CDs, and interactive television will one day be network-addressable. For example, in a world of networked television sets, every TV will effectively become an Internet host and require a unique IP address.

Networked Controlled Devices. A wide range of devices from simple appliances such as lights and electronic security devices, to heavy motors such as heating and air conditioning units, or any other equipment currently controlled by analog switches, will be controlled via networked communications.

Each of these markets is huge, and they all have a common attendant characteristic—they all bring to the table a new set of requirements that IPv4 does not support. For example, there will be a need for (among others) large-scale routing, automatic configuration and reconfiguration of host addresses, and built-in authentication and encryption for security purposes. In addition to providing such functionality, it also is imperative that the new IP continues to support existing applications. Most importantly, though, the expected demand for wireless devices will alone require billions of new IP addresses in the next few years.

90. OK. What does IPv6 bring to the table?

IPv6, which is sometimes called IPng (for next generation), provides a solution that resolves current addressing problems and is capable of delivering the necessary functionality for emerging markets. Designed as an evolutionary replacement, IPv6 maintains most IPv4 functions, relegates certain functions that either were not working or were rarely used in IPv4 as optional, and adds new functionality that is missing from IPv4.

Some of the specific differences between IPv4 and IPv6 are readily apparent when you examine the format and content of their headers, shown in Figures 3.22 and 3.23, respectively. Three obvious differences are (a) size—the IPv4 header length is of variable length because of its Options and Padding fields, but IPv6 is a fixed 320 bits; (b) the number of fields—IPv4 has 14 fields, but IPv6 has only 8; and (c) address field size—source and destination addresses are 32 bits each in IPv4, but 128 bits each in IPv6. The address fields make up 80% of the IPv6 header (256 bits). Without these two fields, the IPv6 header is only 64 bits in length, which makes it much smaller than the corresponding IPv4 header. (*Note:* Although the IPv4 header has 14 fields, it is unusual to see more than 12 in use. The Options field is rarely used and hence the Padding field, which is only used for alignment when options are used, is also rarely required.)

Although IPv6 embodies a "less is more" philosophy-structure, it still supports a wide variety of options. Consider, for example, the Next Header (NH) field, which specifies the type of header that immediately follows the IPv6 header. NH enables extension headers for optional layer 3 data to be inserted between the IP header and the upper layer headers (e.g., TCP or UDP) that precede user data. An example of this field is the use of IPSec's authentication and encryption headers for security. (See Chapter 16.) NH effectively combines the functions of IPv4's Protocol, Header Length and Option fields. IPv6 also provides support for prioritizing traffic via its 4-bit Priority (P) field, and for assigning special handling designations (e.g., QoS) to packets through its 24-bit Flow Label (FL) field. Both of these features were not available in IPv4. Additional information about IPv6 can be found in RFC 1883, which contains a complete set of specifications for the new protocol. The Internet-Draft, "The Case for IPv6," also contains information about IPv6.

91. What's the scoop on IPv6 addresses?

As stated above, one of the most significant differences between IPv4 and IPv6 is address size, which was increased from 32 bits in IPv4 to 128 bits in IPv6. Thus, IPv6 addresses have four times as many bits as IPv4 addresses (128 vs. 32). This means there are 2^{128} IPv6 addresses versus 2^{32} IPv4 addresses. Evaluating 2^{128}, Hinden (1996) calculated this value as 340,282,366,920,938,463,463,374,607,431,768,211,456. When compared to IPv4's 4 billion addresses (i.e., 4,294,967,295), you start to get some feel for the size of IPv6's address space. Hinden also reported that there are enough 128-bit IPv6 addresses to, at worst, provide each square meter of the Earth's surface with 1564 addresses, and at best, 3,911,873,538,269,506,102 addresses. The new address space also allows for more levels of addressing hierarchy, simpler autoconfiguration of addresses, and will support other network protocol addresses. Other notable features include:

- Support for three types of addresses—unicast, anycast, and multicast. Unicast addresses identify a single interface. Anycast addresses are assigned to a set of interfaces and routed to the nearest one assigned to that address. The determination of which node is nearest is based on the metric used by the routing protocol. (See Chapter 7.) Multicast addresses are assigned to a group of interfaces and delivered to all of the interfaces in the assigned group. (*Note:* IPv6 addresses are assigned to physical interfaces, not nodes.)

4	4	8	16	16	3	13	8	8	16	32	32	Variable	
V	HL	ST	TL	ID	F	FO	TTL	P	HC	SA	DA	OPT	PAD

This 4-bit eld speci es the protocol *version* number. For IPv4, it is 0100.

This 4-bit eld speci es the *header length* in 32-bit words. It is needed because the OPT and PAD elds do not have xed lengths. (The header length does not include the length of the user data eld, which immediately follows the PAD eld. The data eld is not shown here because it is not part of the header.)

S This 8-bit *service type* eld speci es the manner in which a packet (i.e., datagram) is routed. It contains three sub elds, as shown below:

Precedence (3 bits)	Type of Service (4 bits)	MBZ (1 bit)

¥ The *precedence* sub eld speci es the priority of the datagram (from 000 = normal to 111 = network control).

¥ The *type of service* (TOS) eld speci es transport control information relative to delay, throughput, reliability, and cost. For example, 1000 = minimize delay; 0100 = maximize throughput; 0010 = maximize reliability; 0001 = minimize monetary cost; and 0000 = normal service (default). (See RFC 1349 for more information about TOS.)

¥ The *MBZ* (must be zero) eld is currently unused.

This 16-bit eld speci es the *total length* of the packet. Given the eld 16 bits, the maximum size of an IPv4 packet is 2^{16} = 65,535 bytes.

This 16-bit eld speci es the unique *identification* number that was assigned to the packet. This number is used to reassemble fragmented packets.

F This 3-bit *flag* eld is used to control fragmentation.

F This 13-bit *fragment offset* eld provides reassembly information for fragmented packets.

This 8-bit *time to live* eld is a counter (often called a *hop count*) that speci es the number of seconds a packet is permitted to remain alive (i.e., active) on the Internet. This eld gets decremented whenever it is processed by a router. When TTL = 0, the packet is discarded and an error message is sent to the source node that sent the packet.

P This 8-bit eld speci es the layer-4 *protocol* used to create user data.

This 16-bit eld contains *header checksum* information, which is used for maintaining packet integrity.

SA This is the 32-bit IP *source address* (see Box 3.2).

A This is the 32-bit IP *destination address* (see Box 3.2).

P This variable-bit eld is reserved mostly for control *options* (e.g., network testing or debugging). Eight options are available, and the length of this eld varies depending on the option used.

PA This variable-length *padding* eld is used in conjunction with the option eld. It pads the option eld with enough 0 bits to ensure that the header length is a multiple of 32 bits.

FIGURE 3.22 Format and contents of an IPv4 header. The length of each field is given in bits. Source: Adapted from RFC 791, RFC 1702, and RFC 1349.

4	4	24	16	8	8	128	128
V	P	FL	PL	NH	HL	SA	DA

This 4-bit eld speci es the protocol version number. For IPv6, it is 0110. This is the only eld that has exactly the same meaning and position in both IPv4 and IPv6 headers.

P This 4-bit eld speci es the priority of the packet data. This eld is new to the IP header; it was not part of IPv4. There are $2^4 = 16$ different priority levels, which are divided into two groups. The rst group, which is speci ed by priority levels 0 through 7, designates packets that can respond to congestion control. For example, in an IP-based frame relay network, in the presence of congestion, the destination node can reset its transmission window value to 0, which effectively informs the sender to stop transmitting data to the receiver until a nonzero window size is received from the receiver (see Chapter 12). The second priority group is speci ed by priority levels 8 through 15 and designates packets that cannot respond to congestion control. This second priority group is used for critical data such as voice and video. These packets will not back off in response to congestion control.

F This 24-bit ow label eld designates packets that require special handling. One use of this eld is to provide quality of service (QoS) via RSVP. This eld is new to the IP header; it was not part of IPv4.

P This 16-bit eld speci es the payload length of the user data that follows the header. IPv6s PL replaces IPv4s total length eld, which speci es the length of the header and data.

This 8-bit next header eld replaces IPv4s protocol eld. NH speci es the type of header that immediately follows the IPv6 header. NH enables extension headers to be inserted between the IP header and the TCP or UDP headers that precede user data. An example of this eld is the use of IPSecs authentication and encryption headers for security. This eld also effectively replaces IPv4s header length and option elds.

This 8-bit hop limit eld is used to specify the number of seconds a packet can remain active on the Internet. The value of this eld is decremented by 1 second each time it passes through a router. HL replaces IPv4s TTL eld.

SA This eld carries the 128-bit IP source address. Except for its length, SA has exactly the same meaning as IPv4s SA eld. Its location within the packet is different, though.

A This eld carries the 128-bit IP destination address. Except for its length, DA has exactly the same meaning as IPv4s DA eld. Its location within the packet is different, though.

FIGURE 3.23 Format and contents of an IPv6 header. The length of each field is given in bits. Source: Adapted from RFC 1752, RFC 1883, and RFC 2373.

- Support for auto-readdressing, which allows a packet to be automatically routed to a new address.
- Support for autoconfiguration of network addresses. IPv6 hosts can acquire their network address dynamically. This is done via a "plug and play" method or through full support for the Dynamic Host Configuration Protocol (DHCP).
- Support for data authentication, privacy, and confidentiality. (See Chapter 16.)
- Support for priority routing. This enables a source node to assign a delivery priority level to its outgoing packets. This enables "real-time" packets (e.g., packets carrying full-motion video data) to be sent at a constant rate without interruption of delivery.

Although these features might sound compelling, the march to convert from IPv4 to IPv6 has slowed somewhat because of the improvements made to IPv4. Thus, many believe that the only real incentive for deploying IPv6 is the worldwide demand for Internet addresses.

92. What's an IPv6 address look like?

An IPv6 address is quite different in appearance than an IPv4 address. IPv6 addresses, like IPv4 addresses, are still grouped into classes. The delimiter, however, is no longer the familiar "dot" notation. IPv6 addresses are expressed in hexadecimal form and written as eight 16-bit integers. Also, instead of IPv4's familiar dot notation, IPv6 uses a colon as its delimiter. An example of an IPv6 address is

2A01:0000:0000:0000:12FB:071C:04DE:689E

To reduce the complexity in writing these address, leading 0s in a hex group can be eliminated. Thus, the sample address can be expressed as 2A01:0:0:0:12FB:71C:4DE:689E. Frequently, many IPv6 addresses also contain contiguous strings of 0 bits. For example, IPv6 support of IPv4 address formats involves placing 0s in either the higher-order 80 or 96 bits of an IPv6 address. (More on this later.) In such cases, a double colon can be used to designate multiple 0 groupings. Thus, the sample IPv6 address just given can be expressed in a reduced form as 2A01::12FB:71C:4DE:689E. In this example, the double colon denotes three hex groupings of 0000. The double colon notation can only be used once in an address. It also can be used to designate leading or trailing 0s of an address.

As noted in the preceding paragraph, IPv6 provides support for IPv4 address formats. Such addresses are referred to as "IPv6 addresses with embedded IPv4 addresses." Two different formats are available. The first consists of 96 higher-order 0 bits followed by the 32-bit IPv4 address. A convenient format for expressing this is to mix the colon and dot delimiters. For example, the IPv4 address, 206.43.152.78, is expressed in IPv6 form as 0:0:0:0:0:0:206.43.152.78. This address can be further reduced by using the double colon delimiter to represent the leading zeros of the address. Thus, an equivalent form is ::206.43.152.78. This address is formally called an IPv4-compatible IPv6 address and is used to represent nodes that support IPv6 but must route IPv4 packets over an IPv6 network. A second format consists of 80 higher-order 0 bits, followed by one hex grouping of Fs, followed by the IPv4 address. Thus, our sample IPv4 address is expressed in this form as ::FFFF:206.43.152.78. This address type is called an IPv4-mapped IPv6 address and is used to represent IPv4 nodes that do not support IPv6.

The first field of an IPv6 address is a variable-length format prefix, which specifies the various categories of addresses. Examples of some prefixes and their meanings follow:

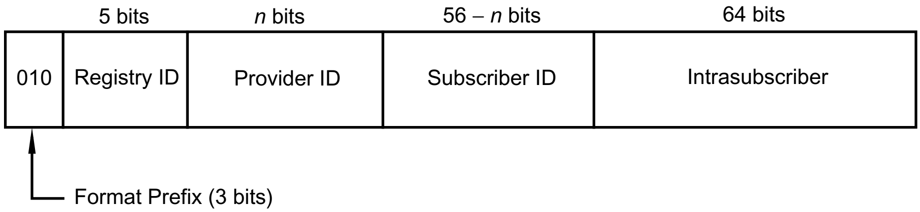
- **010** designates an IPv6 Provider-Based Unicast address, which is assigned to nodes connected directly to the Internet. The general structure of an IPv6 provider-based unicast address is shown in Figure 3.24.
- **1111 1110 10** designates a Link-Local address, which is assigned to isolated network sites. An isolated site is one in which nodes are connected to a network, but the network does not have a router and is not connected to the Internet.
- **1111 1110 11** designates a Site-Local address, which is assigned to sites that have a router connection but are not connected to the Internet.
- **1111 1111** designates a Multicast address.
- **0000 0000 0000 0000 0000 0000** designates an embedded IPv4-compatible IPv6 address.

Additional information about the IPv6 addressing architecture can be found in RFC 1884, RFC 2073, RFC 2373, and RFC 2374.

93. How will the Internet convert from IPv4 addressing to IPv6? It sounds like this is much more than a simple upgrade. Surely, this cannot occur overnight.

You're right. IPv6 is more than an IPv4 upgrade; it's a completely new version of the Internet Protocol. Its addressing is different, its headers are more specialized, it provides for more options including flow control and security, and it supports host mobility autoconfiguration along with some other new features. Converting to IPv6 cannot occur overnight. It is unlikely that every Internet user will switch to IPv6 on a predetermined date and time. Can you image the entire Internet world being in sync and pressing the Return key at the same time? Hardly. Instead, the Internet must slowly and methodically migrate from IPv4 to IPv6 over several years.

Migrating to an IPv6 Internet without disrupting the current operation of the existing IPv4 network is a real issue. It is imperative that IPv6 and key functions of IPv4 interoperate during the transition period. The current IPv6 migration plan supports several types of transitions. First, incremental upgrades will be permitted. This means that any IPv4 device can be upgraded to IPv6 independent of any other networked device, with one exception—the DNS server must first be upgraded to handle IPv6 address records. Second, incremental deployment will be supported. This means that new IPv6 nodes can be installed at any time without any modifications necessary. Third, the transition must support current IP addresses. Thus, if an existing IPv4 node is upgraded to IPv6, it does not have to be assigned a new IPv6 address. Rather, it can continue using its existing IPv4 address. Finally, there should be little or no preparation work needed on the part of a network administrator to either upgrade a device from IPv4 to IPv6 or deploy a new IPv6 device. Overall, this transition is to be processed in two phases. At the end of Phase 1, both IPv4 and IPv6 devices will coexist on the Internet; at the end of Phase 2, only IPv6 devices will exist. During Phase 1, it is believed that IPv6 and IPv4 nodes will interoperate, new IPv6 devices will be deployed in an incremental fashion with few interdependencies, and end users, system administrators, and network operators will be capable of making the transition without too much difficulty.



egi tr Identifies the organization responsible for assigning Internet addresses (known as the Internet registry). For example, the North American registry (ARIN) has ID 11000, and the European registry (RIPE NCC) has ID 01000. The registry assigns the provider ID.

Pro i er Identifies the Internet service provider (ISP). The specific length of this field is a function of the registry that assigns this part of the address. In one configuration, this can be from 16 to 24 bits. The ISP assigns the subscriber ID.

Su cri er Identifies a unique user. The specific length of this field depends on the provider s strategy for structuring and assigning subscriber IDs. In one configuration, this can be from 24 to 32 bits.

ntra u cri er Identifies the local part of the address (i.e., the local network and host). This field can be partitioned into a subnet field and host/interface ID field. For example, in an Ethernet network, the higher-order 16 bits of this field can be used to specify a subnet, and the remaining portion can be assigned the interface s 48-bit Ethernet/802.3 MAC address.

FIGURE 3.24 General structure of an IPv6 provider-based unicast address. Source: Adapted from RFC 2050, RFC 2073, and RFC 2374.

To aid network administrators in their renumbering efforts, an organization called Procedures for Internet Enterprise Renumbering (PIER) is available. Information about PIER can be found at http://www.ietf.org/proceedings/9bdec/charters/pier/charter.html Additional information about IPv6 renumbering issues is also available from RFC 1916, RFC 2071, and RFC 2072.

94. I have one more innocuous question relative to IPv6. What happened to IPv5?

Version 5 of IP (IPv5) is an experimental Internet protocol designed for real-time, parallel data transmissions. So, although it appears that the Internet Engineering Steering Group (IESG) skipped a version number when it assigned the number 6 to the new protocol, in reality this was not the case.

95. What type of services or resources does the Internet provide?

One of the main attractions of the Internet is the services and resources that are available from it. Some of the services include e-mail, remote logins, file transfers, network news—an electronic forum that consists of special interest groups and discussions (there are currently over 35,000 network news groups that cover a very wide and diverse number of topics), search tools or “engines” that allow a user to locate specific information based on user input, communication resources such as Instant Messaging and Internet Relay Chat (IRC), interactive games, and web browsers that enable you to view resources that have been formatted as hypertext files. There is streaming audio and video, which enable users to listen to audio recordings or view videos in real-time. There are also programs that enable two-way interactive videoconferencing to take place via the Internet. Through these services users can acquire information about nearly anything.

There are also several high-profile Internet services or resources available. These include electronic commerce (e-commerce), Voice Over IP (VoIP), and Virtual Private Networks (VPNs). E-commerce involves using the Internet for credit card purchases of items such as automobiles, airline tickets, computer-related products, and books; VOIP enables users to place telephone calls across the Internet; and VPNs enable organizations to establish private interconnected corporate LANs using the Internet (see Chapter 7).

96. With all of these different services and resources available over the Internet, the issue of security emerges. How secure is the Internet?

Internet security is undoubtedly of paramount concern for users. Unfortunately, the TCP/IP protocol suite was not initially designed with security in mind. This was not an oversight on the part of the original designers of TCP/IP. Remember: TCP/IP was initially developed to serve the research and academic communities to facilitate the exchange of research and scholarly activities. Inherent in this academic endeavor was a presumption of trust and honesty. Also, many of the compromises of TCP/IP protocols today were not anticipated by the TCP/IP designers 20 years ago.

The first major security breach of the Internet occurred on November 2–3, 1988, when a student exploited a security “hole” in the Simple Mail Transfer Protocol (SMTP). Now known as the “Worm incident,” many of the computers connected to NSFNET at that time were affected and rendered useless. Since then, there have been many attempts (some successful, some not) to exploit known weaknesses in other TCP/IP protocols, including *denial of service attacks* (see Chapter 16) as well as e-mail viruses. There also is no shortage to the number of individuals who have nothing better to do than search for creative ways to compromise a system.

In response to these attacks, several approaches are available. The easiest thing to do is not connect a system or network containing critical data to the Internet. A second strategy is to encrypt sensitive data prior to transmission across the network. A third approach is to install filters on routers that either deny or permit certain traffic to enter your network. Alternatively, special-purpose firewall devices that serve as buffers can also be installed between your network and the outside world. On the protocol front, there is Secure HTTP (https) for protecting web transactions, e-mail security is available via Secure MIME (S/MIME) and Pretty Good Privacy (PGP), and several protocols have been developed to help secure

VPNs, including the Point-to-Point Tunneling Protocol (PPTP), Layer 2 Forwarding (L2F), Layer 2 Tunneling Protocol (L2TP), and IP Security (IPSec).

It should also be noted that all the hardware based or protocol-software based protection in the world can easily be undermined by irresponsible users. As a result, it is important that organizations establish and enforce network security and acceptable use policies, educate users about network security issues, and employ common sense practices such as don't give unauthorized people your password. See Box 16.3 for additional suggestions. Finally, several resources are available online that are specifically geared to Internet security. The best place to begin is the Computer Emergency Response Team (CERT), located at http://www.cert.org. The subject of network security, including types of attacks, hardware strategies for dealing with these attacks, and security-related protocols such as IP Security (IPSec), is discussed in Chapter 16.

END-OF-CHAPTER COMMENTARY

In this chapter, we discussed various aspects of the Internet, as well as an overview and brief history of TCP/IP and its relationship to the Internet. Various TCP/IP concepts also were presented. These included network layer, transport layer, and application layer protocols such as IP, TCP, UDP, SMTP, MIME, POP, IMAP, TELNET, FTP, and HTTP. We also discussed Internet addressing schemes, including IPv4 and IPv6, subnetting, and name resolution. Some of the information in this chapter is linked to other chapters, and many of the examples used throughout the book are presented from an Internet perspective. These include Chapters 1, 2, 7, 12, 13, 16, and 17. Chapter 7 contains a discussion of subnetting, routing, and virtual private networks that use the Internet. Security issues, including IPSec, are presented in Chapter 16. Finally, in Chapter 17, we examine various convergence technologies including voice over IP (VOIP). You are encouraged to review these chapters to gain additional information about the Internet and its related protocols.

Chapter 4

Physical Layer Concepts

In this chapter we present an overview of the first layer of the OSI model, namely, the physical layer. This is the "touch-and-feel" layer. It provides for the physical transmission of data. As part of our presentation we discuss various transmission media, including twisted pair, coaxial, and fiber-optic cables. We also examine various forms of wireless and satellite communications. An outline of the major topics we discuss follows:

- Physical Layer Issues (Questions 1–8)
- Analog vs. Digital Communications (Questions 9–22)
- Bandwidth vs. Throughput and Data Rate vs. Baud Rate (Questions 23–27)
- Noise (Questions 28–30)
- Shannon's Limit (Questions 31–38)
- Multiplexers and Multiplexing Strategies (Questions 39–43)
- Switching Strategies (Questions 44–45)
- Physical and Electrical Characteristics of Wire (Questions 46–54)
- UTP, STP, and IBM Cable (Questions 55–65)
- Coaxial Cable (Questions 66–69)
- Fiber-Optic Cable (Questions 70–81)
- Wireless Media (Questions 82–96)
- Satellite Communications (Questions 97–104)

1. What is the physical layer?

The physical layer is the lowest layer (layer 1) of the OSI Reference Model. The OSI model was discussed in Chapter 2.

2. What does the physical layer do?

Before sending data on the network, the physical layer on the local node must process the data stream, translating *frames* received from the data link layer (layer 2) into electrical, optical, or electromagnetic signals representing 0 and 1 values, or bits. Frames, which are discussed in more detail in Chapter 5, are specially formatted bit sequences that incorporate both data and control information. The local physical layer is responsible for transmitting these bit sequences through the network medium to the physical layer of the remote node, where frames are reconstructed and passed to the data link layer.

3. If you were to use one word to describe the physical layer what would it be?

Wire. Or, if you are from the state of Texas (U.S.A.), as is one of the authors, it's called "wharr."

4. Wire?

Yup. Actually, it's a little more than just wire, but wire gives you an idea of what we are dealing with when we speak about the physical layer. You see, all aspects related to a transmission medium used for data communications, including both wired and wireless environments, are defined by physical layer protocols and specifications. These include the type of cable and connectors used, the electrical signals associated with each pin and connector—called *pinouts* or *pin assignments*—and the manner in which bit values are converted into physical signals.

5. Can you give me an example?

Sure. In fact, we'll give you two. For the first example, consider a local area network. The physical layer of this network (abbreviated PHY in the documentation) defines, among other things, the type of cable permitted, the type of connectors we can use to attach the network cable to hardware devices, cable length restrictions, and the type and level of termination. A second example of a physical layer specification is the EIA RS-232C standard, which defines the electrical and physical characteristics used in serial communications. RS-232C specifies a 25-pin data bus (DB) connector that serves as an interface between a computer, referred to as the data terminal equipment, or DTE, and a peripheral device such as a modem or printer, referred to as the data communications equipment, or DCE. A later version of the RS-232C standard is RS-423, which defines a 9-pin DB connector. DB-9 connectors implement some of the signals from DB-25 and are used on IBM-PC or compatible microcomputers as a rear panel space-saving measure. The pinouts for the DB-25 and DB-9 connectors are shown in Figure 4.1.

6. Does the physical layer have anything to do with physics?

Why, yes it does. What gave it away?

7. Because the word "physical" has as its root the word "physic." I think I know where we're going with this, but before we get there let me just say that it's been a while since I studied physics, so please be gentle.

We will. We'll avoid quarks, photons, and the Theory of Relativity. However, you need to know a little bit about one simple, physical concept: *Harmonic motion*—and it has nothing to do with Nostradamus.

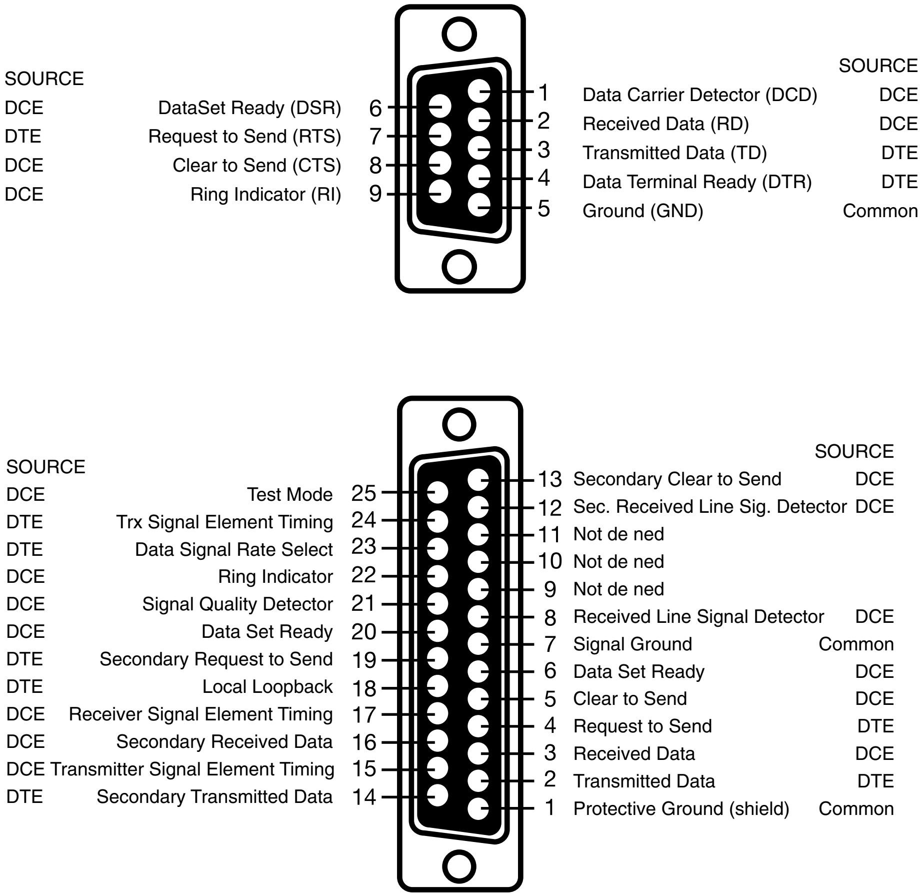


FIGURE 4.1 Pinouts of connectors used in RS-232-C serial communications. The DB-9 connector is frequently used for the low-speed serial port(s) on personal computers. The DB-25 connector is frequently encountered in modems. The numbers correspond to the pins of the connector; the accompanying text describes the signals associated with each pin. The column labeled Source refers to the signal source, either the data terminal equipment (DTE) unit (i.e., the computer) or the data communications equipment (DCE) unit (i.e., a peripheral device such as a modem).

8. What's harmonic motion and why do I have to learn about it?

We'll answer the second part of this question first. Don't worry. It's also called *simple harmonic motion* because it is a very simple concept. You need to be familiar with the concept of harmonic motion because it will help explain *analog communication*. Analog communication is used in conventional telephones (both landline and cellular), modems, fax machines, cable television, and lots of other devices and network services.

9. Tell me more about analog communication.

The term *analog* refers to any physical device or signal that can continuously vary in strength or quantity, for example, voltage in a circuit. The term *analog communication* refers to any method of communication based on analog principles. Typically, this term is associated with voice transmission instead of data transmission because voice transmission facilities, such as the telephone, were initially analog-based. The "other" type of physical communication is *digital communication*, which we'll talk about later.

10. What does harmonic motion have to do with analog communication?

To answer this question, consider an object attached to a spring that is suspended from a ceiling (Figure 4.2). If you pull on the attached weight and release it, the spring begins oscillating up and down. In a frictionless environment, this up and down motion would continue forever. This idealized motion is called simple harmonic motion, which is the basic model for vibratory or oscillatory motion and can occur in many different types of wave motion. Examples include mechanical oscillators such as mass-spring systems similar to that shown in Figure 4.2, and pendulums; periodic motion found in the earth sciences such as water waves, tides, and climatic cycles; and electromagnetic waves such as alternating electric currents, sound waves, light waves, radio waves, and television waves.

11. Okay, but what does this have to do with analog communication?

In any computer communications system, data are transmitted across a medium from sender to receiver in the form of electrical signals. In analog communications, signals flow across a wire in the form of electromagnetic waves. When viewed by an oscilloscope, these signals appear as continuous waves called sinusoidal waves, which resemble a sine curve (Figure 4.3). Sinusoidal waves are characteristic of anything that oscillates, and they have the following three attributes: *amplitude*, which is the level of voltage on a wire (or the intensity of a light beam when dealing with fiber-optic cable); *frequency*, which is the number of oscillations, or cycles, of a wave in a specified length of time; and *phase*, which is the point a wave has advanced within its cycle. A frequency rate of one cycle per second is defined as 1 hertz (abbreviated Hz) in honor of Heinrich Rudolph Hertz (1857–1894), a

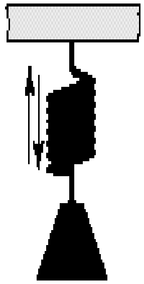


FIGURE 4.2 An object is attached to a spring that is suspended from a ceiling. When pulled and released, the spring oscillates up and down. This oscillation is called simple harmonic.

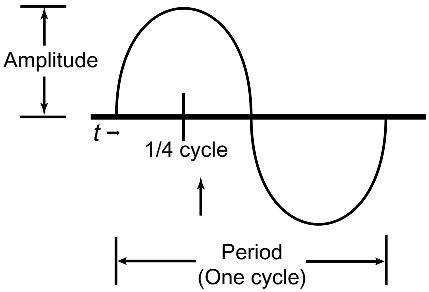


FIGURE 4.3 One cycle of a sine curve.

German physicist who in the late 1880s was the first to produce radio waves artificially. Thus, hertz is a measure of frequency in cycles per second. The reciprocal of frequency is called *period*, which is the amount of time it take to complete a single cycle, that is, seconds per cycle.

12. Can you bring this down to earth for me?

Sure. Consider the act of speaking over the telephone. If we speak softly or whisper, the amplitude (volume) decreases. If we speak loudly or scream, the amplitude increases. If we speak in a high-pitched voice, the frequency changes to more cycles per second than if we speak in a low-pitched voice, which requires fewer cycles per second (see Figure 4.4). AM/FM radio, television speakers, public address systems, and most important of all, traditional telephones, are all examples of analog devices (although there is a fast-growing trend toward a full-digital telephone system both in business and wireless networks).

13. How does this relate to data communications?

In data communications, data are represented in analog form by varying the voltage of the wave (called amplitude modulation, abbreviated AM), by varying the frequency (called frequency modulation, abbreviated FM), or by varying the phase (called phase modulation or phase shifting) of a wave. Phase modulation is often used for modems, which is discussed in Chapter 15.

14. How does the term "wavelength" relate to this discussion?

Wavelength, as the name implies, is a measure of the length of a wave. It is the distance an electrical or light signal travels in one complete cycle. Radio signals are often described and classified according to their wavelength. For example, "the 40-meter ham band" is an explicit reference to radio waves that are approximately 40 m long. In the RF spectrum (discussed later in this chapter), "short-wave radio" and "microwave radar" are relative references to different wavelengths.

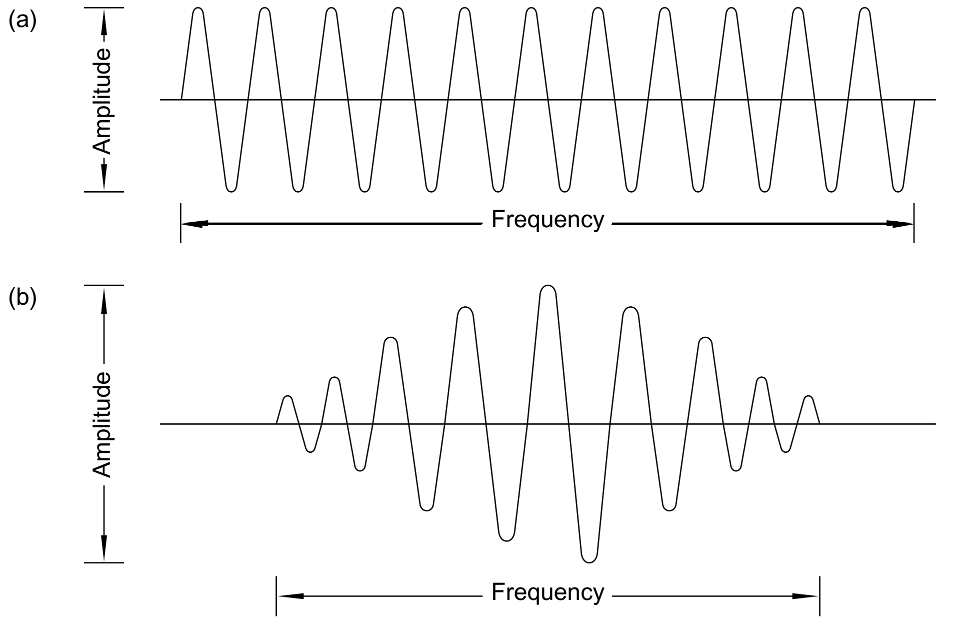


FIGURE 4.4 When represented in electrical form, sound waves produced by the human voice are continuous in nature. These waves vary in strength (called amplitude) and vary in the number of times they fluctuate over time (frequency). For example, the wave in (a) has less amplitude and greater frequency than the wave in (b). When we speak louder, amplitude increases. When we speak in a high-pitched voice, frequency increases.

For light signals, which are used in fiber-optic transmissions, wavelength and frequency are inversely related—as one increases, the other decreases. This relationship is expressed using the formula, $\lambda = \frac{c}{f}$, where λ is equal to the wavelength in meters, *c* is the speed of light, 3×10^8 meters per-second, and *f* is the frequency in hertz. The concept of wavelength will be elaborated on later in the chapter.

15. If analog communication is more often associated with voice transmission, then presumably digital communication must be used for data transmission.

That's correct. The term *digital* refers to any physical device or signal that is coded in a binary form. The term *digital communication* refers to any method of communication based on digital principles. A binary code is a system that uses the two symbols 0 and 1 to represent data. A single symbol, 0 or 1, is referred to as a binary digit, which is commonly called a bit (a contraction of the words *bi*nary and digi*t*). An example of digital data is binary coded text, which consists of letters of the alphabet and numerical data (collectively called alphanumeric data), as well as special character symbols such as $, @, <, and *. The assignment of a binary notation to these data is called a character code. One common code is the American Standard Code for Information Interchange (ASCII). For example, the ASCII representations of the digit 5, the lowercase letter c, and the special character @ are, respectively, 00110101, 01100011, and 01000000. Another example of digital data is graphics, which are produced using specially designed software packages. Some packages transform numerical data into charts or graphs; other packages code the position of a point or points into standard *xy* or *xyz* coordinates for processing. A third example is digital photography in which photographs are captured and saved in digital format instead of using conventional analog methods. Today, nearly every type of signal can be converted into a digital format (i.e., as 0s and 1s), including traditional analog signals such as voice and video.

In digital communications, signals are called *discrete*. That means that they are binary in nature—off or on, one or zero. What makes a signal discrete is that there is no in-between. This is similar to a light bulb—either it is off or it is on—there is no "sort of on" or "sort of off" (although you could put a dimmer in the circuit, making the light bulb an analog device!). Therefore, we can conclude that a digital signal consists of two (and only two) states: electrical current applied, or no current at all. On most systems, if power is applied, it is considered "on" and is usually interpreted as "1" by a computer. If there is no power, then we have an "off" state and this is interpreted as "0." This is called *binary interpretation*, and the "on" and "off" states are interpreted as bits 1 and 0, respectively. Figure 4.5 shows a typical digital waveform as measured over some time interval. Digital communication is more difficult to implement over airwaves (wireless), but it provides a much more reliable communications environment with much less noise and distortion than an analog connection method.

16. So, I create a digital signal by merely applying a current or turning it off? Is it that simple? Does the voltage matter?

Yes, voltage matters, but life in the real world is not that simple. Each type of digital circuit has a particular specification for which a *range of voltages* represents a 0 and another represents a 1. This is necessary because of real-world factors such as electrical noise, cable resistance, and differences in ground potential between the transmitter and the receiver. We'll get deeper into some of these factors later, but for now, consider an example involving RS-232C (also called EIA-232), a standard serial interface we introduced earlier. In RS-232C, a 1 is represented by any voltage between −5 and −15 volts and a 0 is represented by any voltage between +5 and +15 volts (see Figure 4.6). Let's say that the transmitter sends a 0, for which it uses 12 volts. By the time it gets to the receiver, however, the potential might be reduced to 10 volts because of the electrical resistance of the

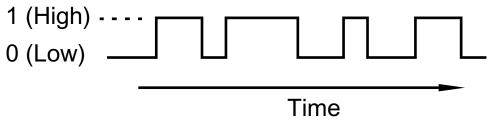


FIGURE 4.5 Example of a digital signal waveform.

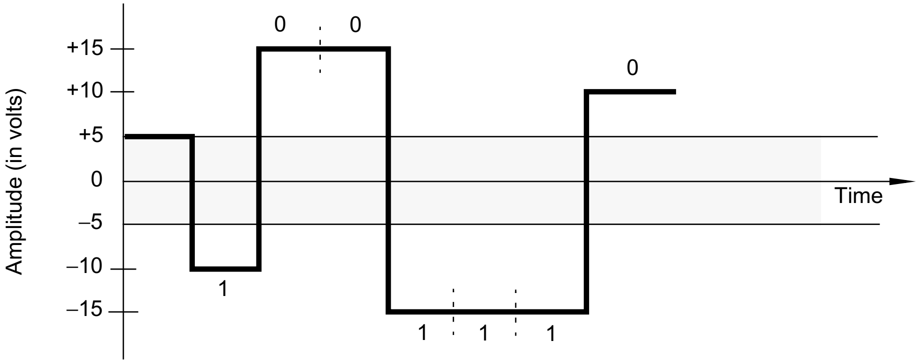


FIGURE 4.6 In RS-232, voltage that varies between −15 and +5 volts is interpreted as a binary 1, and voltage that varies between +5 and +15 volts is interpreted as a binary 0. Voltage that varies between −5 and +5 volts is ignored; that is, no interpretation is made.

wire. Nevertheless, the receiver interprets the signal as a 0 because it falls between +5 and +15 volts. A less technically-oriented example is a game of darts. You get credit for a bulls-eye if your dart hits anywhere within the appropriate circle. If it were required to hit a single point for a bulls-eye, there would be far fewer bulls-eyes. Thus, the "engineers" who designed the game of darts took into account some real-world factors.

17. In your RS-232C example, there is a big "hole" from –5 to +5 volts. Why is this?

This gap provides for random electrical signals, called *noise*, which can create false signals in this range. (The concept of noise is discussed later in this chapter.) When engineers design communication circuits, they must take into account the possibility of noise interfering with communications because in the real world of digital communication signaling, things are not quite as simple as "on" and "off."

18. What happens if the receiver detects, say, +3 volts.

Nothing.

19. Nothing?

Nothing. *Nada, rien, bupkas,* zip, zilch. The receiver "sees" neither a zero nor a one.

20. What happens if the receiver gets more than 15 volts?

This is worse, because both the transmitter and receiver will probably fail completely. At this point, all is lost, and 1s and 0s don't matter anymore. Such destructively high voltages might be the result of a nearby lightning strike, a component failure at either end, or a high-voltage power line short-circuiting to the communication cable.

21. What else do I need to know about analog vs. digital communications?

For one thing, there is often some confusion about the "speed" or capacity of a communications channel and the terms used to describe it.

22. Such as?

Well, for starters, how about bandwidth?

23. Okay. What is bandwidth?

In analog communications, *bandwidth* is the total capacity of a communications channel. It is the difference between the highest and lowest frequencies capable of being carried over a channel. The greater the bandwidth, the more signals that can be carried over a given frequency range. For example, typical voice-grade lines transmit frequencies from 300 Hz to 3300 Hz. Thus, the bandwidth is 3300 Hz – 300 Hz = 3000 Hz, or 3 kilohertz (kHz).

In digital communications, bandwidth refers to *data rate*, which is the amount of data that can be transferred over a communications medium in a given period. Data rate is measured in bits per second (bps) and can vary considerably from one type of channel to another. For example, LANs have data rates ranging from 4 million bits per second (referred to as megabits per second and abbreviated Mbps) to 1000 Mbps; the bandwidth of dialup connections using modems ranges from 300 bps to 33,600 bits per second (33.6 kbps) or 56 kbps; and the bandwidth of WANs that use high-speed circuits can range anywhere from 1.5 Mbps to 45 Mbps to 622 Mbps and higher.

24. You say we measure data rate in bits per second. Isn't this the same as baud rate?

No. There is a difference. A *baud* is a unit of signaling speed, named after the French engineer Jean Maurice Emile Baudot (1845–1903). It is another term used to express the capacity of a channel, but it is, indeed, different from bits per second. The speed in baud is equal to the number of times the line condition (i.e., frequency, amplitude, voltage, or phase) changes each second. At low speeds (under 300 bps) data rate (measured in bps) and baud rate are the same because signaling methods are relatively simple. As speed increases, signaling methods become more complex. Baud rate then differs from data rate because several bits are typically encoded per baud. That is, each signal can represent more than one bit of information.

25. I need an example.

OK. Consider a communications channel transmitting at 2400 baud. This means that the signaling rate of the channel is changing 2400 times per second. If each signal is used to represent one bit, then the baud rate is equal to the data rate—the data rate is 2400 bps. However, if each signal represents four bits, then the baud rate remains at 2400, but now the data rate is $4 \times 2400 = 9600$ bps. Thus, the channel's bandwidth is 9600 bps.

26. So you are saying that a baud rate of *x* does not always equal *x* bps.

Right you are! If each signal event represents exactly one bit, then the baud rate is equal to the data rate. However, since this is rarely the case in data communications, you should not consider baud rate and data rate synonymous. Unless otherwise noted, we will always express bandwidth in terms of bps or bits per second or as a multiple of bits per second such as: kilobits per second (kbps), which is 1000 bps; megabits per second (Mbps), which is 1 million bps; and gigabits per second (Gbps), which is 1 billion bps.

27. While we're at it, is there a difference between bandwidth and throughput? I often hear these two terms used interchangeably.

When applied to computer communications and networking, bandwidth represents the theoretical capacity of a communications channel expressed in bits per second. To understand the difference between bandwidth and throughput, let's assume the network we use is a Fast Ethernet LAN. (*Note:* We discuss Fast Ethernet in Chapter 8.) Fast Ethernet has a maximum transfer rate of 100 Mbps. Does this mean we can expect all data transfer rates to be at 100 Mbps? No. Extraneous factors such as a node's processing capability, input/output processor speed, operating system overhead, communications software overhead, and amount of traffic on the network at a given time all serve to reduce the actual data rate.

Consequently, there is a difference between the maximum theoretical capacity of a communications channel and the actual data transmission rate realized. This "reality rate" is known as *throughput*, which refers to the amount of data transmitted between two nodes in a given period. It is a function of hardware or software speed, CPU power, overhead, and many other items. Summarizing, bandwidth is a measure of a channel's theoretical capacity; it describes the amount of data a channel is capable of supporting. Throughput, on the other hand, informs us of what the channel really achieves. Just because a medium or LAN architecture is specified to operate at a certain data rate, it is not valid to assume that this rate will be the actual throughput achieved on any given node or group of nodes.

28. What else do I need to know about what bandwidth and throughput?

Well, a related issue that is tied to bandwidth and throughput is *noise*. We have mentioned it several times so far but have yet to discuss it formally.

29. Noise? You mean like loud, senseless shouting, or a crying baby in a restaurant?

OK, we'll be precise. We mean *electrical* noise. (You're not that far off, though.) In the context of computer communications and networking, noise is any undesirable, extraneous signal in a transmission medium. It occurs in two forms—*ambient noise* and *impulse noise*. Ambient noise, also called *thermal noise*, is always present and is generated primarily by transmission equipment like transmitters, receivers, and repeaters. Ambient noise also can be induced by external sources such as fluorescent light transformers, electrical facilities, heat, and, in fact, the background radiation from the Big Bang. If ambient noise is present, receiving equipment can have problems in distinguishing between incoming signals. Impulse noise consists of intermittent, undesirable signals induced by external sources such as lightning, switching equipment, and heavy electrically operated machinery such as elevator motors and photocopying machines. Impulse noise increases or decreases a circuit's signal level; this causes the receiving equipment to misinterpret the signal. Whichever the type or source, noise degrades the quality and performance of a communications channel and is one of the most common causes of transmission errors in computer networks. Although some noise is always present, much of it can be avoided through proper cable installation.

30. That was interesting. Are you going to make noise about anything else?

Yes, there's one more type of noise we'll touch on briefly: *intermodulation noise*. Frequency division multiplexing, a concept we will discuss later, mixes multiple frequencies of data transmission on a single transmission medium. Intermodulation noise occurs when two of the frequencies interact to produce a phantom signal at a different frequency, which can be either the sum or the difference of the two original frequencies. For example, let us assume for simplicity that, in a frequency division multiplexing environment, a coaxial cable carries three different signals, at frequencies f_1, f_2, and f_3. Intermodulation noise can occur if f_3 is equal to the sum or difference of f_1 and f_2. Either way, the spurious signal at frequency f_3 can interfere with the transmission of valid data at that frequency.

31. That was a little esoteric, but I'm glad you touched on it. Are we done with noise?

Not yet. There is one more noise-related subject we need to address: Shannon's limit.

32. Okay. You caught my interest. I used to have a cat named Shannon. So what's Shannon's Limit?

We like cats, too, but the only relationship between felines and noise involves wailing related to their hunger and sex drives, not networking. Shannon's limit is a mathematical theorem, named for the mathematician who derived it, Claude Shannon, that describes a model for determining the maximum data rate of a noisy, analog communications channel. Shannon's limit is given by the formula below. (*Note:* A second theorem, Nyquist's theorem, which is presented in Box 7.1, determines the maximum data rate of a channel for noiseless environments.)

$$\text{Maximum Data Rate } (MDR) = H \log_2(1 + \left(\frac{S}{N}\right))$$

- MDR is given in bits per second (bps)
- H = bandwidth in hertz (Hz)
- $\left(\frac{S}{N}\right)$ = a measure of the signal-to-noise ratio in decibels (dB)

33. Forget about it. You just lost me.

Hold on now. We think you'll find Shannon's limit fascinating once we do an example.

34. OK, but before you give me an example, tell me what this signal-to-noise ratio thing is.

The signal-to-noise ratio (abbreviated SNR) is a measure of signal quality expressed in decibels (dB), which is a measurement that quantifies the strength of a signal. (A decibel is one-tenth of a *bel*, which was named after Alexander Graham Bell and is used to compare electrical power levels and sound intensities.) SNR is the ratio of signal strength to background noise on a cable. More specifically, SNR is the ratio between the desired signal and the unwanted noise in a communications medium. In plain, twenty-first century English, it's how badly a line sucks.

35. All right. I'm ready for an example.

A good example of an application of Shannon's limit is modem speeds. During the mid-to-late 1990s, modem speeds increased from 14,400 bps to 28,800 bps, but then topped out at 33,600 bps. Conventional analog modems achieved a peak rate of 33,600 bps (or in some cases, 38,400 bps) because this was the maximum data rate possible for existing analog communications channels based on the channel's frequency and signal-to-noise ratio.

36. Wait a minute now. What about 56K modems? Aren't they capable of 56,000 bits per second?

Shannon's limit does not apply to 56K modems. You see, Shannon's limit applies only to analog channels. The communication channels used for 56K modems are hybrid analog–digital connections that consist of both analog and digital channels. More specifically, 56K modems require a digital link between an Internet service provider (ISP) and the telephone company; the link between a customer and the provider is analog. We explain this in Chapter 15. This configuration enabled modem designers to increase modem speeds to 56,000 bps. However, even this strategy could not circumvent the physical limitations of analog circuitry, and hence, 56K modems became the fastest type of conventional dialup modems available that involve analog channels.

37. So what you are saying is if my ISP does not have a digital link to the telephone company and I maintain my standard analog telephone link in my home, the best connection speed I will ever realize with my modem is 33.6 kbps?

That's correct. However, even with a top speed of 33.6 kbps or 38.4 kbps, many users very rarely achieve this speed. If there is any degradation of the line they are using, the modem cycles down (this is called *link negotiation*) to a lower, more reliable speed. For example, how many times have you actually made a consistent connection at 28.8 kbps or 33.6 kbps? Once again, we discuss this in much greater detail in Chapter 15.

38. Can you give me an example of how to use Shannon's Limit to do a calculation?

Yes. See Box 4.1. You might need to review some college algebra concepts (e.g., logarithmic functions) to fully understand and appreciate this example.

39. What else do I need to know about what happens at the physical layer?

Have you heard of multiplexing?

40. Does this have anything to do with a mux?

Yes it does. *Mux* is a casual abbreviation standing for *multiplexer*. What do you know about a mux?

BOX 4.1: Example of Shannon's Limit

Given $\text{dB} = 10\left[\log_{10}\left(\frac{s}{n}\right)\right]$:

- If $\left(\frac{s}{n}\right) = 10$, then $10\left[\log_{10}\left(\frac{s}{n}\right)\right] = 10[\log_{10}(10)] = 10(1) = 10$. Thus, SNR = 10 dB.
- If $\left(\frac{s}{n}\right) = 100$, then $10\left[\log_{10}\left(\frac{s}{n}\right)\right] = 10[\log_{10}(100)] = 10(2) = 20$. Thus, SNR = 20 dB.
- If $\left(\frac{s}{n}\right) = 1000$, then $10\left[\log_{10}\left(\frac{s}{n}\right)\right] = 10[\log_{10}(1000)] = 10(3) = 30$. Thus, SNR = 30 dB.

and so on.

Example: If $H = 3000$ Hz and the signal-to-noise ratio (SNR) is 30 dB, what is the MDR?

Solution: Note from above that 30 dB implies $\left(\frac{s}{n}\right) = 1000$. Using Shannon's Limit we have

$$\begin{aligned} MDR &= (H)\left[\log_2\left(1 + \frac{s}{n}\right)\right] \\ &= (3000)[\log_2(1 + 1000)] \\ &= (3000)[\log_2(1001)] \end{aligned}$$

At this stage we must now solve for $\log_2(1001)$. There are several ways in which this can be done. We can use a calculator that is capable of solving logarithms in base 2, we can use natural logarithms, or we can estimate the value. We will demonstrate the last two methods since we do not have a calculator capable of solving log functions in base 2.

Using Natural Logarithms

$$\begin{aligned} \log_2 1001 &= x \\ 2^x &= 1001 \\ \ln 2^x &= \ln 1001 \\ x \ln 2 &= \ln 1001 \\ x &= \frac{\ln 1001}{\ln 2} \\ x &\approx \frac{6.909}{0.6931} \\ x &\approx 9.967 \end{aligned}$$

Substituting 9.967 into the equation:

$MDR = (3000)(9.967) = 29{,}902$ bps

Using Estimation

$$\begin{aligned} \log_2 1001 &= x \\ 2^x &= 1001 \end{aligned}$$

$$\begin{array}{ll} 2^1 = 2 & 2^6 = 64 \\ 2^2 = 4 & 2^7 = 128 \\ 2^3 = 8 & 2^8 = 256 \\ 2^4 = 16 & 2^9 = 512 \\ 2^5 = 32 & 2^{10} = 1024 \end{array}$$

Note that $\log_2(1001)$ must be between 9 and 10 since the logarithm's argument (1001) is between $2^9 = 512$ and $2^{10} = 1024$. Since it is closer to 10 we estimate it to be 9.9. Substituting 9.9 into the equation:

$MDR = (3000)(9.9) = 29{,}700$ bps

As a result, the maximum data rate of a communications channel with these parameters is approximately 30,000 bps (30 kbps). Note that this is the upper limit and in practice will rarely be achieved on a consistent basis.

41. We have them at work. I think they have something to do with being able to split a single communications channel into multiple channels. Is this right?

Pretty much so. A mux (acronym for multiplexer) is a device that does multiplexing, which is a process that enables data from multiple transmission channels to share a common link. In its simplest form, multiplexing involves combining data from several relatively low-speed input channels and transmitting these data across a single high-speed circuit. At the receiving end, this multiplexed data stream is then separated (a process called demultiplexing) relative to the data's respective channels and delivered to the corresponding output facilities. This is depicted in Figure 4.7. Through multiplexing, many different transmissions are possible using a single medium. For example, a communication medium can be divided into separate channels with one channel transmitting data, another transmitting voice, and a third transmitting video. Each of these separate, independent transmissions can occur simultaneously.

42. I seem to recall there are different ways to do multiplexing. For example, something called *time domain multiplexing* comes to mind.

Right you are! Several multiplexing strategies abound, including frequency division multiplexing (FDM), time division multiplexing (TDM), statistical multiplexing, demand access multiplexing (DAM), wavelength division multiplexing (WDM), code division multiple access (CDMA), and inverse multiplexing. A brief overview and description of these strategies follow. (CDMA is a multiplexing scheme used in wireless communication, which is described in a later chapter.)

43. Can you give me a quick overview of the more popular multiplexing methods? I don't think it's necessary to spend a lot of time discussing them, but I do want to have some familiarity with them. So just give me the *Reader's Digest* version, please.

Let's take these methods one at a time:

***Frequency Division Multiplexing* (FDM)** This technique partitions the available transmission frequency range into narrower bands, each of which is a separate channel. The

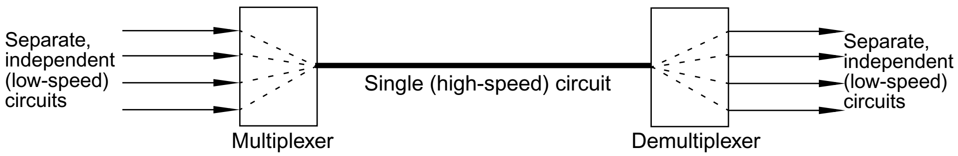


FIGURE 4.7 Multiplexing involves combining several separate (relatively low-speed) transmission facilities into a single (high-speed) communications channel for data transmission purposes. At the receiving side, this single channel is then demultiplexed into the corresponding channels. Multiplexing is performed via a multiplexer, or mux for short; demultiplexing is performed via a demultiplexer, or demux for short.

idea behind FDM is to divide the main frequency into appropriate subfrequencies with each subfrequency customized to the bandwidth of data that it must carry. This makes FDM very efficient and cost-effective. An example of FDM is the broadcast method used by television stations. The FCC allocates a range of frequencies called a channel for a station to use, and the station subdivides this band into various subchannels. One subchannel carries engineering information for the station's technical staff, a second carries the analog signal for audio reception at the television set, and a third subchannel carries the video signal. What a remote unit can receive depends on the frequency for which the unit has been configured. FDM-based transmissions are parallel in nature (Figure 4.8).

***Time Division Multiplexing* (TDM)** This technique enables more than one signal to be transmitted over the same channel but at different time intervals. Time division multiplexing (TDM) assigns to each node connected to a channel an identification number and a small amount of time (i.e., a time slot) in which to transmit (Figure 4.9). Unlike FDM-based transmissions, which are parallel in nature, TDM-based transmissions are serially sequenced. Thus, nodes take turns transmitting over the channel, with each time slot permanently assigned to a specific channel. The amount of time a node gets for data transmission is a function of the number of nodes competing for the channel, the order in which nodes are requested for information (called the polling order), and the *clocking interval* of the TDM device.

Statistical Multiplexing This method of multiplexing allocates part of a channel's capacity only to those nodes that require it (i.e., have data to transmit). This strategy permits a greater number of devices to be connected to a channel because not all devices necessarily

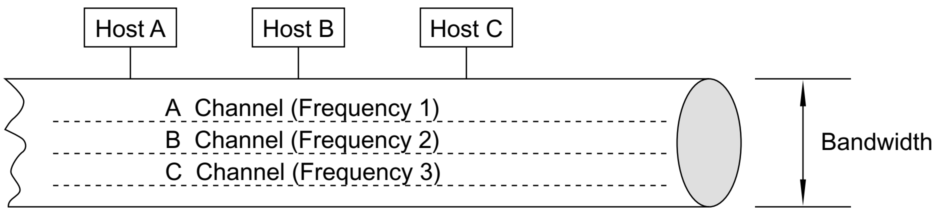


FIGURE 4.8 In frequency division multiplexing, the frequency of a communications medium is divided into subfrequencies, which are assigned to connected nodes, resulting in parallel transmissions.

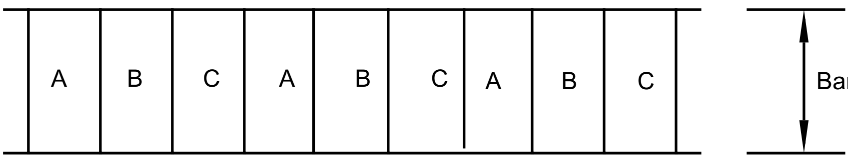


FIGURE 4.9 In time division multiplexing, the channel is partitioned into a sequence of time slots and each node is assigned a specific time slot in which to transmit.

require a portion of the channel at exactly the same time. A statistical multiplexer "senses" which input channels are active and then dynamically allocates bandwidth to these channels. Contrasting FDM and TDM with statistical multiplexing, note that in FDM and TDM a communications channel is partitioned into separate, fixed rate channels, which are not necessarily used all of the time. In statistical multiplexing, however, the channel is allocated to a device only when that device has data to transmit.

***Demand Access Multiplexing* (DAM)** In demand access multiplexing, a pool of frequencies is managed by a "traffic coordinator." The traffic coordinator assigns pairs of communications frequencies to a requesting station—one pair for transmission, a second pair for reception. This is the "demand" part—you demand a pair of frequencies, and if available, the traffic coordinator assigns them to you. The traffic coordinator then connects the two pairs of frequencies to another set of frequencies. This is the "access" part. When one or both stations are finished communicating, the allocated frequencies are deallocated and returned to the frequency pool, where they are made available for other incoming requests. This is the multiplexing part. DAM is similar to virtual memory allocation on computers. A "pool" of memory exists for all running processes. When a new process is started, memory is allocated from the pool. When the process is completed, the associated memory returns to the pool for use by another process. A major use of DAM exists in cellular communications.

***Wavelength Division Multiplexing* (WDM)** Wavelength division multiplexing is used with fiber-optic cables. In fiber-optic technology, electrical signals originating from a sending node are converted into optical signals using a light source such as a laser or a light-emitting diode (LED), which is a semiconductor device that converts electrical energy into light. (Fiber-optic issues are discussed later.) WDM involves the simultaneous transmission of these light sources over a single fiber-optic channel. The light sources, which are of different wavelengths, are combined by a WDM multiplexer and transmitted over a single line. En route to their destination, the wavelengths are amplified simultaneously by optical amplifiers. When the signals arrive, a WDM demultiplexer separates them and transmits them to their respective destination receivers. We give an illustration of WDM in Figure 4.10. WDM provides both cost and performance benefits. On the cost side, WDM saves money because it increases bandwidth without requiring the installation of additional fiber. The idea of minimizing or completely eliminating new fiber installations is attractive because installing new fiber can be expensive. On the performance side, WDM consolidates data from separate channels onto a single line. It also can reduce the number of optical-to-electrical conversions by implementing a strictly optical transmission method. A modification of WDM is *dense WDM* (DWDM) in which wavelengths are more closely spaced. This increases the number of channels the fiber can support.

Inverse Multiplexing Inverse multiplexing is the reverse of multiplexing. Instead of partitioning a single communication medium into several channels, an inverse multiplexer combines several "smaller" channels (i.e., low-speed circuits) into a single high-speed circuit. For example, through inverse multiplexing, two T1 circuits (1.544 Mbps) can be combined to form a 3-Mbps channel. Several Internet service providers use this strategy to offer their customers a larger "pipe" to the Internet. This technique is also sometimes generically called line aggregation.

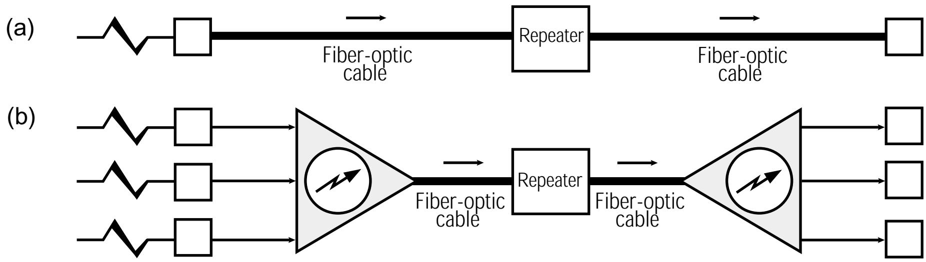


FIGURE 4.10 In a typical fiber-optic installation (a), a fiber-optic transmitter converts an electronic signal into light, which is sent through the fiber-optic cable to its destination. Repeaters, which are optical amplifiers, regenerate the light signal at appropriate points along the way. With wavelength division multiplexing (b), electronic signals originating from multiple data sources, each operating at different wavelengths, are combined into a wavelength multiplexer (WDM) and output onto a single fiber-optic cable. A demultiplexer (IWDM) is employed at the receiving end to split the signals. Source: Adapted from Clark, 1997.

44. How different is multiplexing from switching, which you discussed in Chapter 2?

As we indicated earlier, multiplexing is a technique used to place multiple signals on a single communications channel and is performed by a device called a multiplexer or mux for short. Switching, on the other hand, is a process that involves establishing an appropriate path, which a data message will follow as it travels throughout a network en route between a sending source and a destination node. Switching is performed by switches, which use specific predetermined criteria as the basis of path determination. Multiplexing and switching are the two basic techniques used for transmitting data within a communications network. They make it possible for transmission facilities to be shared among users in an efficient and economical manner.

45. What types of switching strategies are there?

If you review Chapter 2, we provided you with some basic information between the two primary strategies, namely, *circuit-switching* and *packet-switching*. For convenience, a short summary follows.

Circuit-Switching In a circuit-switched network, a dedicated physical circuit must be established between the source and destination nodes before any data transmission can take place. Furthermore, this circuit must remain in place for the duration of a transmission. The public telephone system, known formally as the public switched telephone network (PSTN), is a good example of a circuit-switched network. When we dial a telephone number, a switch that resides at the telephone company's central office establishes a logical connection to a set of wires based on the number we dialed. This set of wires will either connect to a wire center (also called a frame) that services a particular area, or it will connect to another central office that contains yet another switch. Ultimately, a circuit will be established that connects the caller's and receiver's telephones. One feature of circuit-switching is that it promotes link sharing. Circuit-switching enables different data transmissions (i.e., different sender–receiver pairs) to use the same communications channels. There is one caveat, though. Sharing cannot occur at the same time because, during a particular transmission, the communications channel is reserved exclusively for that specific sender–receiver pair.

Packet-Switching In a packet-switched network, instead of using a dedicated physical circuit for every node-to-node communication, nodes share a communications channel via a virtual circuit. A virtual circuit is a nondedicated connection through a shared medium that gives the high-level user the appearance of a dedicated, direct connection from the source node to the destination node. A virtual circuit is created by multiplexing a physical link so that the physical link can be shared by multiple network programs or data transmissions. This concept is extremely valuable for providing low-cost communications capabilities because it is very expensive to provide dedicated links for every data transmission, as in circuit-switched networks. A definition of virtual has been memorably coined in this way: "If you can see it and touch it, it's physical; if you can see it but can't touch it, it's virtual; if you can't see it and can't touch it, it's gone."

In a packet-switched network, messages are partitioned into smaller messages called packets, which may contain only a few hundred bytes of data, accompanied by addressing information and sequence numbers. A packet represents the smallest unit of data that can be transferred via a given network. Packets are sent to the destination node one at a time, at any time, and not necessarily in a specific order. The network hardware delivers the packets through the virtual circuit to the specified destination node, which is responsible for reassembling them in the correct order. Unlike circuit-switched networks, where a dedicated link is established a priori, every packet in a packet-switched network must carry the destination node's address. In a circuit-switched network, only the first data message carries the destination address, which is needed to initially set up the link. Most data communications networks are packet-switched.

As mentioned in Chapter 2, packet-switching can be implemented using either a virtual circuit or a datagram service. The difference between these two transport schemes is that with virtual circuit packet-switching all the packets are transported along the same virtual path as if the path were a dedicated circuit. Furthermore, virtual-circuit packet-switching employs a store-and-forward transmission in which complete packets are stored first and then forwarded. In datagram packet-switching. however, packets are transmitted independently of each other. This implies that packets can travel along separate paths, which requires separate routes to be established for each packet transmission. Moreover, packets can arrive out of order, which requires the destination node to reassemble them in the correct order. (See Table 2.1 for a summary of the differences between the two switching techniques.)

46. Now that I have an understanding of some of the physical layer issues, let's discuss the different types of network media.

OK. We'll begin by first describing attributes that are common among all network media except, of course, wireless media.

All physical media, regardless of their type, share three common physical elements. First, a *conductor* serves as a medium for the physical signal. This conductor is composed of either copper wire or glass or plastic fiber. In the case of copper, the wire can be stranded (i.e., composed of several thin wires) or solid, which can be thought of as a single "thick" strand. Stranded wire is usually stronger and more flexible. Furthermore, the thickness of a wire is given in terms of gauge, which represents the conductor's diameter. The lower the gauge, the thicker the wire. Thus, 22-gauge wire is thicker than 24-gauge. Most often, wire gauges are expressed in terms of AWG—American Wire Gauge—which is a classification system for copper wire based on a wire's cross-section diameter. For example, AWG 24 means that the conductor's diameter is 0.51 mm. The smaller the AWG number, the larger the diameter.

Second, there is usually some sort of *insulation* material surrounding the conductor. The insulation serves as a protective "barrier" to the conductor by preventing the signal from "escaping" and preventing electrical interference from "entering."

Finally, the conductor and insulation are encased in an outer sheath or "jacket." This jacket is composed of any of a number of materials, such as polyvinyl chloride (PVC) for nonplenum cable or Teflon for plenum cable. Plenum cable is used for cable "runs" through a return air system. The Teflon coating provides a low-flame spread and does not release toxic fumes as quickly as PVC does in case the cable burns during a fire. Both PVC and Teflon give off nasty toxic gases when burning. Teflon, however, is fire retardant (which is not the same as "fire *resistant*," where fire would not start) and takes much longer to get to a burning point. This decreases the chance of toxic fumes to affect people in a burning structure at the beginning of the fire.

47. How do these physical characteristics correspond to twisted-pair cable?

Twisted-pair cable, which is commonly used in communication networks, consists of at least two insulated copper wires that have been twisted together. Figure 4.11 shows the physical composition of two such cables: unshielded twisted-pair (UTP) and shielded twisted-pair (STP) cables.

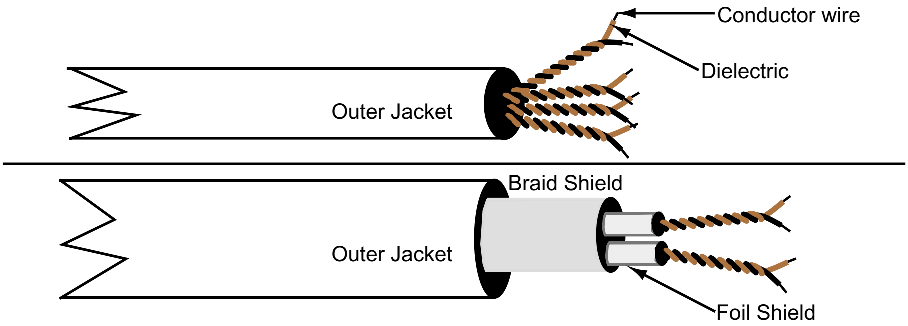


FIGURE 4.11 A UTP cable (a) and an STP cable (b). Pairs of wires are twisted around each other. One pair is used to transmit data; a second pair is used to receive data. Note the extra shielding in the STP cable.

Examining both twisted-pair cable types relative to their physical attributes, we find that each consists of an inner core called a *conductor wire*, which is made of copper (or is comprised of copper and other metals such as tin or silver). The conductor wire is insulated by a *dielectric* material such as polyethylene. In UTP cable, this ensemble is housed within an outer jacket. In STP cable, each conductor wire pair is wrapped in a foil metal shield, and cable pairs are encased within a copper or tinned copper braid shield. The foil and braid shields make the wires less susceptible to electrical interference and noise.

48. What about coaxial cables?

A second type of network cable is *coaxial* cable (or simply, *coax*, pronounced co-axe). Coax consists of a single-wire copper conductor surrounded by a dielectric material and two types of shielding, a foil shield and a braided shield, arranged concentrically and encased in a PVC or Teflon outer jacket (Figure 4.12). This design heavily resists interference and has a high bandwidth. It is not totally impervious to noise, but it offers much more protection against the hazards that afflict UTP. The internal single-wire conductor can be stranded or solid; the former is preferred for hostile environments. Technically, coaxial cables exist in many, many cross-sectional diameters, ranging from several millimeters to several centimeters.

49. How do UTP, STP, and coax compare to fiber?

Fiber-optic cable consists of a glass fiber covered by a plastic buffer coating and surrounded by Kevlar fibers. The Kevlar fibers give the cable its strength. It is the same material used to make bulletproof vests and combat helmets. The Kevlar fibers are surrounded by a protective outer sheath (Figure 4.13). Notice once again the three primary physical attributes of cable: conductor, insulation, and outer sheath. The outer sheath (i.e., jacket) keeps the fiber safe and must meet any local electrical and cabling codes. Typically, fiber is enclosed in a fiber jacket called a buffer that is used to separate the fiber itself from any external contact and to protect the fiber from damage. Buffers range from standard dielectric, foam, or in the case of fibers that may be submerged in water or other liquids, gel-packed fiber. In such environments, gel-pack is useful because water that invades a cable

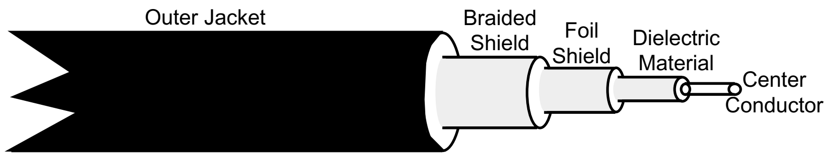


FIGURE 4.12 A coaxial cable typical of the types used in computer networks. Two layers of shielding provide protection against external noise and interference. The outer jacket protects the cable from the elements and may either be polyvinyl chloride (PVC) or Teflon; the latter is appropriate for cable runs in air plenums. Source: Adapted from Leeds & Chorey, 1991a.

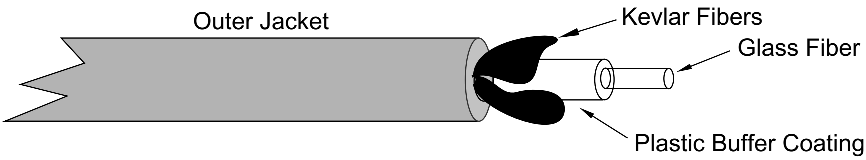


FIGURE 4.13 A fiber-optic cable. Source: Adapted from Codenoll Technology Corporation, 1993.

can expand or contract (based upon pressure and temperature) and damage the fiber(s). Tight buffered cables allow maximum protection for the fiber; loose buffered cables are useful in locations where the fiber (or tube) around the fiber will expand and contract (usually outdoors). A loose buffer (tube) around the fiber allows the buffer to expand and contract independently of the fiber itself.

Cables are jacketed with a variety of materials ranging from aluminum and Kevlar to PVC. Which jacket is used depends on local fire and electrical codes as well as where the cable is to be placed in the structure. As mentioned earlier, some materials such as PVC give off noxious fumes when burned and are therefore not allowed in areas where there is a return air plenum. Each jacket type has a variety of cable "stiffeners" inserted between the fibers to give the cable some rigidity and to afford the cable some strength when pulled or suspended. The number and type of stiffeners used (e.g., aluminum rods inserted in the jacket) vary from vendor to vendor.

50. You've described physical attributes. What about electrical characteristics?

Good question. The performance of a "wired" network is greatly dependent upon the electrical characteristics of the cable used. Since bits ultimately become physical signals at the physical layer, signal quality is an important consideration when selecting a specific medium. Three very important electrical characteristics directly associated with signal quality are *capacitance, impedance,* and *attenuation*.

51. What is capacitance and how does it relate to signal quality?

Capacitance is the property of a circuit that permits it to store an electrical charge. The capacitance of a cable determines its ability to carry a signal without distortion, which is the "rounding" of a waveform (Figure 4.14) due to a stored charge between the conductors of the cable. The more distorted a signal becomes, the more likely a receiving node will be unable to distinguish between 0s and 1s. High-quality cable has low capacitance—the lower the capacitance, the longer the distance a signal can travel before signal distortion becomes unacceptable.

We must make an important point about capacitance and network data cabling. While network cable can have low characteristic capacitance per meter, the overall capacitance of the cable increases as the cable gets longer. Because of noise and other problems in transmission, a maximum cable length of about 100 meters exists for unshielded twisted-pair (UTP) network cable. This is true for even low-capacitance, high-quality UTP cable,

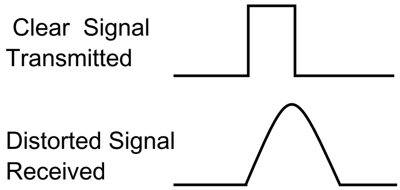


FIGURE 4.14 Capacitance eventually will distort a transmitted signal. Source: Adapted from Leeds & Chorey, 1991a.

which is used in a great preponderance of LANs. If the UTP cable is replaced with shielded twisted-pair (STP), which is very popular in token ring networks, there is still a maximum limitation under perfect conditions of about 100 meters. The reason for this limitation, however, is completely different than that for UTP cable. As STP cable gets longer, the capacitance builds up. This results in enough signal degradation to require some sort of signal amplification or regeneration for a signal to travel a greater distance.

52. So that's why there's a 100 meter length restriction on the UTP cable we use for our Ethernet LAN.

That's part of the reason. There are other reasons as well, which we will address in our discussion on Ethernet/802.3 LANs in Chapter 8.

53. What is impedance?

Impedance is a measure of the opposition to the flow of electrical current in an alternating current circuit. Measured in ohms (abbreviated by the Greek symbol omega, Ω), impedance is a function of capacitance, resistance, and inductance. (*Note:* Inductance is a property of an electrical circuit that opposes any change in the amount of current flowing within the circuit. Inductance in a circuit is analogous to inertia in mechanics and, in fact, is sometimes called electromagnetic inertia.) *Impedance mismatches*, caused by mixing cables of different types with different characteristic impedances, can result in signal distortion. Cable manufacturers always list a cable's impedance, and you should pay close attention to these measurements. Also, different network hardware types may require different impedance values and may not work with values that are out of the range of performance for the hardware. For instance, most token ring equipment requires 150 Ω of impedance. On the other hand, Ethernet/802.3 twisted-pair networks want 85–111 Ω and don't appreciate 150 Ω at all.

54. What is attenuation?

Attenuation is the decrease in signal strength, which occurs as the signal travels through a circuit or along a cable. The longer the cable, the greater the attenuation. Also, the higher the frequency of the signal, the greater the attenuation. Different types of cables are also subject to different amounts of attenuation. For example, in twisted-pair cable, the

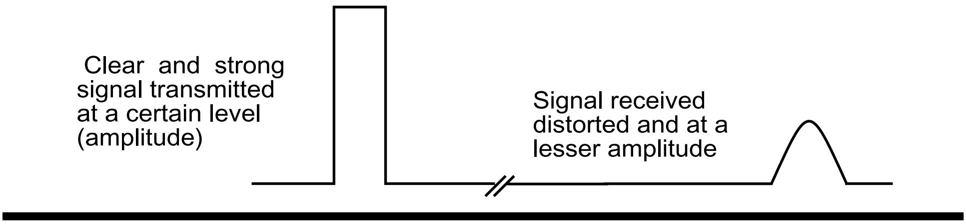


FIGURE 4.15 The combined effects of capacitance and attenuation result in a signal that is received distorted and weaker than what it was when transmitted. This can severely impact the performance of a network. Source: Adapted from Leeds & Chorey, 1991a.

attenuation rises sharply as the signal frequency increases, whereas with coaxial cable, it rises less sharply as frequency increases. Fiber-optic cable, which is tuned for a specific wavelength, exhibits very low attenuation per unit of distance *at that wavelength*. Attenuation is measured in decibels (dB) of signal loss. When selecting cable, you should choose a type that has a low measure of attenuation for the network speeds and distances involved. Signal quality is affected most by the combination of attenuation and capacitance.This is illustrated in Figure 4.15.

55. So of the different wired network media mentioned thus far, there really are only two broad categories: copper and fiber. Is this correct?

Yes. These two categories can then be partitioned into more specific types. For example, copper cable includes unshielded twisted-pair (UTP), shielded twisted-pair (STP), IBM cable, and coaxial cable. Fiber-optic cable includes both glass and plastic fiber. There also is single mode fiber and multimode fiber.

56. Now that I am aware of the common physical and electrical characteristics of copper and fiber cables, I would like to know some specific information about each type as they relate to networks. Let's start with twisted-pair media.

OK. Twisted-pair cable is probably the most popular type of cable used in networks today. It works with all different types of networks. The name, twisted-pair, comes from how the cable is constructed. Twisted-pair cable consists of at least two insulated copper wires that have been twisted together. Data transmission requires four wires (two pairs): one pair to transmit data and one pair to receive data. UTP cable used in data networks has a twist in the cable about every six inches (approximately 15 cm).

57. What organization develops standards for twisted-pair cable?

Standards for UTP and STP are provided by the Electronic Industries Association and the Telecommunications Industry Association (EIA/TIA). These organizations jointly developed the EIA/TIA-568 standard, which is a North American standard used worldwide. EIA/TIA-568 specifies the type of cable that is permitted for a given speed, the type of connectors that can be used for a given cable, and the network topology that is permitted when installing cables. The standard also defines the performance specifications that cables and connectors must meet. In short, EIA/TIA-568 represents a comprehensive standard for premises wiring that addresses network design and performance characteristics for physical media. Within the EIA/ TIA-568 standard, there is an A version (568A) and B version (568B) that are used for industrial and nonindustrial networks, respectively. For more information about TIA, check out its Web site at http://www.tiaonline.org.

58. What are these standards?

The EIA/TIA-568 standard for UTP cable classifies the cable into the following categories: Categories 1, 2, 3, 4, 5, and 5e (Category 5 enhanced). Several other categories also exist, but they have not been standardized by EIA/TIA. They include Category 6, Category 7, Category 6/Class E, and Category 7/Class F. There's also a category for STP. Table 4.1 contains a summary of the various twisted-pair categories.

59. What's the most popular category and why?

Of the various UTP cable types, Categories 3 and 5 receive the most attention in LAN circles today. However, with higher-speed LANs such as 100-Mbps and 1-Gbps Ethernet/

TABLE 4.1 Descriptions of Twisted Pair Cable Categories

Category	Description
Category 1*	Used for voice transmission; not suitable for data transmission.
Category 2*	Low performance cable; used for voice and low-speed data transmission; has capacity of up to 4 Mbps.
Category 3*	Used for data and voice transmission; rated at 10 MHz; voice-grade; can be used for Ethernet, Fast Ethernet, and token ring.
Category 4*	Used for data and voice transmission; rated at 20 MHz; can be used for Ethernet, Fast Ethernet, and token ring.
Category 5*	Used for data and voice transmission; rated at 100 MHz; suitable for Ethernet, Fast Ethernet, Gigabit Ethernet, token ring, and 155 Mbps ATM.
Category 5e* (Enhanced)	Same as Category 5 but manufacturing process is refined; higher grade cable than Category 5; rated at 200 MHz; suitable for Ethernet, Fast Ethernet, Gigabit Ethernet, token ring, and 155 Mbps ATM.
Category 6	Rated at 250 MHz; suitable for Ethernet, Fast Ethernet, Gigabit Ethernet, token ring, and 155 Mbps ATM (not yet formalized as of this writing).
Category 6 (Class E)	Similar to Category 6 but is a proposed international standard to be included in ISO/IEC 11801.
Category 6 (STP)	Shielded twisted pair cable; rated at 600 MHz; used for data transmission; suitable for Ethernet, Fast Ethernet, Gigabit Ethernet, token ring, and 155 Mbps ATM.
Category 7	Rated at 600 MHz; suitable for Ethernet, Fast Ethernet, Gigabit Ethernet, token ring, and 155 Mbps ATM (not yet formalized as of this writing).
Category 7 (Class F)	Similar to Category 7 but is a proposed international standard to be included in ISO/IEC 11801.

* EIA/TIA-568 Standard

802.3 becoming more commonplace, Category 5e as well as nonstandard Category 6 and Category 7 are receiving considerable attention. Category 3 is popular because it is the most common wire used for voice transmission in telephone systems in most commercial buildings. Furthermore, recent IEEE LAN protocols include a specification for Category 3 cable as a LAN medium for 100-Mbps networking. By using its existing cabling plant for local area networking, an organization does not have to modify its wiring infrastructure, thus saving a considerable amount of money. Categories 5, 5e, and the other higher-category copper UTP cables are popular because they are data-grade (not voice-grade, as is Category 3 cable). Category 5 is quite plentiful and has emerged as the cable of choice for 100-Mbps LANs. Organizations retrofitting their wiring infrastructure, or those engaged in new installations, are typically installing either Category 5 or Category 5e cable. Data-grade cable is usable for voice or data. Voice-grade cable is usable for voice and for some types of data connections depending on the transmission speed and technique used.

60. So there's no reason why I shouldn't use UTP for all my network cabling, right?

Although UTP cable is a popular LAN medium, these cables pose problems in data transmission at the higher frequencies required by higher-speed networks. Two major factors are attenuation, which we discussed earlier, and crosstalk.

61. I recall our discussion of attenuation. Attenuation occurs when the strength of a signal degrades as it travels along the cable. What's *crosstalk*?

Crosstalk is electrical interference—it's another example of noise, which we discussed in Chapter 2. Crosstalk occurs when energy radiated from one wire pair is "picked up" on another pair. In one type of crosstalk, called *near-end crosstalk* (abbreviated NEXT), a signal on the transmit pair is so strong that it radiates to the receive pair. A direct consequence of this "spilled-over " radiation is that the receiving device cannot decipher the real signal. The combined effects of distortion and crosstalk result in an irregular variation in the shape or timing of a signal. This irregular variation is called jitter (kind of like a nervous person). Jitter is primarily caused by mixing unshielded and shielded cable.

62. What causes crosstalk?

Several factors, including the closeness of the wire pairs, the quality of the wire, and the number of twists per foot, cause crosstalk. Twisting wire pairs reduces crosstalk between a specific signaling pair. NEXT increases significantly for the first 60 feet (approximately 18 m). Higher-quality cable (read: more expensive) contains higher-quality wire compositions, better insulation, and improved twist per foot ratios between cable pairs, all of which will reduce NEXT.

63. I can understand how UTP cable is conducive to crosstalk, but what about STP cable? Is it also susceptible to crosstalk?

In contrast to UTP, STP cable can dramatically reduce the hazards of crosstalk and noise because individual wire pairs are shielded. STP is still susceptible to the same kinds of interference that can wreak havoc over UTP, however. The difference is that STP can withstand more noise abuse than UTP and therefore can provide a more reliable transmission medium. STP cable, however, typically has a much higher impedance, which can cause signal reflections on transmission systems that require a lower-impedance cable—and that means data errors.

64. In your list of different cable types you mentioned IBM cable. What is this?

IBM has its own classification of cable—the IBM Cable System (ICS)—which specifies nine cable types. Of the nine "types" defined, specifications are available for only seven; types 4 and 7 have no specifications. Table 4.2 contains a summary of ICS. Be very careful not to confuse ICS cable types with the categories of cable specified by EIA/TIA-568. A *type* is a grouping of categories and fiber-optic cables in a bundle based upon which type is being selected. A *category* is an EIA specification for a cable's construction.

65. Why does IBM have its own classification?

Twisted-pair IBM cable is similar to non-IBM twisted-pair cable with one exception: The IBM version has more stringent specifications. This is why IBM adopted its own classification; it wanted to make certain that cable used in proprietary IBM environments satisfied IBM's high standards. IBM is not the only company to develop its own standards. Lucent Technologies, which used to be a major component of AT&T, and Anixter both have their own versions of cable bundles and specifications for installation.

66. Let's move to coax. Please explain some of its specifics related to networking. For example, I've heard of thin and thick coax. What do these mean?

In computer networking, coax is described as either thick or thin. Thick coax is used as the medium for "Thick Ethernet," which is known as IEEE 802.3 10BASE5. Depending on the manufacturer, the cable's outer diameter ranges from 0.375 inch to 0.405 inch (0.96

TABLE 4.2 IBM Cable System

Type	Description
Type 1	2-pair STP, 22-gauge solid wire; used for Token Ring networks.
Type 2	Contains UTP and STP; 4-pair UTP, 22-gauge solid wire used for voice.
Type 3	2-pair UTP, 22-gauge solid wire used for data; 2-, 3-, or 4-pair UTP cable with 22- or 24-gauge solid wire; pairs must have a minimum of 2 twists/foot; voice-grade only.
Type 4	Not defined.
Type 5	Fiber-optic; 2 glass fiber cores at 100/140 micron; 62.5/125 micron fiber also allowed and is recommended by IBM; used as main ring of a Token Ring network.
Type 6	2-pair STP, 26-gauge stranded wire; used mostly as a patch cable to connect a node to a network.
Type 7	Not defined.
Type 8	2-pair STP, 26-gauge flat solid wire; designed for under-carpet installations.
Type 9	2-pair STP, 26-gauge solid or stranded wire; contains a plenum outer jacket; used for between-floor runs.

cm to 1.04 cm). Thick coax resembles a garden hose and is known as “Etherhose” in slang terms in the industry. It has a designation of RG-8 with 50 Ω impedance. This medium is expensive and outdated; networks based on it today are usually inherited, not installed. Thin coax is used as the medium for “Thin Ethernet,” which is known as IEEE 802.3 10BASE2. Its outer diameter ranges from 0.175 inch to 0.195 inch (0.448 cm to 0.5 cm). Thin coax is designated RG-58 and it too has 50 Ω impedance. Although quite popular in the 1980s and early 1990s, thin coax has fallen out of favor for UTP, and as is the case with Thick Ethernet, a Thin Ethernet is more likely to be inherited and not installed today.

In analog coaxial networks, such as residential cable television networks, cable such as RG-9 may be used, which typically has a greater impedance factor (62 Ω to 76 Ω) and differing electrical characteristics from those of RG-8. Similarly, RG-59 with an impedance of 75 Ω is used for home TV cable, but it looks almost the same as RG-58. It is easy to become confused because the cables within each genre look practically the same. In fact, use of the wrong type of cable is one of the most frequent causes of insidious network failure. Therefore, because not all coaxial cables are electrically the same, you must be careful to select the right one for the types of network equipment being considered for use.

67. I’m still a bit confused. Can you go over this again for me?

OK, no problem. We’ll go over it again, because the distinction is very important. Although outwardly, the cable used for cable TV resembles thin or thick coax, it is not the same electrically. Thin cable used for cable television is designated RG-59 and has 75 Ω impedance. This is quite different from thin network coax, which has 50 Ω impedance. As discussed earlier, impedance mismatches—caused by mixing cables with different impedances—can result in signal distortion.

68. Besides impedance differences, are these two coax cables functionally equivalent?

Not really. Coaxial cable can function in two different ways—baseband and broadband. Baseband uses the entire bandwidth of the coaxial cable to carry a single signal, whereas broadband shares the bandwidth of the coaxial cable among multiple signals. Baseband is primarily used in LANs; broadband is primarily used in cable TV applications and high-performance, shared telephone network systems.

Comparing the two, baseband is relatively simple and inexpensive to install, requires inexpensive interfaces, and is ideally suited for digital transmission. Broadband equipment is much more expensive, is based on analog signaling, requires expensive amplifiers to strengthen its signal, and personnel to maintain it. Broadband transmission can achieve higher bandwidth than baseband transmission, however.

69. Are there any other types of copper cable other than twisted-pair and coax?

Yes. We’ll mention one more type of copper cable for completeness, because it exists in some computer networks, albeit proprietary ones that are a vestige of the mainframe days. It’s called *twinaxial* cable (twinax, for short). It is much like coaxial cable except that there are *two* inner conductors instead of one. Because it is not used in the networks we discuss in this book, we’ll stop right here.

70. Okay. Let's discuss fiber-optics cable from a networking perspective.

Fiber-optic cable, which was described earlier, is used in LANs as an alternative to copper cable. It carries data signals in the form of modulated light beams. The electrical signals from the sending computer are converted into optical signals by a light source—a light-emitting diode (LED) or a laser. An LED is a semiconductor device that converts electrical energy into light. With an LED source, the presence of light represents a 1, and the absence of light (i.e., no light pulse) represents a 0; with a laser source, which emits a continuous low level of light, a 0 is represented by the low level and a 1 is represented by a high-intensity pulse. This modulation technique is called *intensity modulation*. The light pulses enter one end of the fiber, travel through the fiber, and exit at the other end. The received light pulse is then converted back to electrical signals via a photo detector, which is a tiny solar cell.

71. Whenever my colleagues talk about fiber, they inevitably use numbers like 62.5 and 125. What do these numbers mean?

Fiber-optic cable is specified by the size of the glass fiber, which consists of two parts: an inner glass cylindrical core and an outer concentric glass cladding that has different optical characteristics than the core. The core and overall outside diameters are measured in microns (a micron is one micrometer, which is 1-millionth of a meter) and abbreviated by the symbol µm. A standard size is 62.5/125 µm—62.5 µm specifies the diameter of the inner glass core and 125 µm specifies the diameter of the outer concentric glass cladding. In Europe, 80/100 µm fiber is common.

72. What's the purpose of having an inner glass core and an outer glass cladding?

The inner glass core and outer glass cladding are the key elements of fiber-optic cable. The outer cladding of the glass is reflective; the inner core of the glass is transparent. Light goes through the transparent core, but remains in the core by bouncing off the reflective cladding, which is like a cylindrical mirror around the core. Thus, light stays in the fiber core as if it were a "light pipe" the same way water stays in the hollow core of a metal clad pipe.

73. I would think that with light bouncing all around inside the cable you would lose some of it at the other end of the cable?

There obviously will be some loss, but not all fiber-optic cable permits light to bounce around.

74. Are you implying there are different types of optical fiber?

Yes. In addition to core size, fiber-optic cable is classified by the manner in which light rays travel through the medium. There are two general classifications: *multimode* and *single mode*. A brief description of each follows:

Multimode Fiber In multimode fiber, the core diameter ranges from 50 µm to 100 µm (i.e., from about 1/500th of an inch to about 1/250th of an inch). Also, in multimode fiber, different rays of light bounce along the fiber at different angles as they travel through the core (Figure 4.16). Therefore, the light rays actually travel different total distances as they

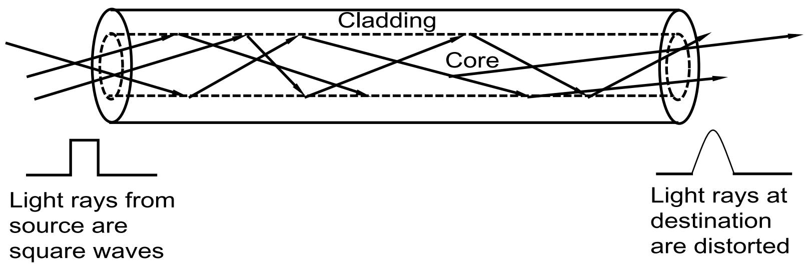


FIGURE 4.16 In multimode fiber, the distance light rays travel through a cable varies. Some rays travel longer distances from sending and receiving nodes; others travel shorter distances. The cladding layer reflects any stray light waves, causing signal distortion at the receiving end. Source: Adapted from Codenoll Technology Corporation, 1993.

go from one end of a long fiber-optic cable to the other end. Some light rays travel longer distances and some travel shorter distances, while the speed of light is a constant; therefore, some of the rays will arrive at the other end of the cable later than others. A consequence is that a pulse of light that enters one end of the cable might exit the other end with a little more spread (or dispersion) because some of the light rays get to the other end sooner than others. Thus, there is some amount of signal distortion at the receiving end of a transmission.

Single-Mode Fiber In single-mode fiber, the core diameter is 7 μm to 9 μm, which is about one-3000th of an inch. Therefore, it is considerably thinner than multimode fiber. When the diameter of the core is reduced to the order of a wavelength, the light cannot bounce off the walls of the core, allowing only a single ray of light, called the axial ray, to pass. (Single mode implies a single ray on a given frequency of light.) In single-mode systems, a light wave entering the fiber exits with very little distortion, even at very long distances and high data rates (Figure 4.17). In advanced systems (e.g., synchronous optical network, or SONET, which is discussed in Chapter 7), single-mode fiber transmission systems may allow multiple light sources in different light spectrums to interoperate over the same fiber (very similar in concept to multiple channels of TV on a cable system in a neighborhood). In this manner, the same fiber can increase its carrying capacity by changing out the electronics on both ends of a fiber link as additional speed is required through the fiber medium.

75. What about applications? When is one favored over another?

In a typical building or campus network, we usually do not have to worry about single mode versus multimode. Either can be used for almost any application. However, if we are connecting cities across a country (such as telephone and cable companies are currently doing), then single-mode fiber is used for these long distances. If either will do, it boils down to cost. Multimode fiber cable is much less expensive than single-mode fiber cable.

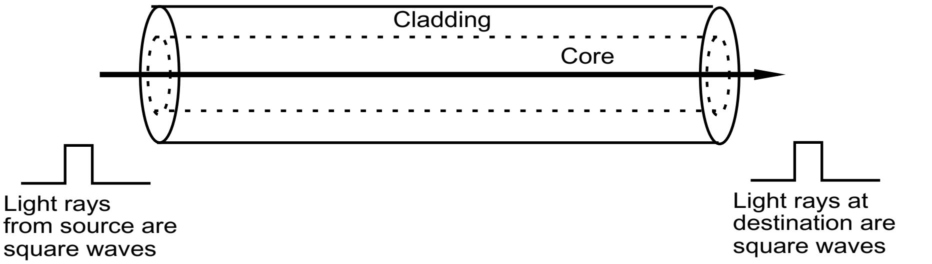


FIGURE 4.17 In single-mode fiber, all light rays travel the same distance from sending to receiving node. A direct consequence of this is no signal distortion at the receiving end, which results in higher bandwidth and lower attenuation per kilometer than multimode fiber. Single-mode fiber is the ultimate medium for long-distance connectivity. Source: Adapted from Codenoll Technology Corporation, 1993.

76. How durable are the fibers?

Fibers, after a time, can develop microfractures (cracks) or strain fractures in suspended fibers. Fiber that has been physically pulled may have stress fractures. Many other nasty things can happen to perfectly good fiber. Glass fiber has very high tensile strength. The problem is that if a crack develops on the surface of the fiber, it may eventually cause the fiber to break or will cause optical dispersion problems in the meantime. This structural problem is similar to that of a pane of glass. Glass is fairly strong, but when scribed, the glass is easily broken. Further, when the pane is scribed and placed under strain, the scribed line will lengthen and weaken the pane, eventually causing breakage. Fiber cable can succumb to this problem.

Another problem is *bend radius*. While fiber is much more flexible than copper cable and can be bent in much smaller radii than equivalent copper, microbends may appear in fiber that has been bent too tightly or “kinked.” Microbends can cause light path disruption and increase the loss of the cable. Areas subject to chemical exposure, radical temperature changes, nuclear radiation, and other disruptive effects require special consideration in cable jacketing and type of fiber used. Check with the local installation specialist to find out which fiber type is best for your environment.

77. What advantages does fiber-optic cable have over copper cable?

Fiber-optic cable is the ideal network medium when cable is involved. It is immune to electromagnetic interference (EMI) and other types of externally induced noise, including lightning. It is unaffected by most physical factors such as vibration. Its size is smaller and its weight lighter than copper. It has much lower attenuation per unit of length than copper. It can support very high bandwidth. With copper cable, we speak of bandwidth on the order of Mbps or Gbps with Gigabit Ethernet. With fiber-optic cable, we can speak of bandwidth in terms of Mbps, Gbps, terabits per second (Tbps), and beyond. In short, fiber is the most effective medium available in terms of bandwidth and reliability.

78. So, given all that good stuff, why do we even bother with copper?

Fiber-optic cable does have its drawbacks, however. First of all, glass fiber (the most common type of fiber) is more expensive than copper. Second, the signal strength of fiber degrades when there is any light loss. Third, if used as the medium for a LAN, it is limited to either a point-to-point or star configuration.

79. Earlier you mentioned plastic fiber. What is this?

An alternative to glass fiber is plastic fiber, which is constructed of plastic rather than glass and is more flexible than glass fiber. It also can be used in areas where it might be subjected to pressure that would crush a glass fiber. For example, if an office chair rolls over plastic fiber, the fiber springs back to its originally cast shape; glass fiber, on the other hand, breaks and, hence, must be cut and spliced. Plastic fiber, although available today, is not yet as popular as glass fiber. There are several reasons, including vendor availability, vendor expertise, manufacturing-related problems, and a very young set of standards for plastic fiber (it is known in IEEE circles as GRINPOF—graded index plastic optical fiber).

Plastic fiber is also very easy to terminate and install. Glass fiber requires cutting and painstakingly careful polishing to ensure that signal loss is minimal at a splice or connection point. Plastic fiber is cut with a knife and heated in a special hand-held device that looks like a hair dryer. Through heat, the fiber is terminated and automatically polished, and it is available for use about 60 seconds later. Over time, plastic fiber will become the medium of choice for high-speed to-the-desktop connections due to its bandwidth, ease of installation, relative cheapness, and flexibility. At this writing, however, it is still pretty new in the industry and will require time to become popular.

80. Is there anything else I should know about fiber?

Yes. One more thing. If you are planning a fiber optic installation today, then you should keep in mind the following:

multimode, graded-index, dual window fiber with a median frequency of 1300 nm

81. What does all this mean?

The multimode mode part of this expression was explained earlier. To understand the concept of graded-index, we first need to explain some properties of light. If a light ray travels within a medium of constant optical density, then the ray travels in a straight line. If, however, a light ray enters a medium of different optical density, and the ray enters the medium at an oblique angle, then the direction of the ray changes. This change of direction is called *refraction* and involves the bending of light. A common example of refraction is the "broken" appearance a stick has when it is immersed in water and viewed from an oblique perspective. The concept of "index" refers to a fiber-optic cable's index of refraction, which is a measure of the amount of bending a light ray undergoes as it travels within the cable. In a *graded-index* fiber, the core's refractive index is modified from its center to its edges to decrease the amount of modal dispersion. If you look at Figure 4.16, you will note that the light pulses are reflected off the cladding—this is what guides the light rays from source to destination. In multimode graded-index, variations in the density of the core medium change its index of refraction such that light is refracted (i.e., bends) toward the center of the fiber. So instead of light rays "bouncing" back and forth in a "V and inverted "V" pattern as shown in Figure 4.16, we have more of a "rounded" pattern . Thus, light rays within a graded-index fiber propagate along an oscillatory path, which decreases the amount of modal dispersion. This change in the fiber's refractive index reduces the amount light "bounces," which enables the signal to travel faster through the cable. In contrast to graded-index, there is *step-index* fiber. Unlike the gradual change of a graded-index fiber core's refractive index, the index of refraction of a step-index fiber's core is uniform. It should be clear that the graded-index concept is appropriate for multimode fiber, and step-index is appropriate for single-mode fiber.

The last part, *dual window*, *1300 nm* refers to the fiber's ability to operate at more than one frequency. Specifically, dual window 1300 nm means that data can be transmitted at a wavelength of 1300 nanometers (a nanometer is 1-billionth, that is, 1/1,000,000,000 of a meter and abbreviated nm) with a corresponding frequency of 625 Mbps, one standard frequency step higher at 1550 nm (corresponding frequency of 2.4 Gbps), or one standard frequency step lower at 850 nm (155 Mbps). This is important if you want to use fiber for existing networks such as Ethernet/802.3 (which is perfectly happy with 850 nm) and later upgrade to ATM (which requires 1300 nm and 1550 nm for higher speed connections).

82. We've covered twisted-pair, coaxial, and fiber-optic cable. These are "wired" or cable-based media. What about wireless communication?

The topic of wireless communication is discussed in greater detail in Chapter 18. However, to get you up to speed, we will try to provide you with an overview of many of the concepts related to this topic.

83. Great! Let's begin with the basics.

OK. First of all, in wireless communication, signals travel through the atmosphere instead of through a physical cable.

84. Through the atmosphere? What about satellite broadcasts? There is no atmosphere in space.

Oops! You're right, reminding us of that old joke about someone giving a review of a restaurant on the moon: Great view but no atmosphere. We stand corrected. Let's substitute *atmosphere* with *space*. So, in wireless communication, signals travel through space.

85. What types of wireless communications are there?

There are two general types of wireless communication: *radio transmission* and *infrared transmission*.

86. What's radio transmission?

Radio transmission refers to any wireless technique that uses radio frequencies (RF) to transmit information. RF transmissions are very popular today for wireless data services. RF frequencies typically used for data communications are in the 800 MHz to 900 MHz range of the electromagnetic spectrum (see Figure 4.18). In the United States, the Federal Communications Commission (FCC) has approved additional frequencies for wireless data services to operate in the 1.85 GHz to 2.20 GHz range. This slice of the RF spectrum is used for, among others, pagers, Personal Digital Assistants (PDAs), laptops with PC cards (formerly known as PCMCIA cards), and cellular telephones. In a data communications network, signals from laptop wireless machines are transmitted via built-in antennas to the nearest wireless access point, which serves as a wireless repeater. These access points are connected to a backbone cable system (see Figure 4.19). Radio waves typically are used to enhance existing cable systems rather than replace them. They are still susceptible to electromagnetic interference (EMI), however, and cannot penetrate interior drywalls or concrete bearing walls, particularly walls in buildings with steel frameworks, unless at a high enough frequency, which shortens range of effective signal reach.

87. How does microwave communication fit into the wireless arena?

Microwave is another RF transmission method. It uses high frequency waves and operates at a higher frequency in the electromagnetic spectrum (see Figure 4.18). The microwave spectrum encompasses frequencies between 2 and 40 GHz. Access to these frequencies is strictly controlled by the FCC in the United States; therefore, users of microwave transmitters must be licensed. The FCC also monitors these frequencies for compliance. Microwave transmissions are considered a *line-of-sight* medium. Since microwave signals travel in a straight line, the transmitter and receiver must be in each other's line-of-sight. If not, because of their very short wavelength, microwave signals degrade once they encounter an obstruction. Even water droplets in the atmosphere attenuate microwave signals. Consequently, it is necessary to "spec out" the environment to ensure that a microwave transmitter and receiver will have a clear line of sight, sufficient power to offset attenuation, and a small enough distance between stations before installing them. A microwave medium uses parabolic antennas mounted on towers up to 30 miles apart. Because of the impact of the curvature of the earth on the line of sight, the higher the tower, the greater the range. (As an exercise, calculate how tall the towers need to be to transmit a line-of-sight signal from, for example, Miami to Lisbon, a distance of about 6700 km.)

88. What kind of data rates are microwave transmissions capable of?

Rates typically range up to 45 Mbps.

89. What are the advantages and disadvantages of microwave communication?

Line-of-sight media such as microwave are less expensive to install than cable for moderate distances in most situations. Microwave also offers a relatively high data rate (for a wireless medium). It also requires little or no maintenance, is fairly easy to implement, and has no recurring monthly or yearly costs as is the case with leased circuits. On the other hand, line-of-sight transmissions are subject to environmental and atmospheric conditions (rain, fog, high humidity), as well as electromagnetic interference from many sources including solar flares and sunspots. Furthermore, if units are placed too close to each other, overloading and signal interference can result. Because of these environmental and atmospheric drawbacks, you should not rely upon microwave communication completely for mission critical operations. However, if an application can endure an occasional failure, then such media are acceptable.

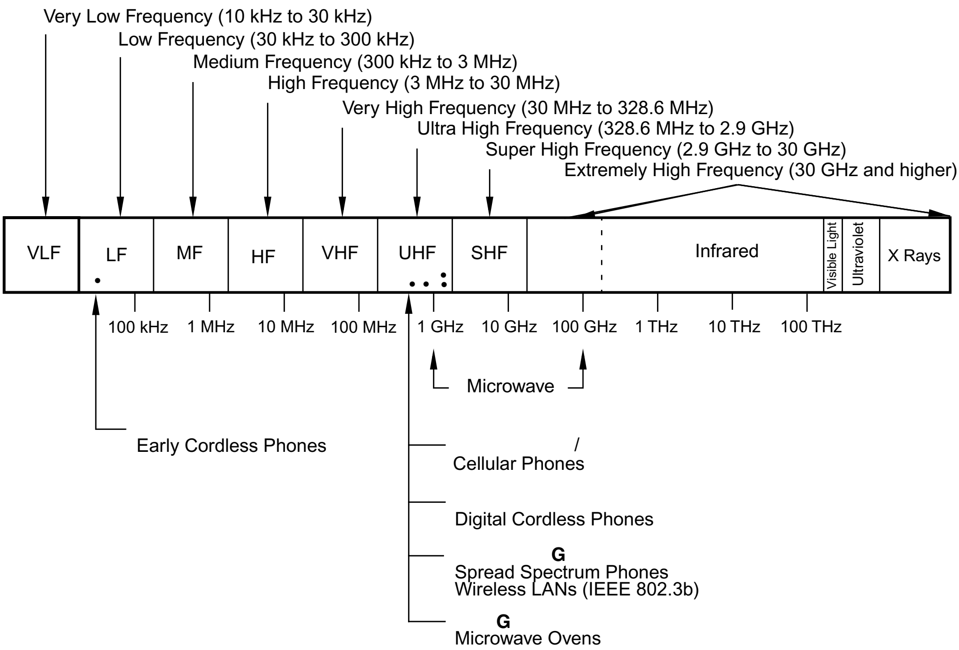


FIGURE 4.18 The electromagnetic spectrum (in Hz). Higher frequencies support greater bandwidth. Source: Adapted from Breidenbach, 1990 and from http://www.jneuhaus.com.

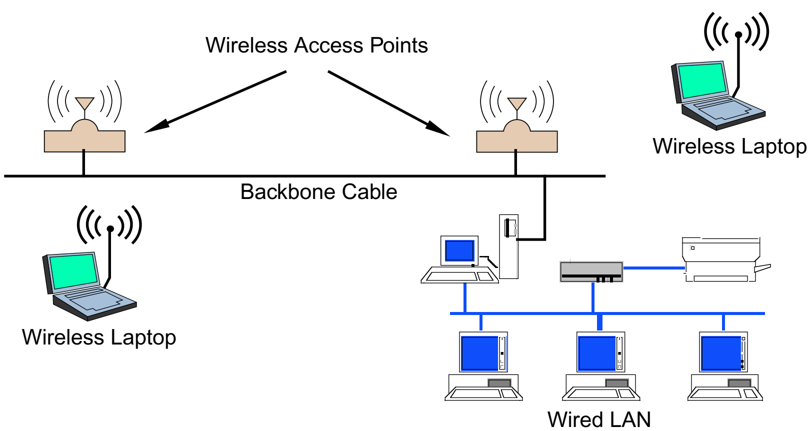


FIGURE 4.19 One example of a hybrid wireless environment. The "wireless access points" are directly connected to a backbone cable and serve as wireless repeaters for the wireless laptop computers. Completely wireless LANs also are possible. Source: Adapted from Karvé, 1997a.

90. Looking at Figure 4.18, I was wondering if you could give me some additional common frequencies and their applications?

OK. Let's see: Early cordless phones operate in the 43–50 MHz range; FM radio operates in the 88–108 MHz range; cellular communications (i.e., cell phones) generally operate in the 824–849 MHz and the 869–894 MHz ranges; digital cordless phones operate in the 902–928 MHz range; pagers and beepers operate in the 931–932 MHz range; broadband personal communication services operate in the 1.85–1.99 GHz range; so-called third-generation (3G) cellular services (see Chapter 18) will operate in the 2.11–2.15 GHz range; the latest "spread spectrum cordless phones as well as some unlicensed wireless LANs (WLANs) operate in the 2.4–2.483 GHz range; and microwave ovens operate at 2.54 GHz. The FCC also recently opened up three new segments of the spectrum for wireless LANs: 5.15 GHz–5.25 GHz, 5.25 GHz–5.35 GHz, and 5.75 GHz–5.85 GHz. Also note that the three frequency ranges, 902–928 MHz, 2.4–2.5 GHz, and 5.8–5.9 GHz are commonly referred to as the *Industrial, Scientific, Medical* (ISM) band. Prior to 1985, these three bands were allocated respectively for industrial, scientific, and medical use. Since then, however, the FCC made these frequency bands available for unlicensed spread spectrum communication.

91. While we're at it, please explain "spread spectrum" to me. I bought my Mom a new spread spectrum cordless phone, which I told her made it better, even though I didn't understand the technology.

Did she ask you if it would make you call her more often? Go ahead—take a break and call her. She'll appreciate it and we'll still be here when you get back. *Spread spectrum*, which is another radio technology, involves varying the frequency of a transmitted signal. This results in greater bandwidth than an unvaried signal. Spread spectrum was introduced in the 1940s and was used to camouflage radio signals. Without it, a signal's frequency will remain constant, which makes the signal more easily susceptible to interference or interception. Spread spectrum transmission camouflages data by mixing signals with a pseudonoise (PN) pattern and transmitting the real signal with the PN pattern. The transmitted signal is spread over a range of the frequencies in the radio spectrum. Thus, to intercept a signal, the intercepting receiver must have two specific pieces of information: the mathematical function the transmitter is using to generate the PN pattern and the exact point in time the function is generated. Today, spread spectrum typically is employed in devices that operate within the ISM range, and many people equate the term "spread spectrum" with these devices. In reality, though, spread spectrum refers to a security technique and not a specific frequency. This is also why devices that employ spread spectrum technology are less prone to interference or unauthorized electronic eavesdropping than devices without it.

92. What about infrared technology? I noticed that my laptop has an infrared port.

Infrared (IR) transmission is another line-of-sight medium. It uses electromagnetic radiation of wavelengths between radio waves and visible light, operating between 100 GHz and 100 THz (Terahertz). IR is generally restricted to LANs within or between buildings. IR transmission can occur in one of two ways: directed and diffused. *Directed IR* requires an unobstructed line-of-sight connection between transmitter and receiver. It is basically a "point and beam" medium. In a *diffused IR* environment, a transmitter "floods" a specific area with a strong infrared signal. The light emitted from the transmitter is spread over a wide angle. The IR signal is transmitted by reflecting off of ceilings, walls, and other surfaces. This is how a TV remote control device works. Thus, diffused IR can be thought of as a broadcast medium, whereas directed IR is point-to-point. Note that as a line-of-sight medium, IR is susceptible to some of the same kinds of problems as microwave, although it is less susceptible to electromagnetic interference than microwave.

93. I have also heard of something called "lightwave wireless." What is that?

In countries without a wire-based network infrastructure, it is much easier to implement a wireless connection topology than a wire-based topology. *Lightwave* connectivity refers to the use of line-of-sight laser-based connection facilities that allow long-distance light-based wireless networking without the need to install cable. Through the use of lightwave, a highly sophisticated transmission network is installed much more quickly than running fiber and yet provides a high bandwidth solution for the installer. Lightwave is an ideal solution for many African countries and even for China, where cable infrastructure is almost nonexistent.

94. What kinds of standards are there for wireless LANs?

After eight years of existence, a standard was finally developed for wireless LANs (WLANs). It is called IEEE 802.11, and it was approved in 1997. This standard defines both the physical and data link layers for wireless communication. (We discuss the IEEE 802.11 data link layer in Chapter 5.) Of the three physical layer signaling methods, two are RF-based and one is infrared. Although IEEE 802.11 specifies three different physical layers, only a single MAC method is used regardless of the physical layer selected (see Chapter 5).

The two RF-based physical layers are called *direct sequence spread spectrum* (DSSS) and *frequency hopping spread spectrum* (FHSS). Both operate at the 2.4-GHz to 2.4835-GHz ISM band. Data transmission rates for DSSS and FHSS initially were defined at either 1 Mbps or 2 Mbps. DSSS operates by spreading a signal over a wide range of the 2.4-GHz band. FHSS operates by transmitting short bursts of data on different frequencies. One burst is transmitted on one frequency, a second burst is transmitted on a second and different frequency, and so forth. Since each uses a different transmission method, the two RF physical layers cannot interoperate. The third physical layer defined in 802.11 is diffused infrared. WLANs based on diffused IR are not popular, however, and hence, most wireless media products currently support one of the RF physical layers.

Shortly after the 802.11 standard was approved, the IEEE began a new standards effort for a higher rate WLAN technology called "802.11 High Rate." This endeavor led to the ratification of IEEE 802.11b in Fall 1999. IEEE 802.11b, which was jointly developed by Lucent Technologies and Harris Semiconductor, supports data rates of 5.5 Mbps and 11 Mbps for 802.11-compliant LANs that employ a DSSS physical layer. FHSS and IR WLANs are still limited to 802.11's original 1-Mbps or 2-Mbps data rates. The new higher-rate standard relies on the same 2.4-GHz ISM band and is backward compatible with original 802.11-compliant WLANs that also use DSSS. Thus, 802.11b will cycle down automatically to 5.5 Mbps, 2 Mbps, and 1 Mbps.

Because IEEE 802.11b does not take advantage of the higher 5-GHz frequencies that are now available courtesy of the FCC, most wireless networking products currently available operate in the 2.4 GHz band, which, as noted earlier, is also used by spread spectrum cordless telephones. The growth of WLANs, coupled with the popularity of spread spectrum cordless telephones, is making the 2.4 GHz frequency band crowded and leading to increased interference that reduces WLAN performance. To resolve this situation, another WLAN standard was approved: IEEE 802.11a (yes, the letters are going in reverse), which uses the 5-GHz frequency band. When the standard is completed, IEEE 802.11a will support data transmission rates of 36 Mbps, 48 Mbps, and 54 Mbps with throughput being scaled in 6-Mbps increments. For example, IEEE 802.11a uses a form of FDM (discussed earlier) called *orthogonal frequency division multiplexing* (OFDM). OFDM partitions the frequency band (5.15–5.35 GHz) into eight 20-MHz channels. Each channel comprises 52 narrow-band carriers of 300 kHz, and each carrier supports a 125-Mbps transmission rate. Furthermore, of the 52 carriers, 4 are used for control purposes, leaving 48 for the payload. Thus, you only get 6 Mbps for data with every 20-MHz channel. OFDM transmits data across the carriers in parallel and aggregates the throughput. This results in an effective data transmission rate of 48 Mbps. Products based on 802.11a will have a smaller signal range than products based on 802.11b because of higher electromagnetic radiation, which is attendant with higher frequencies. There are also proposed proprietary technologies that could double the rates of 802.11b WLANs to 22 Mbps and 802.11a WLANs to 108 Mbps.

Another wireless standard is HomeRF, which is based on the *shared wireless access protocol* (SWAP). HomeRF, expressly developed for wireless home-based networks, is capable of supporting voice and data transmissions at less than 2 Mbps. Its physical layer is based on FHSS and the technology is considered a “watered down” version of IEEE 802.11 because it eliminates some of 802.11’s more complex underpinnings. Although we stated earlier that IEEE 802.11 LANs based on FHSS was limited to a 1- or 2-Mbps data rate, a recent FCC ruling enables HomeRF to support a 10-Mbps transmission rate. HomeRF and IEEE 802.11/b/a are considered competing technologies and are often viewed in the same light as the old Betamax vs. VHS VCR-technology competition of the 1980s. This is because IEEE 802.11/b/a, originally designed as a data transmission WLAN for the office place, has suddenly become quite popular for home-based networks. Finally, there is *Bluetooth*, which is an emerging wireless convergence technology designed to interconnect various devices, including computers, mobile phones, mobile computers, and handheld or portables terminals using short-range (approximately 10 m) radio links. We will discuss Bluetooth in Chapter 17.

95. I heard that another name for IEEE 802.11b is “Wi-Fi.” Is this true?

Wi-Fi, which stands for *wireless fidelity*, is a name that corresponds to IEEE 802.11b networks. Prior to IEEE’s ratification of 802.11b, WLANs operated at 2 Mbps or less and WLAN equipment was expensive. When 802.11b was approved as an IEEE standard, however, WLANs now boasted an 11-Mbps rate. This increase in data rate suddenly made WLANs more popular, which led to products becoming more affordable. Unfortunately, interoperability issues among 802.11b-compliant products became more pronounced. As a result, the Wireless Ethernet Compatibility Alliance (WECA) was formed. WECA’s primary mission is to certify interoperability of IEEE 802.11b wireless products so that products from different vendors will interoperate with one another "out of the box." Products that pass WECA's tests, which are currently performed by Silicon Valley Networking Laboratories, receive a Wi-Fi seal of interoperability. So, although many people equate "Wi-Fi' as another (and more sexy) name for 802.11b, it is simply a trademark. It signifies that an 802.11b device is "true" to the IEEE 802.11b specification and will interoperate with other such devices.

96. What else can you tell me about wireless networking?

There is a tremendous amount of information about wireless networking. There are several books devoted to the topic alone. In keeping with the objectives of this book, we developed a separate wireless networking chapter (Chapter 18), which provides a general overview of the subject. You are also encouraged to consult the references listed in the Bibliography as well as some of the Web sites devoted to wireless networks. These include the Wireless LAN Alliance at http://www.wlana.com, WECA at http://www.wirelessethernet.org, Bluetooth at http://www.bluetooth.com, and IEEE at http://grouper.ieee.org/groups/802/11/main.html.

97. Where does satellite communication fit into the realm of wireless networking?

Satellite communications are based on RF transmissions. Satellite communication systems consist of ground-based (also called terrestrial) stations made up of a parabolic antenna (transmitter/receiver) and orbiting transponders. The transponder receives a microwave signal from the ground unit (this transmission is called an *uplink*), amplifies it, and then transmits it back to Earth (the return signal is called a *downlink*). The higher the altitude of a transponder, the longer it takes to traverse its orbit around the earth. An object located approximately 22,000 miles (36,000 kilometers) above the equator is said to be in a *geosynchronous* orbit or a *geostationary Earth orbit* (*GEO*). A satellite placed at this altitude (called a GEO satellite) traverses its orbit at approximately the same rate as the Earth rotates. Thus, the satellite appears stationary with respect to the Earth's rotation (see Figure 4.20). (*Note:* A geostationary earth orbit is also known as the *Clarke orbit*, named after author Arthur C. Clarke.)

98. What kind of bandwidth are satellite communications capable of?

The bandwidth capability of satellite systems varies and is a function of the frequency at which the satellites transmit. Four common frequencies are *C-band, Ka-band, Ku-band,* and *V-band*. C-band is 6 GHz uplink and 4-GHz downlink. Ka-band is 28-GHz uplink and 18-GHz downlink. Ku-band is 14-GHz uplink and 12-GHz downlink. V-band is above 30 GHz and is still being researched and developed.

99. Doesn't it take time for a signal to travel 22,000 miles up and 22,000 miles down?

Yes it does. Although signals travel at the speed of light, GEO satellite systems have high latency. Consider, for example, a typical satellite transmission involving two remote nodes. The sending node transmits a message to the satellite (sender-satellite uplink), the satellite transmits this message to the destination node (satellite-destination downlink), the

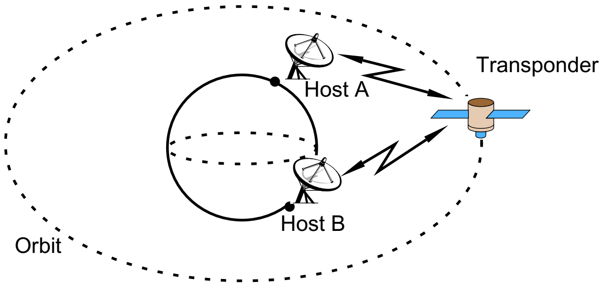


FIGURE 4.20 A satellite communication system. The transponder, a type of repeater, listens to some part of the spectrum. When it hears an incoming signal, it amplifies the signal and then rebroadcasts it at a different frequency. The downward signals can cover a large or narrow area.

destination node sends an acknowledgment to the satellite (destination-satellite uplink), and the satellite transmits the acknowledgment to the sender (satellite-sender downlink). The total distance involved in this transmission is four times 22,000 miles or 88,000 miles. Dividing by the speed of light (186,000 miles per second), this transmission incurs a total *propagation delay* of nearly one-half second (470 milliseconds). GEO satellites can incur propagation delay anywhere from 274 milliseconds to as much as 1050 milliseconds. If you happen to have a home-based satellite system and cable TV, with each system connected to a different television, you probably know what we mean by propagation delay. If both televisions are set to receive the same program, the cable TV will receive its broadcast slightly ahead of the satellite-based TV. A direct consequence of this latency is that the longer it is, the less bandwidth the system can support. In contrast, microwave transmissions have a 3-ms/km delay, and coax has a 5-ms/km delay.

100. How many satellites can be placed in a GEO orbit?

Not many. Since satellites can operate at different frequencies, to prevent frequency interference, satellites cannot be placed any closer than 4° apart at the same altitude. A quick calculation reveals that only 90 satellites can be placed at the same altitude. A constellation of only eight GEO satellites, however, is needed to provide worldwide coverage.

101. Besides propagation delay and the limited number of GEO satellites, what other disadvantages of satellite communications are there?

First, satellite communications are expensive. It costs a lot of money to build and then deliver a satellite to its correct orbit. For example, you would probably not want to contract with NASA to launch a satellite into geosynchronous orbit to network two geographically dispersed sites. Well, you might want to, but your chief financial officer or budget manager might blow a gasket. Second, satellites have a limited operating life, and required maintenance or service can get very expensive. Again, please check with your boss before launching one. Signals from satellite communications can also be interfered with or blocked by buildings, foliage, or various atmospheric conditions. Finally, satellite transmissions are easily susceptible to eavesdropping (that is, they can be monitored by unauthorized personnel). Thus, implementing appropriate security precautions is important.

102. Are GEO satellites the only type used for satellite communications?

No. In addition to GEO satellites, there are *low-earth orbit* (LEO) satellites and *medium-earth orbit* (MEO) satellites. Compared to GEO satellites, which orbit at approximately 22,000 miles, LEO satellites orbit anywhere from 300 miles to 1200 miles. Comparing velocities, the velocity of LEO satellites is greater than that of GEOs, which approximates the earth's orbit rate. Also, only eight GEO satellites are needed to provide global communications. Depending on their orbit, a constellation of up to 48 LEOs is needed for global coverage. Because GEO satellites' orbit matches the earth's, they are always in communication with an earth-based antenna. LEO satellites are only in sight of a terrestrial antenna for at most 15 minutes. Consequently, a LEO-based satellite communication must be transferred from one satellite to another within the constellation. LEO satellite systems also do not suffer from the type of propagation delay found with GEO satellite systems. The propagation delay of LEO satellites ranges from 40 milliseconds to 450 milliseconds, making them better candidates to support interactive applications such as videoconferencing. MEO satellite systems usually orbit from 6000 miles to 12,000 miles and have a latency of approximately 250 milliseconds; MEOs need a constellation of 20 satellites for global coverage.

103. What advantages do satellite communications offer over land-based links?

Probably the biggest advantage of satellite communication is they can reach geographically remote areas. This includes countries that have no or little communications or wiring infrastructures. Another advantage if you own the satellite system (e.g., NASA) is there are no recurring leased line charges. If you do not own your own satellite, then there are recurring transponder fees or charges for leasing satellite time.

104. I've heard about various satellite communications projects that have not been very successful. The two that come to mind are Iridium and Globalstar. What can you tell me about them?

You're right. Several commercial satellite communications projects have recently been proposed or deployed. A brief description of some of the major ones follows.

Iridium Iridium began providing hand-held global satellite-based telephone and paging services on November 1, 1998, but filed for bankruptcy shortly thereafter. Iridium operated under the parent company, Iridium LLC, which was an international consortium headquartered in Washington, DC, and comprised several key telecommunications companies, including Motorola and Sprint. The operating assets of Iridium LLC were purchased by Iridium Satellite LLC in late 2000, and service was restored in Spring 20001. The Iridium system consists of 66 LEO satellites orbiting 485 miles above the earth. Twelve ground station gateways link the Iridium satellite constellation to terrestrial wireless and landline public telephone networks. The satellites, which effectively serve as sky-based cellular towers, enable wireless signals to be transmitted overhead instead of through ground-based cells. Furthermore, the low-earth orbit makes the satellites close enough to the ground station gateways that they are able to receive signals of hand-held devices. This makes Iridium a truly global mobile voice and data satellite system with complete coverage of the earth, including oceans, airways, and polar regions. Iridium subscribers can make and receive telephone calls using their Iridium phones via a single telephone number. Furthermore, subscribers are able to use their Iridium phones in satellite mode when landline wireless cellular service is unavailable, but still operate them in normal cell mode when terrestrial wireless services are available. Finally, if subscribers' phones are turned off, they could still be contacted via their Iridium pager, 24 hours a day, 7 days a week. Additional information about Iridium is available at http://www.iridium.com or by calling toll free in the U.S. at 1-866-947-4348.

Globalstar A second satellite communications project is *Globalstar*, which is led by founding partner, Loral Space & Communications and consists of a consortium of international telecommunications companies. The Globalstar system comprises a constellation of 48 LEO satellites orbiting at an altitude of 876 miles above the earth. Four additional satellites are also in orbit as spares. The 48-satellite constellation was completed in November 1999, and the four backup satellites were launched in January 2000. The constellation's design involves eight orbital planes of six satellites each, inclined at 52°. This configuration enables Globalstar to provide earth-based service from 70° north latitude to 70° south latitude. (Consult an atlas to identify areas that are and are not covered.) Furthermore, unlike other satellite communication systems, Globalstar software resides on the ground, not on the satellites. The Globalstar system delivers satellite telephony services through a network of exclusive service providers. Subscribers include cellular users who roam outside their coverage areas, people who work in remote areas where terrestrial systems do not exist, residents of underserved markets, and international travelers who need to keep in constant touch. Globalstar enables subscribers to use a single phone for both cellular and satellite calls and provides an array of services via a Globalstar phone, including traditional voice calls, short messaging service, global roaming, facsimile, and data transmission. Additional information about Globalstar can be found at http://www.globalstar.com.

Teledesic The Teledesic network project, which has Bill Gates as one of its partners, comprises 840 interlinked LEO satellites that are to provide global access to voice, data, and video communications beginning in the year 2002. It is supposed to support 1 million full-duplex E1 (2.048 Mbps) connections and have a capacity that will support millions of simultaneous users.

Metro-Area Satellite A fourth and final project we will mention is a "metro-area satellite" project by Angel Technologies, a privately held St. Louis-based company. Using piloted FAA-certified High Altitude Long Operation (HALO) aircraft flying 52,000 feet over cities, Angel Technologies intends to deploy wireless broadband "super-metropolitan area" networks on a city-by-city basis that will interconnect subscribers at multimegabit per second data rates. Equipped with a telecommunications payload, a HALO aircraft serves as the centralized hub of a wireless, line-of-sight, star-based topology network. Thus, connectivity between two hosts connected to the HALO network is via a single hop. Connectivity between HALO hosts and non-HALO hosts will be provided via dedicated

HALO gateways connected to public switched networks. Compared to conventional high-speed wireless networks, the deployment of a HALO network within a metropolitan region does not require attention to matters such as roof rights negotiations for tower placement, compliance with local zoning laws, or tower construction. A HALO network can begin providing service to its subscribers almost immediately to anyone within its coverage area, which ranges from 50 miles to 75 miles in diameter. Commercial service, which began in 2001, is expected to provide businesses with broadband services at data rates ranging from 5 Mbps to 25 Mbps and higher. Individual consumers can also subscribe to the HALO network for Internet access and network-based entertainment services at data rates ranging initially from 1 Mbps to 5 Mbps, with higher data rates available as the broadband market grows. Additional information about the HALO network can be found at http://www.angelcorp.com.

END-OF-CHAPTER COMMENTARY

You should now have a good understanding of the fundamental concepts and terms related to the physical layer and the different types of transmission media (both wired and wireless) used at the physical layer. Many of these concepts are discussed in later chapters in specific contexts. In Chapters 7, 8, 9, 10, and 14, the various copper and fiber-optic cables presented here are discussed from the perspective of their specific LAN applications; in Chapter 6, key physical layer concepts are applied to specific physical layer hardware components. In Chapter 8, we embellish our discussion of post-Category 5 cable when we introduce Gigabit Ethernet and the latest "gigabit copper" cabling designed to support gigabit speeds. Chapter 15 extends the discussion of wireless networking by examining its implementation in home-based networks, and Chapter 18 provides additional information about wireless networking. Finally, Appendix C contains guidelines for installing UTP cable. Our next order of business is the data link layer, which is the subject of Chapter 5.

Chapter 5

Data Link Layer Concepts and IEEE Standards

In this chapter, we discuss the second layer of the OSI model, namely, the data link layer. This layer handles the transfer of data between the ends of a physical link—it is responsible for transferring data from the network layer on the source machine to the network layer on the destination machine. We begin the chapter with an overview of the data link layer and provide information about IEEE, which developed data link layer protocols that serve as the basis for various LAN technologies such as Ethernet and token ring. In the remaining part of the chapter, we examine the data link layer from IEEE's perspective. We also include at the end of the chapter information about the concepts of data prioritization and quality of service (QoS), which are integral to the delivery of time-sensitive data such as real-time voice and video traffic. An outline of the terms and concepts we define and discuss follows:

- Data Link Layer Overview and IEEE's Perspective (Questions 1–6)
- Framing (Questions 7–11)
- Ethernet/802.3 Frames (Questions 12–15, 44–45)
- Flow Control and Flow Control Protocols (Questions 16–27)
- Error Control (Questions 28–43)
- MAC Sublayer (Questions 46–52)
- Random Access and Token Passing Protocols (Questions 53–58)
- Data Prioritization and Quality of Service (QoS) (Questions 59–69)

1. What is the data link layer?

In network architecture, the purpose of the data link layer is to regulate and format transmission of data from software on a node to the network cabling facilities. The data link layer is the "glue" between the wire and the software on a node. Without it, the particular network connection will not operate at all. The data link layer creates the network environment for the wire and dictates data formats, timing, bit sequencing, and many other activities for each particular type of network.

As the second layer of the OSI reference model, the data link layer provides a service to the network layer (layer 3) using the services of the physical layer (layer 1). Some of the services the data link layer provides to the network layer include:

- provisioning links between network entities (generally these are adjacent nodes within a subnetwork);
- *framing*, which involves partitioning data into frames with recognized frame boundaries and exchanging these frames over the link;
- *frame sequencing* (if required), which involves maintaining the correct ordering of frames as they are being exchanged;
- establishing and maintaining an acceptable level of *flow control* as frames are being exchanged across a link;
- detecting (and possibly correcting) errors in the physical layer, which includes error notification when errors are detected but not corrected;
- selecting *quality of services* (QoS) parameters associated with a specific transmission, including ensuring that sufficient bandwidth is available and that transmission delays (i.e., latency) are predictable and guaranteed.

In short, the data link layer enables data frames to be transmitted error-free between two end nodes over the physical layer.

2. What is the IEEE and what does it have to do with local area networks?

In the early days of local area network development, there were no standards for LANs. Chaos and instability were the order of the day. Proprietary vendor standards ruled, customers became customers for life, and companies got fat. The dearth of industrywide standards effectively prevented customers from using "outside" products for fear of incompatibility. In February 1980, the IEEE (pronounced "eye triple E") assumed responsibility for setting LAN standards, primarily for the physical and data link layers, using the OSI reference model as a framework. IEEE, which stands for the Institute of Electrical and Electronics Engineers, is a professional society founded in 1963. IEEE members include engineers, scientists, and students. One of its many activities is to act as a coordinating body for computing and communication standards. Many international standards from the International Organization for Standardization (ISO) and the International Electrotechnical Commission (IEC) are based on IEEE networking standards. For additional information about these organizations, see http://www.ieee.org; http://www.iso.ch; and http://www.iec.ch. IEEE conducted its LAN standards development under the auspices of the IEEE Computer Society. A list of these standards is provided in Table 5.1, with corresponding ISO/IEC standards given in parentheses.

3. Looking at Table 5.1, I notice that all of the IEEE standards start with 802. Why?

IEEE's development of LAN standards was assigned the project number 802, for February 1980 (get it? 2/80 or 802), and the committee's collective body of work has become known as Project 802. The standards that have resulted are identified as IEEE 802.*x*.

TABLE 5.1 Summary of Various IEEE Project 802 LAN Standards

IEEE 802.1	Defines an architectural overview of LANs.
IEEE 802.2	Defines the Logical Link Control, which describes services for the transmission of data between two nodes. (ISO/IEC 8802-2)
IEEE 802.3	Defines the Carrier Sense Multiple Access/Collision Detection (CSMA/CD) access method commonly referred to as Ethernet. Supplements include 802.3c (10 Mbps Ethernet); 802.3u (100 Mbps Ethernet known as *Fast Ethernet*), 802.3z and 802.3ab (1000 Mbps Ethernet known as *Gigabit Ethernet*), and 802.3ae (*10-Gigabit Ethernet*). (ISO/IEC 8802-3)
IEEE 802.4	Defines the token bus network access method.
IEEE 802.5	Defines the logical ring LAN that uses a token-passing access method; known also as token ring. (ISO/IEC 8802-5)
IEEE 802.6	Defines metropolitan area networks (MANs).
IEEE 802.7	Defines broadband LANs (capable of delivering video, data, and voice traffic).
IEEE 802.9	Defines integrated digital and video networking—Integrated Services LANs (ISLANs). (ISO/IEC 8802-9)
IEEE 802.10	Defines standards for interoperable LAN/MAN security services.
IEEE 802.11	Defines standards for wireless media access control and physical layer specifications.
IEEE 802.12	Defines the "demand priority" access method for 100Mbps LANs; known also as 100 Base-VG or *100VG-AnyLAN*.
IEEE 802.13	Defines nothing—IEEE was concerned about the superstitious overtones associated with "13."
IEEE 802.14	Defines a standard for Cable-TV based broadband communication.
IEEE 802.16	Defines standards for the wireless local loop (WLL).

4. Enough of the history lesson. What does this have to do with the data link layer?

IEEE initiated its development of LAN standards with an architectural model, defined in IEEE 802.1. This architectural model corresponds to the two lowest layers of the OSI Model. The difference between the IEEE and OSI models, though, is that the IEEE divides OSI's data link layer into two parts—the *logical link control sublayer* (LLC) and the *medium access control sublayer* (MAC) (see Figure 5.1). Note that MAC has nothing to do with Apple Computer's Macintosh.

5. How is the data link layer implemented within a network?

The data link layer is typically implemented on a node as a *device driver.* A device driver is a software component that is specific to both a piece of hardware, such as an Ethernet/802.3 network interface controller card (NIC), and the operating system of the computer in which it is installed. For instance, the data link layer of Ethernet/802.3 could be a card you purchased from a network vendor such as SMC, 3Com, or Intel. The vendor includes a device driver with the card to enable the operating system on your computer (such as Windows, MacOS, NT, or UNIX) to recognize the card and allow the software protocol(s) to "talk" to the card when they are accessed.

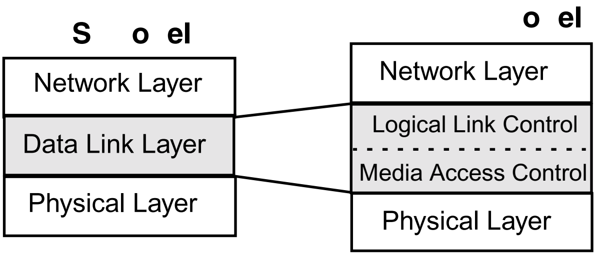


FIGURE 5.1 The IEEE's perspective of the data link layer.

6. So what do the sublayers within the data link layer do?

The LLC sublayer, defined in IEEE 802.2, is the upper half of the data link layer. It encompasses several functions, including framing, flow control, and error control. The MAC sublayer is the lower half of the data link layer. It provides media access management protocols for accessing a shared medium.

7. Huh? Could you please give me a little more detail? Why don't you start with the LLC sublayer and framing. What is framing?

Framing refers to the process of partitioning a bit stream into discrete units or blocks of data called *frames*. Thus, it is the manner in which a specific network type formats the bits sent to the cable. Specific formats and timing sequences exist for each LAN type.

8. Why is it necessary to partition a bit stream into frames?

Framing enables sending and receiving machines to synchronize the transmission and reception of data because frames have detectable boundaries. Framing also facilitates error detection and correction. Once a bit stream has been partitioned into frames, specific information about the contents of a frame is computed and transmitted within the frame. Using this information, a receiving node can determine the integrity of a received frame.

9. How is framing accomplished?

A common procedure involves inserting start and stop bits before and after the transmitted data message. For example, if we use the bit pattern 01111110 (six consecutive 1-bits) as our start-stop flag, and if the data set (i.e., the message being transmitted) consists of 11101111, then the frame is 011111101110111101111110. (The framing protocol associated with each type of network specifies the actual bit pattern used for these start-stop flags.) Thus, the data set has been "framed" by distinct boundaries consisting of start-stop flags.

10. What happens if a data message has the same pattern as the start-stop bits? How do we distinguish real user data from the start-stop bits?

Aha! Good thinking! Because the start-stop flags must be distinguishable from user data, they must be unique. If a data message contains a sequence of bits identical to the

Data set to be transmitted:	1111110011111011
Data set after bit-stuffing:	11111 10011111 011
Data set after bit-stuffing and start-stop bits have been inserted:	11111 10011111 011

Thus, the frame to be transmitted is:

01111110	1111101001111110011	01111110
Start of Frame	*User Data with Bit-Stuffing*	*End of Frame*

The data link layer on the receiving machine removes the start and stop bits and unstuffs the data set by removing the 0-bits that follow each set of five consecutive 1-bits.

FIGURE 5.2 An example of bit-stuffing, which is used for framing when the data set contains the start-stop bits pattern.

start-stop flag, then we must alter the data set to guarantee the uniqueness of the start-stop bit patterns. One way to do this employs a process known as *bit stuffing*. For example, suppose a data set consists of the bit string 0111111001111101, and our start-stop flag is 01111110. Note that the data message includes one instance of our start-stop flag. We implement bit stuffing by "stuffing" a 0-bit immediately after every fifth consecutive 1-bit in the data stream. The receiving node "unstuffs" these 0-bits by deleting every 0-bit that follows five consecutive 1-bits. We illustrate bit stuffing in Figure 5.2.

11. Can you give me an example of a "real" frame?

Sure thing. How about an Ethernet frame?

12. Good. Show me an Ethernet frame.

Before showing an Ethernet frame, we need to make a distinction between Ethernet and IEEE 802.3, which many people call Ethernet. This distinction is necessary because Ethernet and IEEE 802.3 frame formats are different. Instead of getting into a detailed discussion about these differences (we save that for Chapter 8), just be aware that Ethernet and IEEE 802.3 are not really the same. IEEE 802.3 is broadly deployed but old habits die hard and people continue to call it Ethernet. To avoid confusion, or to create more if you're so inclined, we use the nomenclature Ethernet/802.3 in this book when we want to refer to what people commonly call Ethernet.

13. There you go again, throwing a monkey wrench into the works. So show me the IEEE 802.3 frame, already.

Let's look at the general format of an IEEE 802.3 frame, shown in Figure 5.3. Note that an IEEE 802.3 frame consists of eight fields: preamble, start frame delimiter, destination address, source address, length count, data, pad, and CRC checksum (see below). The preamble, used for synchronization, consists of seven identical bytes (56 bits). Each byte

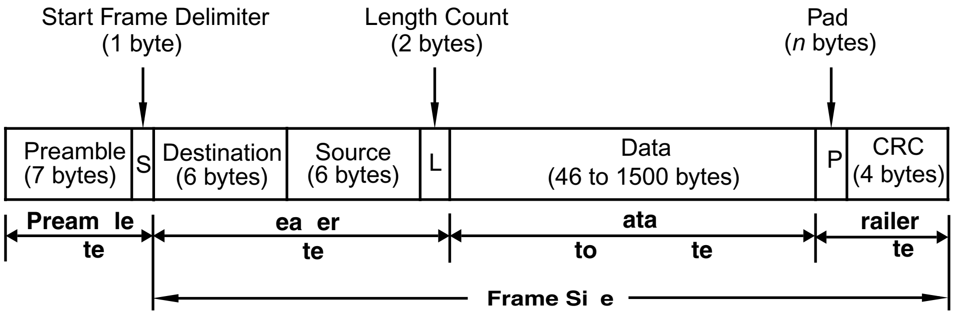


FIGURE 5.3 The contents and structure of an IEEE 802.3 frame.

(8 bits or octet) has the bit pattern 10101010. The start frame delimiter (S), which indicates the start of a frame of data, consists of the bit pattern 10101011. This is the start-stop flag used in IEEE 802.3. The destination address is the hardware address of the receiving station—normally 48 bits (the standard provides for a 16-bit value, but no one uses it). The source address is the hardware address of the sending station, also 48 bits in length. Both of these addresses are MAC sublayer addresses, which we will discuss shortly. (Also see Appendix A.) The length count (L) is a two-byte field that indicates the length of the data field that follows. The data field contains the actual user data (i.e., the transmitted message). The data field is subject to minimum and maximum sizes, which are 46 and 1500 bytes, respectively. The pad field (P) contains dummy data that "pads" the data field up to its minimum length of 46 bytes, if necessary. Its size ranges from 0 to *n* bytes, where *n* is the number of bytes needed. Finally, the checksum field (CRC) contains the information needed for error detection, which is yet another task performed by the data link layer in the LLC sublayer. It is also called the Frame Check Sequence (FCS) and it is four bytes long. (*Note:* CRC stands for Cyclic Redundancy Check and is discussed later in the chapter.)

14. Now I know what an IEEE 802.3 frame is all about. I have another question, though. I have seen a protocol analyzer display frames with only four fields instead of eight, and you also show four fields in Figure 5.3. Why is that?

Obviously, you bought a cheap protocol analyzer. Seriously, though, it's because the eight individual fields of an IEEE 802.3 frame can be grouped into four main fields: *preamble* (preamble plus start frame delimiter), *header* (source and destination addresses plus length count), *data* (user data plus padding), and *trailer* (the CRC checksum). If we ignore the preamble, the aggregate of the header, data, and trailer fields ranges from a minimum of 64 bytes to a maximum of 1518 bytes. These represent the minimum and maximum sizes of an IEEE 802.3 frame.

15. In your response to Question 13, you stated that the source and destination addresses of an IEEE 802.3 frame are "MAC sublayer addresses," but you also called them "hardware addresses." Please elaborate.

Certainly. We explain this in Box 5.1. As you read this box, it is important to note that a MAC sublayer address is not the same as an Internet address. The *MAC sublayer address* is nothing more than the hardware address of a particular node. If the node is connected to an Ethernet/802.3 network, then we refer to the hardware address as the Ethernet/802.3 address. If the node is connected to a Token ring network, then the hardware address is called the *token ring address*. It all depends on the MAC sublayer protocol. You also should not confuse a MAC address as the address of a Macintosh computer. Information about Ethernet/IEEE 802.3 vendor prefixes is given in Appendix A.

16. Yes! It's starting to making sense. I've seen this terminology before and I think I'm starting to understand it. Can we move on to the concept of flow control?

Sure. *Flow control*, which is another function of the LLC sublayer, refers to a process that controls the rate at which data are exchanged between two nodes. Flow control involves a feedback mechanism that informs the source machine of the destination machine's ability to keep up with the current flow of data transmission. Usually, a source node may not transmit frames until it receives permission from the destination machine.

17. Why is flow control important?

Flow control is necessary because it is possible for a sending node to transmit frames at a rate faster than the destination node can receive and process them. This can happen if the source node is lightly loaded and the destination node is heavily loaded, or if the source node has a faster processor than that of the destination node. Thus, flow control provides a mechanism to ensure that a sending node does not overwhelm a receiving node during data transmission. For example, consider a network that consists of several 80486 PC clients and a Pentium PC file server. Clearly, the processing speed of the server is much greater than that of the clients. In this scenario, the server is able to transmit frames to a client faster than the client can process them. Without flow control, the client's buffers will eventually fill with "backlogged" frames sent by the server. With its buffers full, the client will discard any subsequent data it receives. This can result in retransmissions by the sending node, which can exacerbate network congestion.

18. Can you give me an example of a flow control protocol?

You bet. A very simple flow-control protocol is the *stop-and-wait protocol*. As inferred by its name, stop-and-wait requires the sender to transmit one frame and then wait for the receiver to acknowledge receipt of this frame. The acknowledgment sent by the receiver is a basic frame (i.e., it does not have to carry any user data) that simply informs the sender that the receiver is now ready to accept another data frame. As a result, a receiver effects flow control via acknowledgments. A sender must wait until it receives an acknowledgment from the receiver for the frame it transmitted before it is permitted to transmit another frame. If a receiver withholds an acknowledgment, then the flow of data between sender and receiver stops.

BOX 5.1: IEEE 802.3 MAC Sublayer Addresses

Ethernet addresses, also known as hardware or MAC addresses, are installed in ROM by the manufacturers of Ethernet controllers (i.e., the Ethernet network cards). The custodian of these addresses is the IEEE, which assigns addresses on a group basis to vendors who manufacture Ethernet hardware devices used in a heterogeneous network environment.

Ethernet addresses consist of 48 bits, represented as 12 hexadecimal digits (0–9, A–F), and partitioned into six groups of two. An example of an Ethernet address is 08:00:20:01:D6:2A. Dashes are sometimes used in place of colons, and sometimes no delimiter is used to separate the byte-pairs. Also, the letters A–F are used to represent the numbers 10–15, respectively. These letters are usually written in uppercase. The higher-order three bytes (i.e., the leftmost six hexadecimal digits) correspond to the manufacturer of the Ethernet device, and the lower-order three bytes correspond to the serial number of the device. For example, the address 08:00:20:01:D6:2A corresponds to an Ethernet controller manufactured by Sun Microsystems because the leftmost three bytes, 08:00:20, have been assigned to Sun; the remaining three bytes specify a unique serial number assigned by Sun to the device. Other examples of vendor prefixes include 00:00:0c (Cisco), 00:00:1D (Cabletron), and 08:00:07 (Apple). (See Appendix A for more information about Ethernet vendor prefixes.)

There are three different types of destination addresses. The first is called a *unicast address*, which essentially means that data are being transmitted to only a single network interface (i.e., device). Unicast destination addresses are denoted by either a 0 or an even number (2, 4, 6, 8, A, C, E) in the second hexadecimal digit of the vendor prefix. A *multicast* or *broadcast address* enables data to be transmitted to more than one destination node at the same time. A multicast address specifies a vendor-specific group of nodes. For example, the multicast address, 09-00-87-90-FF-FF, is for Xyplex terminal servers. Ethernet frames that have their destination fields set to this address will be received by all Xyplex terminal servers connected to the network. If a node wants to send data to *all* network connected devices, then a broadcast address is used. This address is given as FF:FF:FF:FF:FF:FF. Multicast and broadcast Ethernet addresses are denoted by using an odd number (1, 3, 5, 7, 9, B, D, F) as the second hexadecimal digit.

Ethernet addresses should not be confused with Internet addresses. The former are six-byte addresses that have been "burned" into the controller card; the latter are logical addresses that correspond to a specific network protocol and are assigned by a network administrator and configured through software. If a device is connected to an Ethernet network, which in turn is connected to the Internet, then the device will have both Ethernet and IP addresses assigned to it. The Address Resolution Protocol (ARP), which is part of the TCP/IP protocol suite (see Chapters 2 and 3), is used to resolve IP addresses to Ethernet addresses. An ARP request is a network broadcast that announces the target node's IP address and requests the node to return its MAC address. On UNIX and Windows NT systems, this protocol can be invoked on the command line by the *arp* application program. Using the "a" extension (*arp -a*) displays the contents of the current ARP table maintained by the local host.

19. This is pretty straightforward. Is this what is used in most networks today?

Yes, the stop-and-wait protocol is very straightforward and it is also very effective. However, it is not very practical for modern networking environments and hence is rarely (if at all) implemented. First, as described, stop-and-wait uses a simplex transmission; data frames flow in only one direction. In most data communication environments, data transmission is full-duplex. Second, the protocol is ideal when transmitting large frames. Unfortunately, large frames are generally partitioned into smaller data units to accommodate a receiver's limited buffer size. Small frame sizes also facilitate faster error detection and reduce the amount of data that requires retransmission in the event that an error is detected. (Error-control concepts are discussed later in the chapter.)

20. Well, then, how about a more practical example?

OK. An enhancement to the stop-and-wait protocol is the *sliding window concept*, which improves data flow by having the receiver inform the sender of its available buffer space. The sliding window concept improves data flow by having the receiver inform the sender of its available buffer space. Doing so enables the sender to transmit frames continuously without having to wait for acknowledgments to these frames as long as the number of frames sent does not overflow the receiver's buffers. The sliding window concept is implemented by requiring the sender to sequentially number each data frame it sends and by having the sender and receiver maintain information about the number of frames they can respectively send or receive. Flow-control protocols based on this concept are called *sliding window protocols* and are used in one form or another in many contemporary networking applications.

21. How does this protocol work?

Well, first of all, you need to understand that the protocol requires frames to be numbered using a *modulo* numbering system that is based on the size of the protocol's window field.

22. Hold it! What is a modulo system and how is it used to number frames?

A modulo system is a mathematical system that cyclically repeats itself. For example, the set $\{0, 1, 2, 3\}$ is a finite set that contains only the four given elements, namely, 0, 1, 2, and 3. This means that if we were to work with these set elements in sequence, after using the element 3, we must cycle back and start with 0. In general, a mathematical system that consists of the set of elements $\{0, 1, 2, 3, \ldots, m - 1\}$ is called a *modulo* or *mod m system*. Most sliding window protocols support either a three-bit or seven-bit window field. A three-bit field specifies a mod 8 system because $2^3 = 8$, and a seven-bit field specifies a mod 128 system because $2^7 = 128$.

23. Does this mean that a three-bit window field uses sequence numbers from 0 to 7, and a seven-bit window field uses sequence numbers from 0 to 127?

You got it! That's exactly the case. With a three-bit window field, frames are numbered modulo 8. Thus, after a frame is assigned sequence number 7, the next frame is numbered

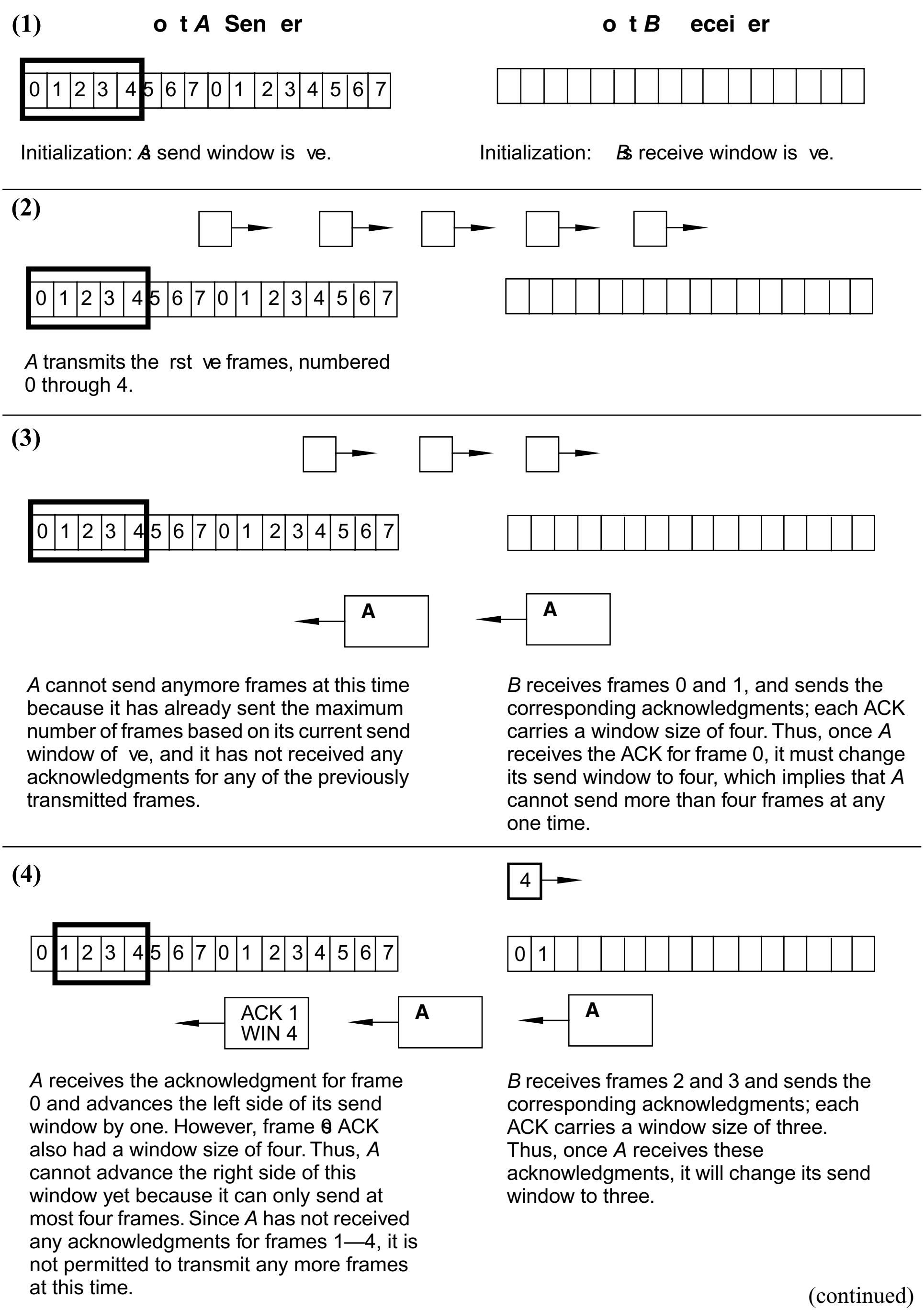
0. Similarly, in a protocol that uses a seven-bit window field, frames are numbered modulo 128 and use sequence numbers ranging from 0 to 127.

As an illustration, look at Figure 3.15, which depicts the format and contents of a TCP header. Recall from Chapter 3 that TCP is the Internet's transport control protocol, which provides reliable data transmissions across the Internet between two end nodes. Note that the TCP header contains a two-byte window field. This field specifies the maximum number of bytes the receiving host is capable of accepting. Thus, this field is what regulates end-to-end flow control; it enables the receiver to control the flow of bytes from the sender. As we indicated earlier, the amount of data the sender transmits can never exceed the window size established by the receiver. The receiver can change the window size anytime during an established connection as additional buffer space becomes available. A window size of 0 informs the sender to stop transmission until it receives a non-0 window size. Since TCP supports full-duplex, both sender and receiver may provide flow-control information via the window field.

24. Is it possible for you to demonstrate how this is done?

Yes, we can do this, but we will keep things simple to help facilitate understanding. We provide a general example of how the sliding window concept is used for flow control in Figure 5.4, which shows a simplex data transmission from host *A* (the sender) to host *B* (the receiver). We assume a three-bit window field. An explanation of how this concept is implemented follows:

- In part 1, an initialization process takes place in which host *A* learns of host *B*'s buffer size. In this illustration, *B*'s receive buffer space is fixed at five, which means that *B* can store at most five frames at any one time before it runs out of storage space. To prevent *A* from overrunning *B*'s buffer space, *A* maintains a list of consecutive sequence numbers that correspond to the frames it may send. Since *B* cannot receive more than five frames at a time, *A*'s initial send window is of size five and consists of the frame sequence numbers 0 through 4, which respectively correspond to the five frames it may send.
- In part 2, *A* transmits frames 0 through 4 without acknowledgments.
- In part 3, *B* receives frames 0 and 1 and sends an acknowledgment for each of them. These acknowledgments also contain information about *B*'s buffer size. Here, *B* informs *A* that its receive window is now four. Note that *A*'s transmission is effectively throttled by its send window. *A* has already transmitted all the frames it was permitted to send. Since no acknowledgments have been received yet, *A* cannot send any more data.
- In part 4, *A* receives an acknowledgment for frame 0. As a result, *A* may advance the left side of its send window by one. However, *A* cannot advance the right side of this window because *B*'s new receive window size is four, which was indicated by frame 0's ACK. In the meanwhile, *B* receives and acknowledges frames 2 and 3 but, once again, reduces its receive window size to three.
- In part 5, *A* receives frame 1's acknowledgment and advances the left side of its send window one unit to the right to indicate that frame 1 was received by *B*.



(continued)

FIGURE 5.4 The sliding window concept used for flow control.

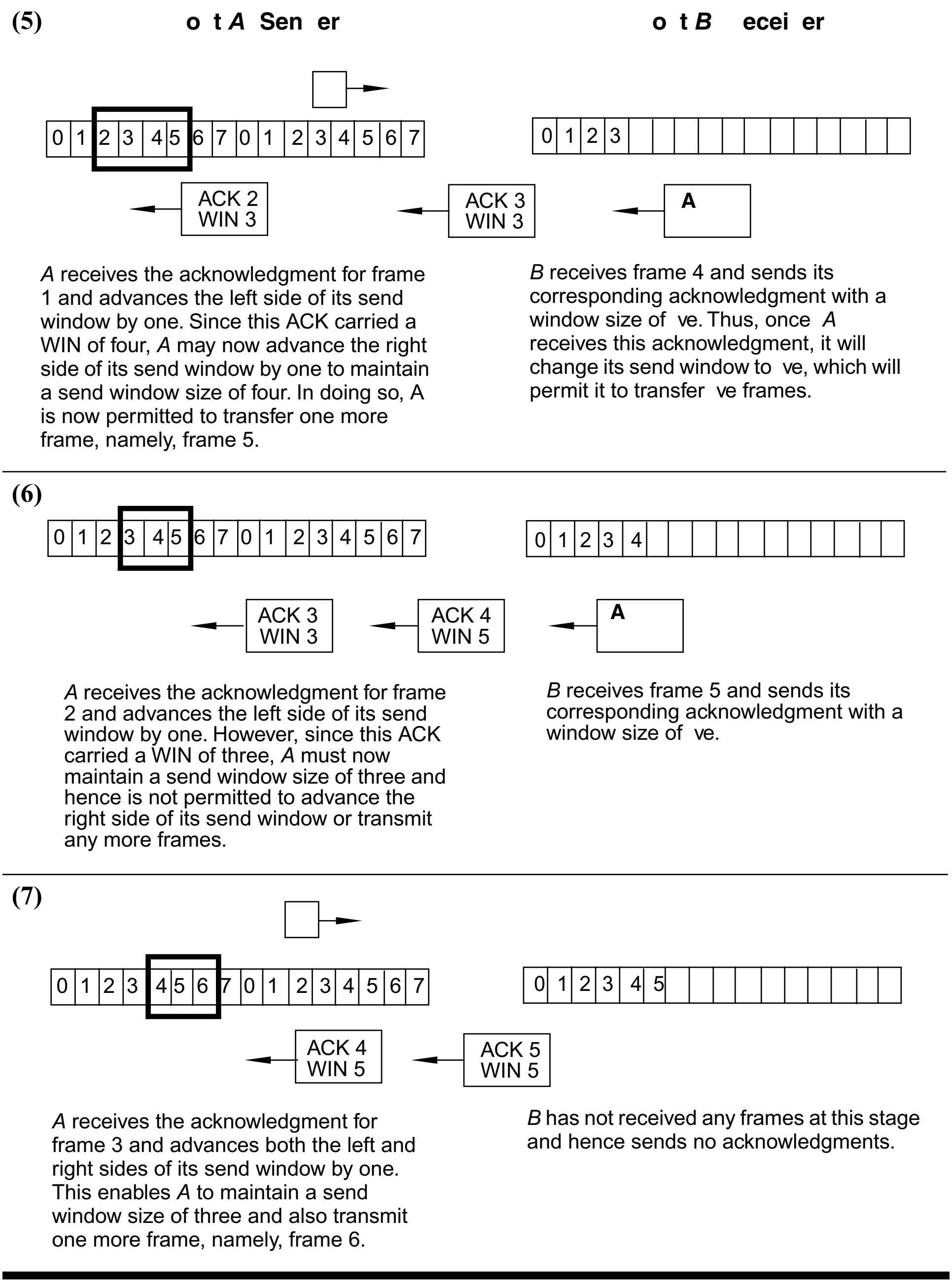


FIGURE 5.4 (continued). The sliding window concept used for flow control.

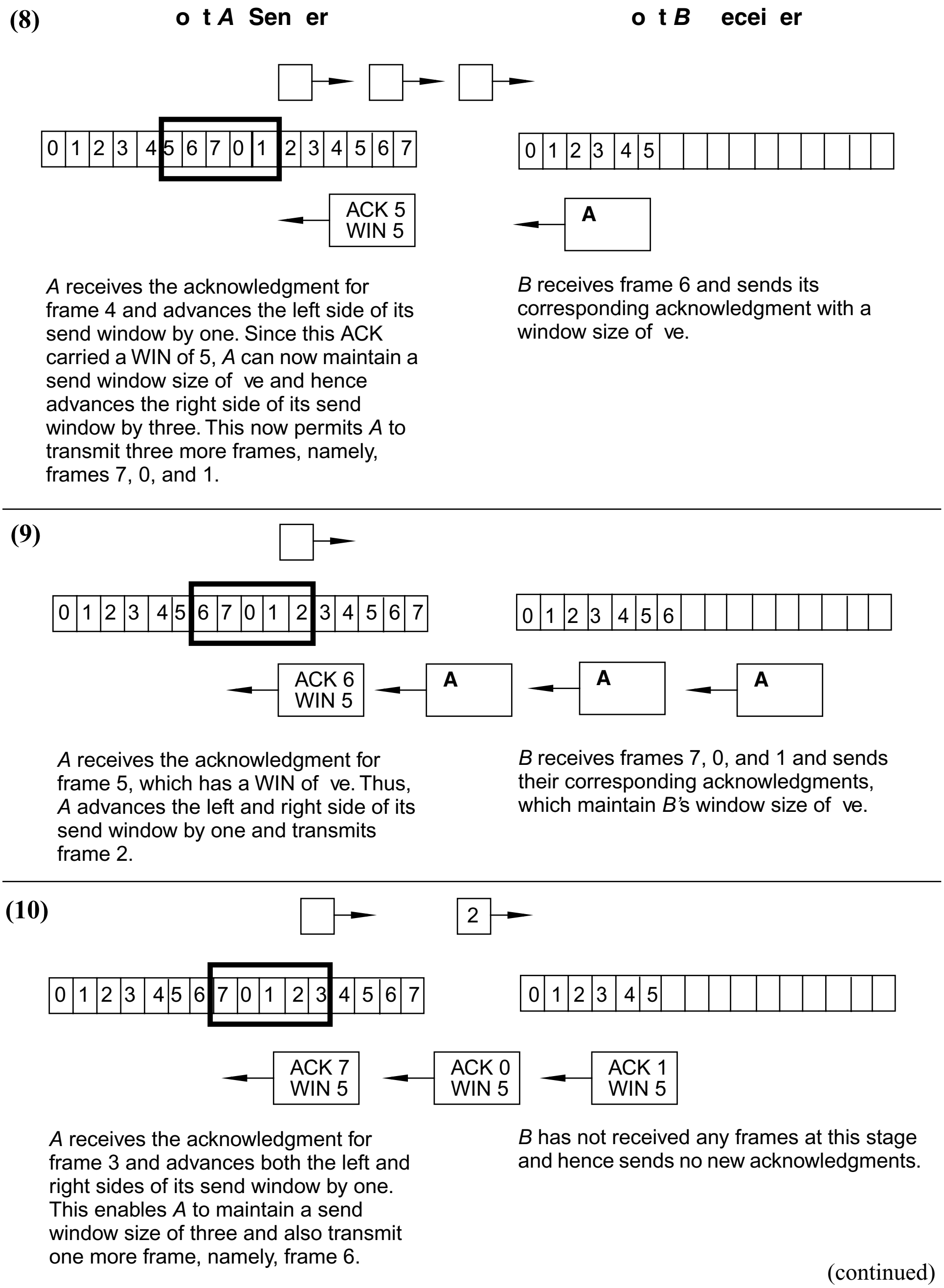


FIGURE 5.4 (continued). The sliding window concept used for flow control.

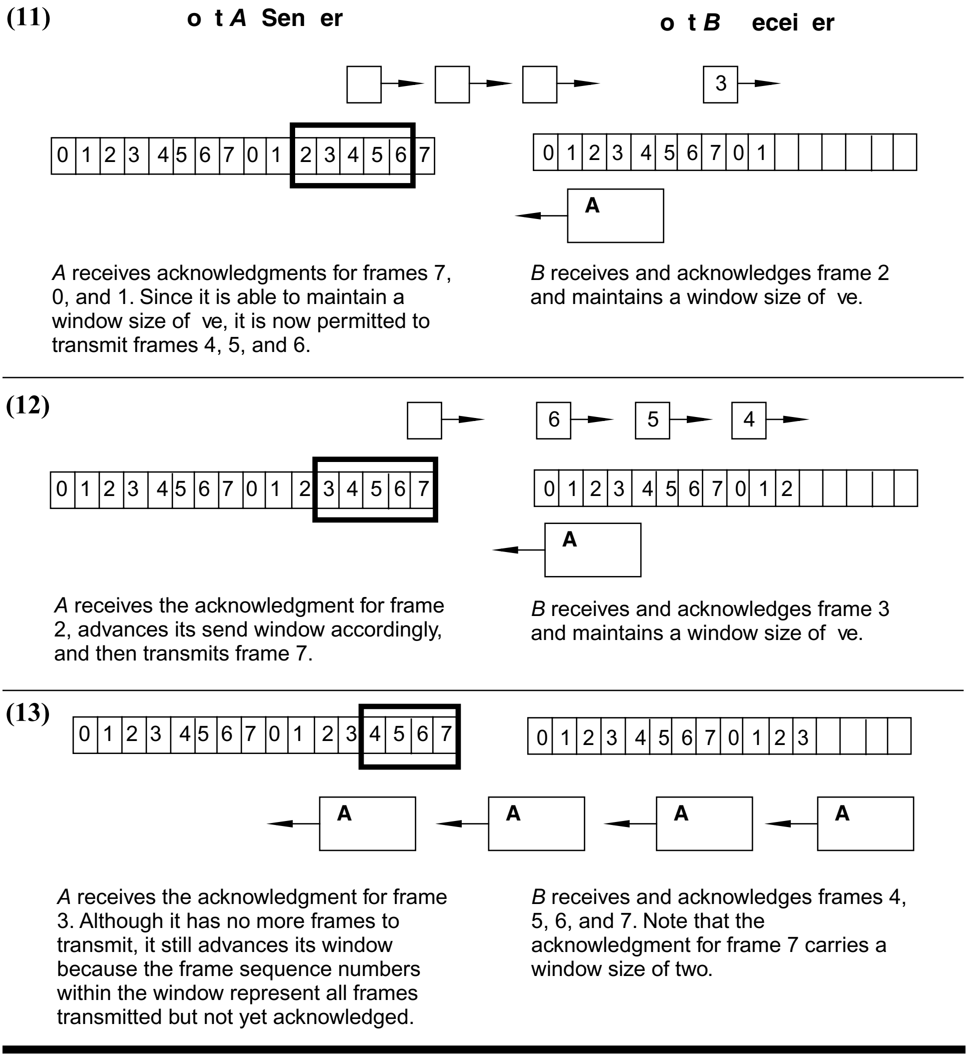


FIGURE 5.4 (continued). The sliding window concept used for flow control.

When *A* does this, notice that its send window is now at three. Since *B*'s receive window is currently at four, *A* is permitted to advance the right side of its send window one unit, which enables it to now transmit frame 5. At the other end of the link, *B* receives and acknowledges frame 4. *B* also, once again, changes its receive window—this time back to five.

- In part 6, *A* receives frame 2's acknowledgment and advances the left side of its send window accordingly. However, since this ACK carried a receive window of three, *A* must now maintain a send window of three and hence is not permitted to transmit any more data. *B* receives and acknowledges frame 5 and maintains a receive window of five.

- In part 7, *A* receives frame 3's acknowledgment and advances the left side of its send window accordingly. *A* can also slide the right side of its window one unit to the right and still maintain a window size of three. This enables *A* to transmit frame 6.
- In part 8, *A* receives frame 4's acknowledgment, which carries a receive window size of five. *B*'s throttling of *A* has now been loosened, and *A* may slide its window to the right to reflect the acknowledgment it received and to maintain a send window of five. Thus, *A* is permitted to transmit frames 7, 0, and 1. (Recall that we are assuming a three-bit window field, which indicates a modulo 8 numbering system.) *B* receives and acknowledges frame 6 and continues to maintain a receive window of five.
- In parts 9 through 13, the sliding window concept continues to play out until *A* has completed transmitting all 16 frames.

25. Pretty neat!

Yes. As can be seen from this simple example, the sliding window concept effectively enables a receiver to control the sender's data flow. Note that when the sender advances the left side of its send window by one after it receives the corresponding acknowledgment, the send window represents a list of frame numbers that correspond to all transmitted but unacknowledged frames. Further note that if the maximum window size is one, then we have the stop-and-wait protocol since the sender cannot send a succeeding frame until the previous frame is acknowledged.

26. You said that this was a simple illustration. How different is this from reality?

In our illustration of the sliding window concept, we took some liberties to facilitate its understanding. In reality, protocols based on this concept are not quite as straightforward. First, although not shown in our demonstration, a receiver's window size can be fixed. That is, it remains at its initial size. Second, some sliding window protocols do not send an acknowledgment for each frame received. Instead, one acknowledgment is sent that contains the next expected sequence number. For example, in Figure 5.4, part 6, instead of sending separate ACKs for frames 0–5, *B* would send one ACK that contains sequence number 6. This implies that *B* has correctly received all frames up to and including 5 and is expecting frame 6. Third, our illustration does not accommodate for lost frames or frames received out of order, which are two common occurrences in any data transmission facility. Thus, the sender must always maintain sufficient buffer space to store all transmitted frames until they are acknowledged in the event a frame has to be retransmitted. Similarly, a receiver must maintain sufficient buffer space to store frames that arrive out of order. Fourth, our illustration in Figure 5.4 depicts a simplex transmission. In most situations, a full-duplex operation is common. This enables both ends of a communications link to send and receive data frames at the same time. Finally, sliding window protocols can also be designed to facilitate reliable data transfers and ordered frame delivery in addition to flow control.

27. I see. It was noted that TCP provides a full-duplex connection. How is the sliding window protocol implemented in full-duplex mode?

Full-duplex data transmission is implemented by having nodes attach acknowledgments to their data frames, a concept called *piggybacking*. Thus, instead of sending separate acknowledgment frames, a host waits until it has data to send and includes this acknowledgment with its outgoing data frame. A full-duplex transmission requires each host to maintain send and receive windows since a host can simultaneously be a sender and receiver. Also, the manner in which TCP uses the window field to provide flow control represents a modified version of the simple sliding window protocol. In TCP's version, the window size is dynamic—it varies over time with each transmitted segment. As a receiver's buffer fills, it advertises a smaller window size; as more space becomes available, it advertises a larger window size. All the while, the sender is calculating the maximum amount of data it can transmit. A variable size window field improves flow control and facilitates reliability.

28. How about error control? You mentioned error detection in the description of the IEEE 802.3 frame. Is this part of error control?

Yes it is. The term *error control* refers to the process of guaranteeing reliable data delivery. That is, the data received are identical to the data transmitted. Two basic strategies exist for dealing with errors. The first method, *error correction through retransmission*, involves providing just enough information in the data stream so the receiving node can detect an error occurred during transmission. Once an error is detected, the receiving node can then request the sender to retransmit that unit of data. The second method, *autonomous error correction*, involves providing redundant information in the data stream so the destination node can both detect and correct any errors autonomously. Both methods are forms of *error correction*.

29. How does the sending node know if it has to retransmit a frame?

Error control using retransmission involves the use of acknowledgments—the receiving node provides the sending node with feedback about the frames it has received. A positive acknowledgment means a frame was received correctly; a negative acknowledgment (or no acknowledgment) implies a frame was not received correctly. Negative acknowledgments imply that the sending node needs to retransmit the frame.

30. What safeguards does error control employ?

To guard against the possibility of lost or destroyed frames (due to hardware failure), the data link layer also supports timers. If a frame or acknowledgment is lost, the sending node's time limit eventually expires, alerting it to retransmit the frame. To guard against a destination node accepting duplicate frames, outgoing frames are assigned sequence numbers, which enable a destination node to distinguish between a frame that has been retransmitted from an original one. The management of timers and sequence numbers is an important function of the data link layer because it ensures that each frame is received by the network layer exactly once and in the correct order.

It is important to note that not all LAN types implement sequence numbering at the frame level. In fact, very few do. In most LANs, the software above the data link layer must determine that something is missing and negotiate retransmission of the information. Just because the feature might exist in a specification doesn't mean it is implemented in practice.

31. How is error correction actually accomplished?

As previously stated, error correction can be performed in one of two ways. The first strategy, error correction through retransmission, is straightforward. If a destination node detects that a frame of data it received is not identical to the frame sent, then it requests the sending node to retransmit the original frame. The second strategy, autonomous error correction, is a little more complex and does not rely on retransmissions. Instead, the destination node, upon detecting a bad data frame, corrects the error(s) itself. Autonomous error correction requires that a transmitted frame contains redundant information to enable the destination node to correct any detected errors without requesting a retransmission. Note that error correction implies error detection.

32. I'm curious. I understand the need for error control, but what causes network errors in the first place?

The reason error control is necessary is because networks are fallible, complex systems, and errors occurring during data transmission are an inherent part of these systems. Network errors are caused by a wide variety of conditions. For example, errors can be caused by interference on the wire, hardware problems, software bugs, protocol-related problems (e.g., incompatible protocols at either end), and buffer overflow. Some errors are intermittent—they come and go. These are the most difficult to detect. Others are "hard" errors that are easily identified. In a properly functioning network, however, errors usually are a function of line synchronization failures, crosstalk, defective hardware, and protocol-related problems (e.g., "collisions" in Ethernet/802.3).

33. I read that the ARQ light on my dialup modem indicates that error control is being implemented. What is ARQ?

The concepts of error detection, acknowledgment, and retransmission are collectively referred to as *automatic repeat request* (ARQ). As you observed, most dialup modems have an ARQ indicator. ARQ is a general term for error-control protocols that feature error detection and automatic retransmission of "bad" frames. Commonly used ARQ standards are based on some of the flow-control protocols discussed earlier. Two in particular are *stop-and-wait ARQ* and *go-back-n ARQ*. Both protocols are patterned after their flow-control namesakes and involve retransmitting frames that either time out or are not positively acknowledged. A third protocol, *selective-reject ARQ*, involves a receiver sending only negative acknowledgments to indicate that an error was detected in a received frame. (This is in contrast to the previous two protocols in which an acknowledgment is sent for every frame received correctly.) Thus, only frames that time out or have a corresponding negative acknowledgment are retransmitted.

34. Can you give a concrete example of error control in action?

Yes we can. We'll illustrate a single-bit error correction strategy that is based on the concept of *parity*. Are you familiar with this concept?

35. Yes, but it's been a while. So why don't you review it for me.

OK. *Parity* refers to the use of an extra bit (called a *parity bit* or a *redundancy bit*) to detect single-bit errors in data transmissions. Parity can be specified as even, odd, or none. Even parity means that there must be an even number of 1-bits in each bit string; odd parity means that there must be an odd number of 1-bits in each bit string; and no parity means that parity is ignored. The extra bit (i.e., the parity bit) is forced to either 0 or 1 to make the total number of bits either even or odd. For example, consider the character *A*. Its ASCII representation is 1000001. Note this has two 1-bits, and two is an even number. If we require even parity then we must append a 0 to this bit string because we need to maintain an even number of 1s. If we require odd parity, then we need to append a 1 because we need an odd number of 1s (see Figure 5.5). When a single-bit error occurs, the receiver interprets the bit string as a different character than the one sent. Parity checking with one additional parity bit can *detect* this type of error. Unfortunately, however, a single parity bit does not provide enough information to *correct* the error.

36. I thought you said that parity can be used for both detection and correction of errors. How can parity be used for error correction?

Parity can also be used for autonomous error correction. To do so, however, requires that additional information be included in a transmitted frame to enable the receiving node to correct any detected errors on its own, thus avoiding the need for retransmission. Redundancy bits (also called *check bits*) provide this information. A data set composed of both user data and redundancy bits is called a *codeword*. Using parity, we can construct codewords to correct single-bit errors only. An example of how this is effected is provided in Appendix B.

37. I just reviewed Appendix B. That's a lot of work just to correct a single-bit error.

You bet it is. Autonomous error correction is expensive to implement. Several extra bits are required to convey redundant information so that the receiving node can locate an error. In the example given in Appendix B, we needed four extra bits just to be able to detect the location of a single-bit error. Once the position of such an error is located, though, correction is easy—simply flip the bit (i.e., complement the bit at that position). Consequently, autonomous error correction is usually implemented in simplex channels, where retransmissions cannot be requested, or in those instances where retransmission is more costly than implementing an autonomous error correction scheme. For most situations, though, using retransmissions is the preferred method of error correction.

¥ Bit string of A :	1 0 0 0 0 0 1
¥ For even parity, parity bit is 0: (even number of 1-bits)	1 0 0 0 0 0 1
¥ For odd parity, parity bit is 1: (odd number of 1-bits)	1 0 0 0 0 0 1

FIGURE 5.5 Examples of parity bits.

38. What about correcting multibit errors?

To correct multibit errors, the price goes up. What we mean by this is we must increase the codeword's size to enable us to correct additional bits, adding a significant amount of overhead to the transmission. If the physical layer is screwing up so badly that we expect many multibit errors, we probably ought to consider reengineering or replacing the equipment.

39. What error control strategy does Ethernet/802.3 use?

Ethernet/IEEE 802.3 uses error detection with retransmission. If you look back at Figure 5.3, you will see that the last field in the frame is the *checksum*, used to detect errors. The checksum technique employed by Ethernet/802.3 is called *CRC-32*.

40. What's a CRC-32 checksum?

We're glad you asked. Are you ready for a little bit of mathematics?

41. OK. Just a little bit. Promise?

Sure. CRC stands for *cyclic redundancy check*, which is an extremely powerful and robust error detection method. To check a series of bits, CRC first constructs a polynomial whose terms' coefficients are the values of each of the bits. Thus, a data set with n bits corresponds to an $n-1$ degree polynomial; the leftmost bit is the coefficient of the x^{n-1} term. For example, the eight-bit data set 10111101 is equal to the seventh-degree polynomial, $1x^7 + 0x^6 + 1x^5 + 1x^4 + 1x^3 + 1x^2 + 0x^1 + 1$, or equivalently, $x^7 + x^5 + x^4 + x^3 + x^2 + 1$. In the next step, the polynomial, is divided by a predetermined *generator polynomial*. (Thus, the data set is the dividend and the generator polynomial is the divisor.) The remainder of this division is the CRC checksum, which is included with the frame. In an Ethernet/IEEE 802.3 frame, for example, the CRC checksum is the last field. The receiving node performs an analogous procedure on a received frame, using the same generator polynomial. If the CRC checksum calculated by the receiving node is equal to what was sent, then the frame is interpreted as correct. If the two CRC checksums do not match, then the sending node is notified of this and the entire frame is retransmitted. Three standard generator polynomials are:

- **CRC-16,** a 16-bit checksum used for various file transfer protocols. Its generator polynomial is $x^{16} + x^{15} + x^2 + 1$.
- **CRC-CCITT,** a 16-bit checksum that serves as an international standard. Its generator polynomial is $x^{16} + x^{12} + x^5 + 1$.
- **CRC-32,** a 32-bit checksum used in most LAN protocols. Its generator polynomial is $x^{32} + x^{26} + x^{23} + x^{22} + x^{16} + x^{12} + x^{11} + x^{10} + x^8 + x^7 + x^5 + x^4 + x^2 + x$. (CRC-32 is used in Ethernet/802.3 and token ring LANs.)

A simple example of CRC is shown in Box 5.2.

42. How efficient is CRC in detecting errors?

The efficiency of CRC is a function of the generator polynomial used. CRC-16 and CRC-CCITT detects 100 percent of all single and double errors, all errors with an odd number of

BOX 5.2: CRC Checksums

The cyclic redundancy code (CRC) is an error-detection scheme that uses an algebraic polynomial to represent a bit stream. The sending node divides a data frame, T_b, which is interpreted as a single bit stream, by a specific divisor, D, called a generator polynomial. The remainder, R, of the division is then included in the frame's checksum field and this frame, T_a, is transmitted as part of the frame. T_b represents the frame *before* a CRC checksum is calculated, and T_a represent the frame *after* the checksum is calculated. Thus, T_a is the frame that is actually transmitted by the sending node.

When the destination node receives the frame, it divides T_a by D. This implies that the destination node does exactly the same division using the same generator polynomial as the sender. If the transmitted frame is error free, then the remainder of the destination node's division is 0 and the destination node assumes the frame is valid. If the remainder is not 0, then an error occurred during transmission and the destination node requests a retransmission of the frame. Following is an illustration of the CRC procedure. For this example, let us assume that the generator polynomial is $x^5 + x^4 + x^2 + x + 1$. This implies that the divisor, D, is 110111. Note that the x^3 term is represented as 0 in our divisor. Let us further assume that the data frame to be transmitted, T_b, is 111000111000.

Step 1
Since the generator polynomial is a fifth-degree polynomial, we first append five 0s to T_b.

111000111000 $\longrightarrow$ 111000111000

Step 2
We next divide the modified bit string from Step 1 by D.

```
         101011101000     = Quotient
110111 | 11100011100000000    = Tb
         110111
           111111
           110111
             100010
             110111
              101010
              110111
               111010
               110111
                 110100
                 110111
                     11000    = Remainder
```

Note that binary polynomial arithmetic uses mod 2 arithmetic, that is, there are no carries or borrows. Addition and subtraction corresponds to XOR. For example:

```
    101101
  – 110011
    011110
```

(continued)

BOX 5.2: CRC Checksums (continued)

Step 4

The destination node receives the transmitted frame from step 3 and performs the same division.

```
                101011101000
110111 | 1110001110001100 0
         110111
         ------
          111111
          110111
          ------
           100010
           110111
           ------
            101010
            110111
            ------
             111010
             110111
             ------
               110111
               110111
               ------
                 00000    = Remainder
```

Since the remainder is 0, the destination node assumes the frame is valid. If the remainder was not 0, the destination node requests a retransmission of the frame from the sender.

bits, all bursts of 16 bits or less, 99.997 percent of 17-bit error bursts, and 99.998 percent of 18-bit and longer error bursts. With CRC-32, however, the chances of bad data being received and not detected are approximately 1 in 4.3 billion (2^{32-1}).

43. Wow! So although errors are inevitable in networks, thanks to CRC-32 there is little chance that they will go undetected, right?

That's right. However, this does not mean that you should be negligent in your responsibility as a network manager or administrator. You should always strive to minimize network errors by replacing *IS Class* media and equipment. OK. Enough of the LLC sublayer. We need to move on to the MAC sublayer.

44. Before we proceed to the MAC sublayer, could you give me some examples of bad Ethernet/802.3 frames?

Sure, but before doing so we suggest you first review Figure 5.3. For instance, it will be helpful to know that the length of an IEEE 802.3 frame ranges from a minimum of 64 bytes to a maximum of 1518 bytes. Go ahead. We'll wait. We have lots of time.

45. OK. I've finished my review. Now give me some examples of bad Ethernet/802.3 frames.

Ethernet/802.3 errors manifest themselves as bad or invalid frames. We will discuss causes of these errors in Chapter 8, but for now, we give a summary of them:

Oversized Frames. Oversized frame have more than 1518 bytes but also have a valid CRC checksum. Oversized frames usually indicate a software problem such as a faulty network driver. The industry slang term for this condition is a "long" frame.

Runt Frames. Runt frames are short frames. They are at least 8 bytes but less than 64 bytes long and have a valid CRC checksum. Runts usually indicate a software problem such as a faulty network driver.

Jabbers. Jabbers are oversized frames that have an invalid CRC checksum. Jabbering is caused when a station has transmitted for too long. This normally causes an invalid CRC. Jabbers usually indicate a hardware problem, typically a faulty transceiver.

Alignment or Frame Errors. These are frames that do not end on a "byte-boundary." A frame error is detected when the total number of bits is not a multiple of eight (i.e., they cannot be grouped into an exact number of eight-bit bytes).

CRC Errors. Frames with CRC errors are of the proper size and alignment but have an invalid CRC checksum. CRC errors are caused by noise, bad connections, and faulty network hardware.

46. Okay. I'm ready for the MAC sublayer! Why is it necessary and what does it do?

LANs employ a broadcast topology, meaning the nodes of a LAN share a single communications channel, and must all contend for the same medium in order to transmit data. This is analogous to a large group of college students contending for the only working telephone in a dormitory lobby—it can be a long wait when things are busy. Because of the potential chaos associated with such contention, LANs must employ protocols that define the *who*, the *how*, the *when*, and the *for how long* of channel allocation. Enter the medium access control sublayer! The MAC sublayer provides the protocols that define the manner in which nodes share the single physical transmission medium. The name says it all: Media Access Control. Two broad categories of access methods are most suitable for LANs—*random access* (sometimes called *stochastic*) and *token passing* (referred to as *deterministic*).

47. Random access. Hmm … Does this have anything to do with RAM?

Only indirectly, in that they both involve random access. Random access protocols define how a node can access a communications channel. These protocols employ the philosophy that a node can transmit whenever it has data to transmit. Random access protocols imply contention; in fact, sometimes we also call them *contention protocols*.

48. What do you mean by contention?

Consider a classroom in which students and teacher are engaged in an open discussion—anyone can begin speaking without needing acknowledgment. In such a setting, anyone who wants to speak must contend with others who also want to speak simultaneously. Thus, "contention" is a phenomenon in which more than one entity competes to do something at the same time.

49. Doesn't this create chaos? Not everyone will be able to hear what one person is saying?

Exactly! In order for meaningful communication to take place, speakers must follow a rule that both ensures they can speak when they have something to say, and resolves the problem when more than one person begins speaking at the same time. For example, one rule might be: "If more than one person begins speaking at the same time, everyone is to stop talking for just a split second, and then one person should take the lead and begin talking." Another rule might be to have a facilitator who calls on individuals to speak. Have you ever seen a Presidential press conference? Reporters are always shouting to get their question asked. Regardless of how many reporters attempt to ask a question, only one person at a time speaks. In some cases it is the reporter who shouts the loudest; in other cases, the President identifies the person who is permitted to speak.

50. Okay. Put this in the context of a LAN for me.

Just as when more than one person attempts to talk at the same time, when two or more nodes try to communicate at approximately the same time, their transmissions "collide." In LAN terminology, we refer to this as a *collision*.

51. How does a node know when a collision has occurred?

The physical characteristics of a specific medium enable it to detect collisions. More specifically, when a collision occurs, a channel's energy level changes and nodes on the network are equipped to detect this condition. During such times, the nodes' signals become garbled.

52. What do nodes do if their signals collide?

The answer is, "It depends." It depends on a node's MAC sublayer protocol. Similar to the way people might observe different protocols when contending to speak, nodes employ various protocols to transmit data in a shared media environment. For example, to minimize the occurrence of collisions, nodes might follow a protocol that requires them to first "listen" for another node's transmission (somewhat incorrectly called a *carrier*, a term borrowed from radio terminology) before they begin transmitting data. We call these types of protocols *carrier sense* protocols. Carrier sense protocols require that before a node begins transmitting it must first listen to the wire to determine whether another node is transmitting data. Carrier sense transmission systems also employ circuitry that requires the system to "listen" to every bit transmission going out and compare that to what is actually heard on the transmission medium. If they match, wonderful. If they don't, something caused the bit to get hammered and this means the transmission is garbled. How it is handled from there depends on the transmission framing method being used.

Other protocols that do not involve collision detection involve nodes being given permission to transmit by possessing a special control frame, called a token. We call such protocols *token passing* protocols.

53. Tell me more.

OK. Let's begin with carrier sense protocols. There are four of them: (1) 1-persistent CSMA (Carrier Sense Multiple Access), (2) nonpersistent CSMA, (3) CSMA with Collision Detection (CSMA/CD), and (4) CSMA with Collision Avoidance (CSMA/CA). We'll describe these one at a time.

1-persistent CSMA When a node has data to transmit, it first senses the channel to determine if another node is transmitting. If the channel is not busy, then the node begins transmitting. If the channel is busy, then the node continuously monitors the channel until it senses an idle channel. Once it detects an idle channel, the node seizes the channel and begins transmitting its data. With this protocol, nodes with data to transmit enter a "sense and seize" mode—they continuously listen for a clear channel, and once detected, begin transmitting data. This protocol is similar to telephones with a multiple redial feature. If it detects a busy signal when the call is first attempted, the telephone repeatedly dials the number until a connection is finally established. The "one" in 1-persistent CSMA represents the probability that a single waiting node will be able to transmit data once it detects an idle channel ($p = 1$). However, collisions can and do occur if more than a single node desires to transmit data at approximately the same time.

Nonpersistent CSMA This protocol is similar to 1-persistent CSMA, except a node does not continuously monitor the channel when it has data to transmit. Instead, if a node detects a busy channel, it waits a random period and rechecks the channel. If the channel is idle, the node acquires the channel and begins transmitting its data. If, however, the channel is still busy, the node waits another random period before it checks the channel again. Both 1-persistent CSMA and non-persistent CSMA protocols eliminate almost all collisions except for those that occur when two nodes begin transmitting data nearly simultaneously. For example, node *A* begins transmitting data on a clear channel. A few microseconds later (which is not enough time for node *B*'s sensing circuit to detect node *A*'s transmission), node *B* erroneously declares the channel clear and begins transmitting its own data. Eventually the two transmissions collide.

1-persistent CSMA involves a significant amount of waiting and unfairness in determining which node gets the medium. It is "selfish" because nodes can grab the channel whenever they feel like it. Nonpersistent CSMA seems better because in its randomness there is fairness. In either case, though, there still remains the nagging problem of what to do about collisions. The next two CSMA protocols incorporate *collision detection* to provide a solution to the collision problem.

CSMA with Collision Detection (CSMA/CD) In this variant of either 1-persistent or nonpersistent CSMA, when a collision occurs the nodes (1) stop transmitting data, (2) send out a jamming signal, which ensures that all other nodes on the network detect the collision, (3) wait a random period of time, and then, (4) if the channel is free, attempt to retransmit their message. We illustrate the CSMA/CD algorithm in Figure 5.6.

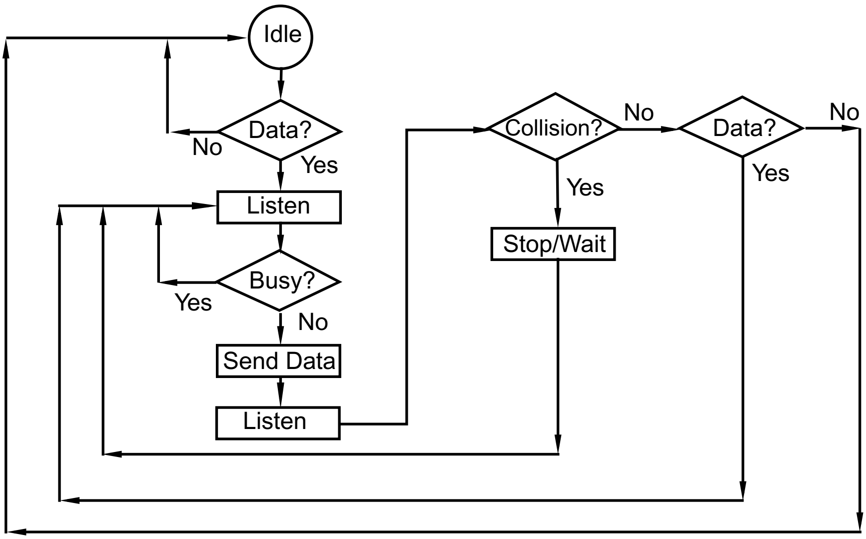


FIGURE 5.6 Flowchart depicting the CSMA/CD protocol.

CSMA with Collision Avoidance (CSMA/CA). This protocol is similar to CSMA/CD except that it implements *collision avoidance* instead of collision detection. As with straight CSMA, hosts that support this protocol first sense the channel to see if it is busy. If the host detects that the channel is not busy then it is free to send data. What if every host connected to a LAN is not always able to sense each other host's transmission? In such instances, collision detection will not work because collision detection is predicated on nodes being able to hear each other's transmissions. This is the case with wireless LANs (WLANs). We cannot always assume that each station connected to a WLAN will hear each other station's transmissions. One direct consequence of this is that a sending node will not be able to detect if a receiving node is busy or idle. To address this issue, we replace CD with CA and use something called *positive acknowledgment*. It works like this:

> The receiving host, upon receiving a transmission, issues an acknowledgment to the sending host. This informs the sending host that a collision did not occur. If the sending host does not receive this acknowledgment, it will assume that the receiving node did not receive the frame and it will retransmit it.

CSMA/CA also defines special frames called *request to send* (RTS) and *clear to send* (CTS), which further help minimize collisions. A sending node issues an RTS frame to the receiving node. If the channel is free, the receiving node then issues a CTS frame to the sending node. CSMA/CA is used in IEEE 802.11 (WLANs) and with Apple Computer's LocalTalk, a low-speed, peer-to-peer network schema that remains very popular for small sites with Apple Macintoshes.

54. CSMA/CD sounds like Ethernet/802.3. Is this what Ethernet/802.3 is based on?

Yes, 1-persistent CSMA/CD is, indeed, the MAC sublayer protocol used in Ethernet and IEEE 802.3 networks. We discuss this further in Chapter 8, but because it is so broadly deployed, we will answer any other questions you might have about CSMA/CD at this time.

55. Thanks. I'll take you up on that. One question that arose when you described CSMA is just how long is "a random period?"

It can be a long time, a short time, a moderate amount of time—it's random, but it is also extremely critical. If the wait time is too short, collisions will occur repeatedly. If the wait time is too long, the medium could be in a constant idle state. Propagation delay also has an important effect on the performance of these protocols. Consider the example given for nonpersistent CSMA: Node *A* senses an idle channel and begins transmitting. Shortly after node *A* has begun transmitting, node *B* is ready to transmit data and senses the channel. If node *A*'s signal has not yet reached node *B*, then node *B* will sense an idle channel and begin transmitting. This will ultimately result in a collision. Consequently, the longer the propagation delay, the worse the performance of this protocol. Even if the propagation delay is zero, it is still possible to have collisions. For example, assume both nodes *A* and *B* are ready to transmit data, but they each sense a busy channel (a third node is transmitting at the time). If the protocol is 1-persistent CSMA, then both nodes would begin transmitting at the same time resulting in a collision. The specific amount of wait time is a function of the protocol. For quick explanation purposes, though, Ethernet/802.3's "random" amount of time is between 1 and 512 *slot-times* on the cable. We will expand on this concept when we discuss Ethernet/IEEE 802.3 later (so please don't ask what a slot-time is at this point).

56. What can you tell me about token passing protocols?

Well, unlike random access protocols, token passing protocols rely on granting nodes permission to transmit. Permission is provided in the form of a special control frame called a token. The underlying principle of token passing protocols is simple: the node that possesses the token may access the medium. Since possession of the token controls access to the medium, the possibility of contention is eliminated. The absence of contention also implies an absence of collisions. Thus, token-passing schemes are both contention-free and collision-free protocols.

The IEEE has defined two LAN protocols based on token passing. They are IEEE 802.4 (token bus) and IEEE 802.5 (token ring). While not defined by an IEEE standard, Fiber Distributed Data Interface (FDDI) also uses a token passing technology. Token ring and token bus are discussed in Chapter 9; FDDI is discussed in Chapter 10.

57. Please compare and contrast random protocols and token passing protocols.

Certainly. Random access protocols are stochastic in nature. They are predicated on the principle that the probability two computers will transmit simultaneously (i.e., within a few microseconds of each other) is near zero. Given such a low probability, these protocols permit simultaneous transmissions. They are engineered to enable nodes to both detect and resolve collisions resulting from such transmissions. Token passing protocols, in contrast, are deterministic in nature. They are predicated on the principle that no two nodes should be permitted to transmit at the same time. As such, these protocols effectively eliminate contention and are viewed as collision-free or collision-avoidance protocols. A direct consequence of determinism is that it is possible to accurately determine the worst-case performance of a LAN.

58. Which of these two protocol types is more advantageous?

Well, that depends. Each type of protocol has its own advantages and disadvantages. For example, contention protocols yield high performance for lightly loaded LANs, but when LAN traffic increases, protocol performance decreases. Performance of collision-free protocols is very predictable when a LAN is heavily loaded. However, this same feature also invokes a fixed delay even when a LAN is lightly loaded. To better understand these advantages and disadvantages, let us consider two example scenarios involving automobiles and street intersections.

In Scenario A, we have an intersection that has no traffic control device (i.e., no stop or yield signs, no flashing yellow or red lights, no traffic signal, etc.). In Scenario B, we have the same intersection, now controlled by a standard traffic signal light operated by a timer. The protocol for passing through the intersection in Scenario A is as follows: As you approach the intersection, beep your horn. If you do not hear a horn in return, you may proceed without slowing down. If you hear another horn in response to yours, slow down and proceed with caution. The protocol for Scenario B is simple: Obey the traffic signal. If it's green, you may proceed through the intersection; if it's red, you must stop and wait for a green light. Clearly, in Scenario A, vehicles can pass through the intersection quite easily when traffic is light or nonexistent. On the other hand, once traffic increases, delays become more frequent and longer lasting, creating greater likelihood of collisions. Scenario A is analogous to the schemes used for contention protocols. In Scenario B, traffic is controlled via a traffic signal. Consequently, during heavy traffic loads you can predict approximately when you will be able to negotiate the intersection by counting the number of vehicles passing through while the signal is green. At the same time, however, what happens if you are stuck at this red light in very light traffic? You must still wait until the light turns green, whether the intersection is clear or not. This is a very inefficient use of time. Scenario B is analogous to collision-free protocols. Table 5.2 contains a summary of these advantages and disadvantages. Box 5.3 also provides a summary of statistical and deterministic LANs.

TABLE 5.2 Advantages and Disadvantages of Contention and Collision-free Protocols

Protocol	Advantage	Disadvantage
Contention	Faster access on lightly loaded systems	Poor performance at heavy loads
Collision-free	Very predictable at high loads	Fixed delay required at low loads

BOX 5.3: Statistical (Stochastic) versus Deterministic LANs

Statistical LANs

- can only estimate when the network will be available for use by a specific node
- are load-metric-based and not node-metric-based
- are difficult to predict performance
- produce burst-oriented traffic
- match more closely how network applications use the network

Deterministic LANs

- can determine exactly when a node will get to access the network
- are node-metric based (i.e., number of nodes determines round-trip time)
- can determine exact round-trip time
- can determine overall performance in advance of using the network
- can determine the impact of additional nodes and load prior to network connection

59. Okay. In the chapter opener, you mentioned something about data prioritization and quality of service. Please explain what you mean by these terms?

Data prioritization involves assigning a priority level to data frames, and *quality of service* (QoS) involves establishing certain parameters for a specific transmission. Both data prioritization and QoS are required to deliver real-time voice and video traffic. Note that tagging data with a priority level is also called *class of service*, or CoS.

60. How is this accomplished?

Let's examine these two concepts from the perspective of Ethernet/802.3, which is discussed formally in Chapter 8. Thus, if you find some of the following information unclear, we suggest you jump ahead and review the pertinent sections of Chapter 8.

As we noted in previous discussions, Ethernet/802.3 is a shared-media standard based on contention. Thus, congestion is an inherent feature of Ethernet/802.3 networks, and it is normal for frames to be dropped. Two questions emerge from this: (a) How much congestion is acceptable? and (b) Does it matter which frames are dropped? If congestion levels are too excessive and unacceptable, or if we do not want to lose any frames, then increasing bandwidth will help resolve these problems; data prioritization will not. An Ethernet/802.3 network will continue to experience congestion and lose frames regardless of whether a data prioritization scheme is implemented. The function data prioritization adds is control over which frames get lost. For example, if e-mail is the highest priority application on our network, then e-mail frames will be tagged as such and their transmission will take precedence over all other traffic. If Web-based applications are the highest priority, then they can be tagged as such using a data prioritization scheme. Tagging data frames with a specific priority level is important for real-time transmissions because during periods of congestion we do not want voice or video data sets to be dropped by switches. A high-priority assignment to these data sets ensures their delivery.

61. Okay. I understand the concept. However, just because you've tagged high priority data as such, what assurances do we have that these data will indeed get through if the network is congested?

Good question. That's where QoS enters the picture. As you observed, data prioritization is only part of the equation. The delivery of time-sensitive data also requires that sufficient bandwidth be available and that transmission delays (i.e., latency) be predictable and guaranteed. This is the essence of QoS. A properly implemented QoS strategy ensures that all transmissions, regardless of their type (data, voice, or video), receive the necessary bandwidth, delivery times, and appropriate priority based on the importance of data delivery.

62. How are data prioritization and QoS implemented technically?

To address CoS (i.e., data priority) and QoS, the IEEE introduced two data link layer protocols: IEEE 802.1p and IEEE 802.1q. IEEE 802.1p is an extension to IEEE 802.1d, which specifies how MAC-level bridges are to interoperate regardless of the IEEE LAN standard being used. (Bridges are discussed in Chapter 6.) What 802.1p adds to the 802.1d standard is a specification for implementing prioritization in 802.1d-compliant bridges. IEEE 802.1p defines a three-bit priority scheme that provides eight different levels of priority. This new standard can be used on LANs that have an inherent prioritization scheme. This includes IEEE 802.4 (token bus, see Chapter 9), IEEE 802.5 (token ring, see Chapter 9), IEEE 802.6 (DQDB, see Chapter 13), and IEEE 802.12 (100VG-AnyLAN, see Chapter 8). Data prioritization can also be incorporated into an FDDI network (Chapter 10).

63. What about Ethernet?

IEEE 802.1p cannot be used with current Ethernet/802.3 networks because Ethernet was never designed with data prioritization. To resolve this, the IEEE developed 802.1q, which provides a data prioritization scheme to Ethernet/802.3. There is one small problem, though—to do so, the 802.3 frame had to be altered. Specifically, a four-byte 802.1q header is inserted between the source address and length fields (Figure 5.7).

64. Doesn't this make it incompatible with the standard Ethernet/802.3 we've come to grow and love?

Yes. An 802.3 frame that includes an 802.1q header is incompatible with a standard 802.3 frame. This also makes an 802.1q-compliant device incompatible with an 802.3 device. Instead of seeing the length field, an 802.3 device will be greeted with 802.1q header information. One workaround to this device incompatibility issue is for vendors to manufacture 802.1q devices with a switch that disables 802.1q. Although disabling 802.1q also disables data prioritization, it enables 802.1q-compliant products to be purchased and installed in anticipation of migrating to a data prioritization scheme.

65. But what about the frame-length difference.

The frame-length difference is not as easily addressed. At issue is the frame size. If you reference Figure 5.3 and ignore the preamble, note that the maximum size of a standard 802.3 frame is 1518 bytes. The maximum size of an 802.3 frame that includes the 802.1q

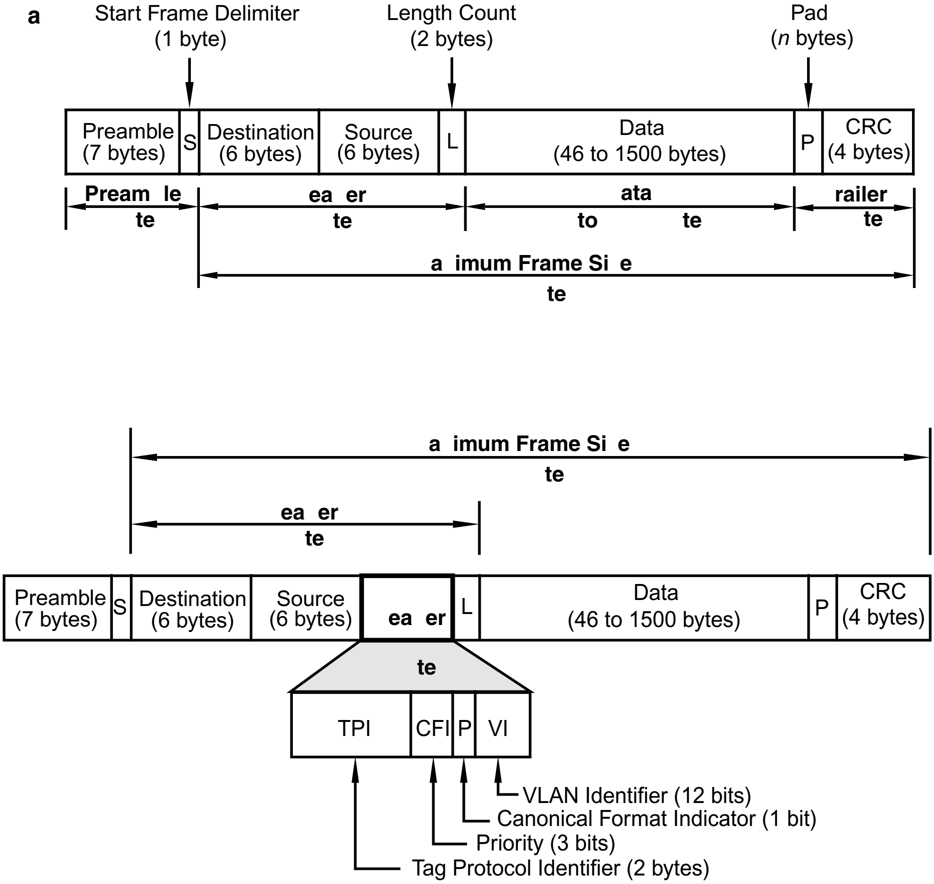


FIGURE 5.7 An IEEE 802.3 frame (a) with an IEEE 802.1q header (b) adds 4 bytes to the 802.3 frame. This changes the size of the frame's header from 14 bytes to 18 bytes, which increases the maximum frame size from 1518 bytes to 1522 bytes. The 3-bit priority (P) field of the 802.1q header provides for seven different priority levels.

header is 1522 bytes. Two possible solutions abound: (a) we can increase the size of 802.3 frames by four bytes or (b) we can reduce the length of the payload (i.e., the user data field) of 802.3 frames by four bytes so they can accommodate 802.1q header information.

66. So which "road" did IEEE ultimately decide to take?

Well, instead of making a decision, the IEEE put the onus on the vendors. That is, the IEEE 802.1q specification allows for either solution to be incorporated. This means that it is up to the vendors to decide which method they want to implement. It also raises the question of interoperability.

67. So now I have something else to worry about. Oy!

Before stressing yourself out over this, remember that as is the case with any networking decisions, a thorough network analysis should be undertaken before implementing CoS and QoS. For example, if your network is transmitting only data (no voice or video) and you do not care about what frames are being discarded, then you probably do not need to be concerned with data prioritization. However, if your network is currently supporting, or if you are planning to implement on the network bandwidth intensive applications such as videoconferencing, distance education applications, or those electronic whiteboards that enable people to collaborate on-line, then you might want to consider a data prioritization scheme. If you are unsure whether you need a prioritization scheme, then you can always play it safe by ensuring that the Ethernet controller cards and switches you purchase support IEEE 802.1p and 802.1q. Try to get them from the same vendor to reduce your interoperability woes. You should also confirm that all layer-3 devices (e.g., routers, switching routers) support multiple levels of queuing so that, if data prioritization is implemented, queues will not get overloaded.

68. This might seem like a dumb question, but why not just provision more bandwidth to accommodate any high-bandwidth, low-latency applications instead of worrying about implementing a QoS strategy?

Actually, this is an excellent question. Some people believe that increasing bandwidth is one way to achieve QoS. Others, however, regard this mindset as 20th-century thinking. For some companies that still have lots of cash left over from their IPOs, building networks with a bandwidth capacity that significantly exceeds demand might be acceptable: Having a sufficiently large bandwidth pool will indeed eliminate network congestion, which is the heart of most network degradation. Furthermore, this might work for awhile when usage is low. What happens, though, when more bandwidth intensive applications appear? For most organizations, throwing bandwidth at the quality of service problem is neither economically prudent nor the correct way to address QoS issues. Consider for a moment what increasing capacity involves. You will have to continuously replace network hardware devices, upgrade software, and emplace new media. (On second thought, isn't this what we do now?) Seriously, though, this can be extremely expensive. Until we see the day of free and infinite bandwidth, then throwing bandwidth at the quality of service problem is not the way to address QoS issues.

69. Okay, you've made your point. What about QoS for WANs such as the Internet? I would imagine that this is something that many e-businesses need to address.

You're right. This is a very real concern because e-businesses want to ensure that their network resources are accessible to customers in a timely manner. Given the wide diversity of network applications that permeate the Internet, coupled with the manner in which TCP responds to packet loss, several approaches to QoS have emerged. One protocol that was designed for QoS is the *resource reservation protocol* (RSVP), which is an IETF standard (RFC 2205 and RFC 2750). RSVP allows network applications to reserve bandwidth for their data flows. It is receiver-oriented, which implies that the receiving host initiates and maintains the reservation for a particular data flow. Routers also use RSVP to forward bandwidth reservation requests.

Another protocol is the *multiprotocol label switching* (MPLS), which is currently a consortium standard, but may very well become a de facto or de jure standard in the near future. An MPLS implementation is similar to that of ATM (Chapter 14) or frame relay's virtual circuits (Chapter 12) in that it establishes fixed bandwidth channels for an aggregate data flow. MPLS is used only by routers, and as given by its namesake, can be used with different network protocols including IP, ATM, and IPX. We will discuss MPLS in more detail in Chapter 7 in the context of virtual private networks (VPNs).

END-OF-CHAPTER COMMENTARY

On this note, we conclude our discussion of the data link layer. Many of the concepts we discussed here will be expanded in subsequent chapters. For example, our examination of Ethernet/802.3 LANs in Chapter 8 includes a discussion of the layer-2 differences among conventional Ethernet (10 Mbps), Fast Ethernet (100 Mbps), 100VG-AnyLAN (another 100-Mbps "Ethernet"), Gigabit Ethernet (1000 Mbps), and 10 Gigabit Ethernet. We also examine 802.3 performance issues and address the concept of collision management, both of which are part of the data link layer. Similarly, our discussions in Chapters 9 and 10 on token ring and FDDI, respectively, include matters related to the data link layer. In Chapter 12, we examine frame relay, which is a synchronous data link layer protocol based on the concept of packet-switching. Frame relay is designed with a variable-length frame size. Chapter 13 contains information about another interesting data link layer protocol: IEEE 802.6, distributed queue dual bus (DQDB). DQDB is used as the data link layer of a packet-switched, broadband MAN called switched multimegabit data service (SMDS). Finally, the concepts of data prioritization and quality of service are revisited in the chapters on Ethernet, token ring, SMDS, and ATM. You are also encouraged to review the material in Chapter 3 on TCP, specifically, the information about TCP's header's window field as well as the TCP enhancements summarized in Box 3.1. This material should be more meaningful to you now that you have an understanding of the sliding window protocol concept.

Chapter 6

Network Hardware Components (Layers 1 and 2)

In Chapter 4, we focused mostly on network media and the physical layer (layer 1 of the OSI model) but did not discuss any specific layer 1 components. Similarly, in Chapter 5, we examined the data link layer (layer 2 of the OSI model) but did not provide any information about layer 2 devices. In this chapter we make up for these intentional omissions by presenting various layer 1 and layer 2 devices, including connectors, transceivers and media converters, repeaters, network interface cards and PC cards, bridges, and switches. An outline of the devices we discuss follows:

- Connectors (Questions 1–6)
- Transceivers (Questions 7–12)
- Repeaters (Questions 13–19)
- Media Converters (Question 20)
- Network Interface and PC Cards (Questions 21–38)
- Bridges (Questions 34–45)
- Switches (Questions 46–57)

1. What are connectors?

Connectors attach components together. Several types of connectors are available, serving various purposes. For example, connectors are used to: (a) connect network interface cards, such as an Ethernet card, to a cable; (b) connect cable segments (e.g., thin coax to thin coax); and (c) terminate a segment. In this last category, connectors actually connect the cable to a terminating resistor or an array of resistors and are consequently known as *terminators*. The type of connector used is usually a function of cable type. For example, *eight-pin modular connectors* are used with UTP cable (Figure 6.1). Connectors are also classified by their gender.

2. Gender? What does gender have to do with connectors? Do connectors have sex?

Yes, connectors "mate." You can use your imagination about how the gender is derived, but a "plug" is usually "male" and a "jack" is usually "female." This universally under-

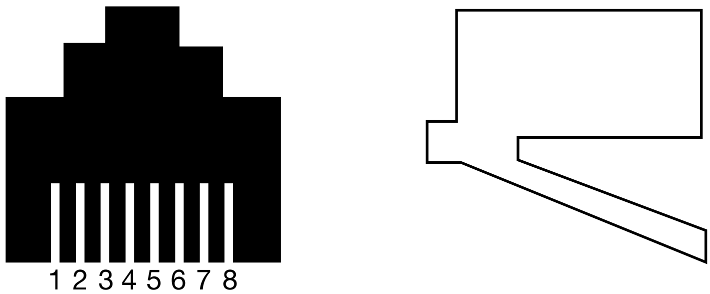


FIGURE 6.1 Top (a) and side (b) views of an RJ-45 connector. Formally known as eight-pin modular connectors, RJ-45s resemble standard telephone jacks and are used with UTP cable.

stood terminology enables a clear specification of exactly which of a mating pair of connectors should be used in the particular application.

3. Referencing Figure 6.1, what's the difference between an eight-pin modular connector and an RJ-45 connector, which looks a like modular telephone "jack"?

Eight-pin modular connectors do indeed resemble the standard modular telephone connectors used in the United States. As for any differences, there is none. Eight-pin modular connectors are commonly called *RJ-45 connectors*. In the strictest sense, the RJ designation refers to a specific series of connectors defined in the Universal Service Order Code (USOC) definitions of telephone circuits. "RJ" is telephone lingo for "registered jack." For example, RJ-11 refers to a four-wire connection used for standard home telephone lines in the United States. Hence, the correct term for UTP connectors used in LANs is eight-pin modular and not RJ-45. In some networking circles, though, the RJ designator has become a generic designation and implies any modular connector.

4. What connectors are used for coax cable?

Connectors used with thin coax are known as *BNC connectors*. There are several interpretations of BNC, including *Bayonet Neill-Concelman* (named after its developers), *Bayonet Nut Connector*, and *Barrel Nut Connector*. We have also seen *British National Connector*. Several different types of BNC connectors are available and used for specific purposes. For example: *barrel connectors* are cylindrical and connect two segments of cable; *T connectors* (Figure 6.2) are shaped like the letter "T" and connect a device to a cable—the horizontal part of the T connects two segments of cable (like a barrel connector) and the vertical part of the T connects the device; *end connectors* are attached to the ends of a cable segment and used to mate the cable to either a barrel or T connector; and BNC *terminators* are attached to each end of a thin coaxial trunk cable to prevent signal reflections, which can interfere with other signals. Terminators provide electrical resistance at the end of a cable and "absorb" signals to keep them from bouncing back and being heard again by the devices connected to the cable.

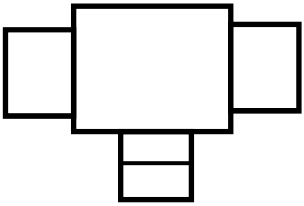


FIGURE 6.2 Example of a BNC T connector.

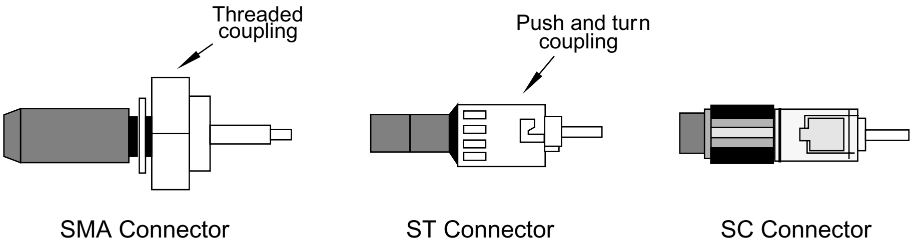


FIGURE 6.3 Examples of different types of fiber-optic cable connectors

For the thick coaxial cable used in early Ethernet applications a *Type N* connector was employed to connect cable segments to each other (via barrel connectors) and to end terminators. N connectors are large, threaded connectors that accommodate the half-inch (12.5 mm) size of the coaxial cable. In "ThickWire" Ethernet networks, transceivers were attached to the medium either by "intrusive tap," which required the cable to be cut and female N connectors installed on both the cut ends, in turn connecting to the male N connectors on the transceiver; or by "vampire tap," which penetrated the cable. These methods are discussed in Chapter 8.

5. How about fiber-optic cable?

Several types of fiber-optic connectors exist. Before the issuance of the EIA/TIA-568A standard, the most popular were ST connectors, which are similar to BNC connectors in that you push and turn the connectors to mate them to a device. SC connectors are those approved by the EIA/TIA-568A standard, by virtue of which they are destined to eclipse the popularity of the ST type. SC connectors, which are also called 568SC, are available in single and dual varieties, the latter designed to aid differentiation between transmit and receive fibers. SMA connectors use a threaded coupling mechanism—you attach them by screwing one end onto another. SMA connectors are designed to meet stringent military specifications. Another type of fiber-optic connector is the MIC (medium interface connector) connector used in FDDI networks (Figure 6.3).

6. What about a printer cable connector or a modem cable connector? Do they also operate at the physical layer?

You are referring to *DB* (data bus) and *DIN* (Deutsche Industrie Norm, a German industrial standard) connectors. DB connectors serve as an interface between a computer and a peripheral device such as a printer or external modem. Several types of DB connectors exist and are distinguished by the number of "pins" they contain. Common types include DB-9 (a 9-pin serial or video interface), DB-15 (a 15-pin video interface), DB-25 (a 25-pin serial interface—RS-232—or parallel printer interface), and DB-37 (a 37-pin serial interface based on RS-422). (Illustrations of DB-9 and DB-25 connectors are shown in Figure 4.1.) DIN connectors are similar, but they are circular instead of rectangular. DIN connectors are typically used to connect a keyboard to a computer.

7. OK. Enough about connectors. Let's move on to transceivers. What are they?

Transceivers are devices used in Ethernet/802.3 networks to connect nodes to the physical medium. They serve as both the physical connection and the electrical interface between a node and the physical medium, enabling the node to communicate with the medium.

8. Given the name, transceivers presumably transmit and receive signals. Right?

Precisely. Transceivers transmit and receive signals simultaneously. When a node sends data, the transmitting circuitry of the transceiver places the data bits on the medium. Simultaneously, the transceiver's receiving circuitry listens to the transmission. If what is heard is the same as what was sent, then everything is fine. If not, the transceiver presumes that an error has occurred and notifies the node of this condition. (This error is called a "collision," which is discussed in Chapter 8.) In a nutshell, a transceiver essentially does three functions: it sends data, it receives data, and it notifies its host node if an error condition has occurred during a transmission. (*Note:* In Ethernet V2.0, a transceiver performs a fourth function: it asserts signal quality error (SQE). This is discussed in Chapter 8.)

9. Exactly where is a transceiver located in a network? Is it a separate device? Is it built into the computer? Where is it?

Today, transceivers are usually integrated into *network interface cards*, which are discussed later. This integration allows a node to be connected directly to the medium via a cable without the need for an external transceiver. A network interface card's on-board transceivers support UTP, ThinWire, and fiber-optic cable connections. Many network interface cards also can be purchased with a "universal" connector, called a 15-pin *attachment unit interface* (AUI), which allows a device to be connected to UTP, thick or thin coax, or fiber-optic cable via an external transceiver (Figure 6.4). Ethernet/802.3 network interface cards (Figure 6.5) can be purchased with any three connector types.

10. Could you expand on the notion of AUI, please?

We shall, although the use of AUI connectors and cables is rapidly fading into Ethernet history. As we stated above, AUI stands for attachment unit interface. It is a 15-pin D-shell

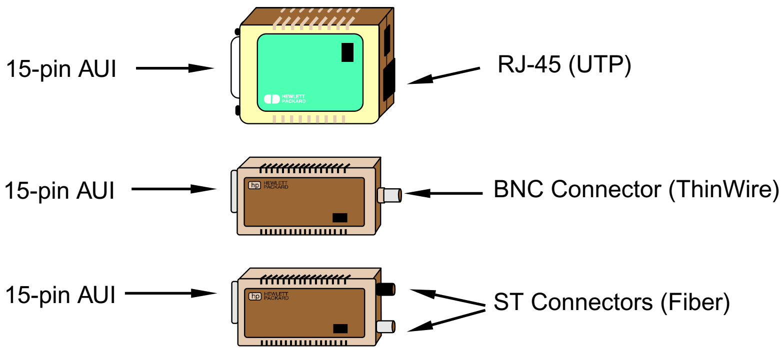


FIGURE 6.4 Three types of Ethernet/802.3 transceivers that serve as media converters. When connected to a network interface card's 15-pin AUI connector, the card is able to support twisted-pair, ThinWire, or fiber-optic cable.

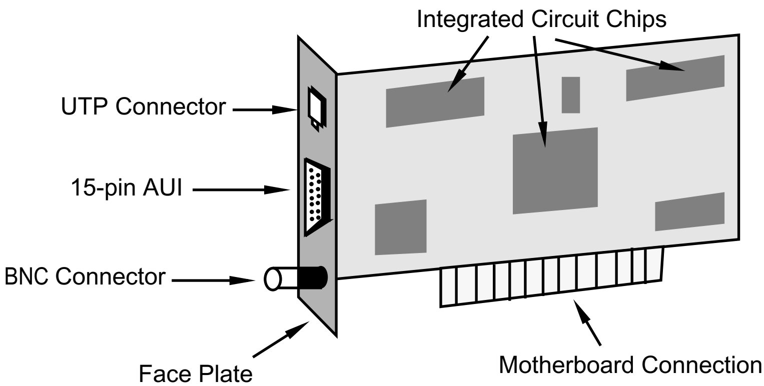


FIGURE 6.5 Sample Ethernet network interface card. This NIC can support any one of three media types: UTP, transceiver cable via an AUI, and ThinWire. NICs are installed into a node by inserting them into an expansion slot on the motherboard.

type connector (Figure 6.6). On the computer side, the AUI is female; on the transceiver side it is male. Many older Ethernet/802.3 devices such as bridges or switches (discussed below) could be purchased with AUI connector ports so that users could connect any of the three media types to any port using one of the external transceivers shown in Figure 6.4. As we have stated, however, the use of AUI is becoming less and less prevalent in the late 1990s.

ThickWire Ethernet networks also use external transceivers with AUI connectors to connect a node to the cable. In this setting, the transceiver was a bulky stand-alone device that required special transceiver cable to connect the node to the transceiver. The trans-

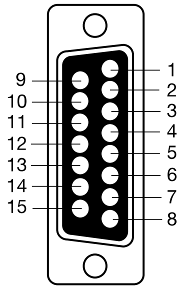


FIGURE 6.6 Standard 15-pin attachment unit interface (AUI).

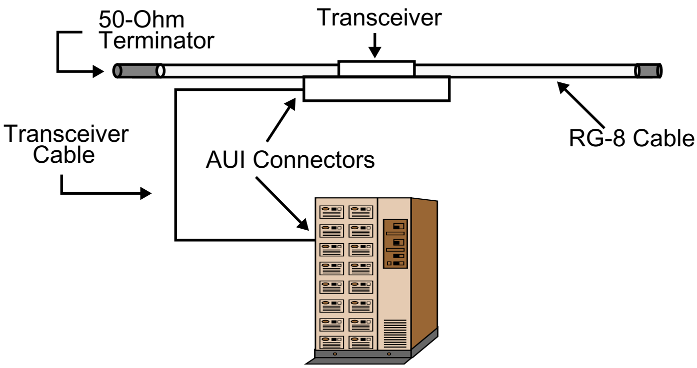


FIGURE 6.7 An external transceiver used in ThickWire Ethernet LANs. Special transceiver cable is used to interconnect the node to the transceiver. The connector on the computer side is a 15-pin female AUI, and the connector on the transceiver is a 15-pin AUI male.

ceiver cable carried signals between the network cable and the node (Figure 6.7). There also used to be multiport transceiver units such as the Digital Equipment Corporation DELNI, which contained eight AUI ports providing a total of eight independent transceiver cable connections. This technology is archaic, having been replaced by miniature external transceivers such as shown in Figure 6.2. There is no longer a need for transceiver cable since network cards provide a direct interface for different network media.

11. Are AUI connectors the ones that have those weird slide-lock things on them that never seem to work?

Yes, and it sounds like you have used them before. We were not able to get the slide lock to work correctly all of the time either. AUI connectors are also available with screw-in retainers.

12. OK. I now understand AUI, but what do AAUI and MAU mean?

AAUI stands for Apple Attachment Unit Interface, which was Apple's proprietary AUI. Older model Macintosh computers like the Centris 650 and Power Macintosh 7100 have built-in AAUI ports, as evidenced by yet another variety of subminiature 15-pin connector on their rear panels. To connect these machines to an Ethernet/802.3 LAN, you must first purchase an AAUI-to-AUI adapter to change the proprietary interface to AUI, BNC, or RJ-45. Why Apple chose to do this is anyone's guess. Later model Macintoshes have built-in standard connectors—mainly RJ45. As for *MAU*, this stands for Media Attachment Unit; it is another term for a transceiver.

13. Enough about transceivers. Let's talk about repeaters.

OK. *Repeaters*, like transceivers, provide both physical and electrical connections. Their function is to regenerate and propagate a signal.

14. Give me a little more detail. Where are they used? Why are they used?

Repeaters, which are also called *concentrators*, are used in Ethernet/802.3 LANs to extend the length of the LAN. You see, depending on the type of medium, the length of an Ethernet/802.3 LAN segment has specific length restrictions. For example, a 10 Mbps Ethernet/802.3 LAN that uses UTP cable (10BASE-T), has a maximum length restriction of 100 meters, and a ThinWire (coax) Ethernet/802.3 segment (10BASE2) cannot exceed 185 meters. The reason for these restrictions is signal quality. As a segment exceeds its maximum length, the signal quality begins to deteriorate. (Recall the concept of attenuation from Chapter 4.) In many instances, these length restrictions are not always practical, so network managers have to extend their LANs by interconnecting individual segments. A repeater makes this possible. It receives signals from one cable segment, regenerates, retimes, and amplifies them, and then transmits these "revitalized" signals to another cable segment. We discuss the application of repeaters in Ethernet/802.3 LANs in more detail in Chapter 8.

15. Do repeaters introduce delay in a network?

Yes. Repeaters are a source of *propagation delay* in a network. Propagation delay is the amount of time a signal spends getting from one point in a circuit to another. It is affected by the speed and efficiency of the components between the two points.

16. Are repeaters also called hubs?

It depends on who you talk to. Some people think of a *hub* as a repeater, whereas others view a hub generically as any device that connects two or more network segments. A hub can also be a device that supports several different media (see Figure 6.8).

17. What about a stackable hub? What's that?

Stackable repeater hubs consist of individual repeater units "stacked" one on top of another. Instead of a common shared backplane that is part of a chassis-based repeater, stackable hubs use a "pseudo-backplane" based on a common connector interface. An external cable interconnects the individual hubs in a daisy-chained manner. Once intercon-

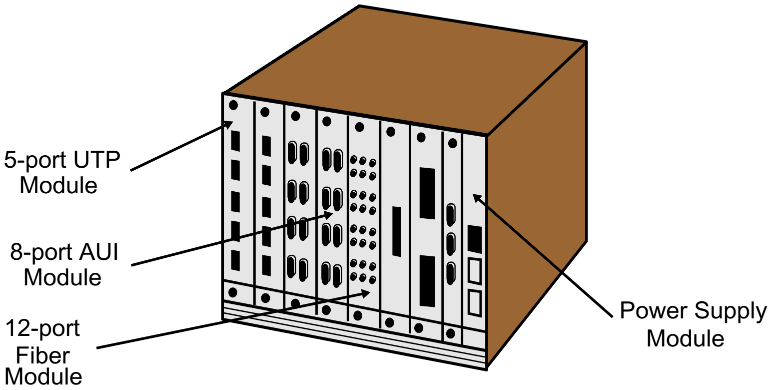
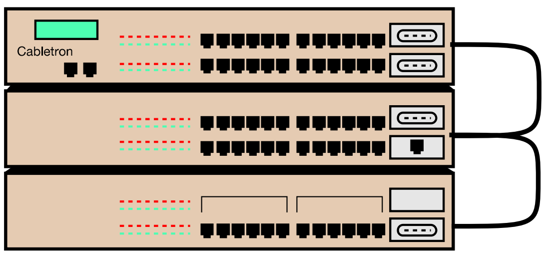


FIGURE 6.8 This multislot chassis-based repeater hub (also called a concentrator) can accommodate several different media types, including UTP, coax, and fiber. Since each interface module shares the same backplane as the repeater module, the Ethernet ports on each module use the same repeater. Thus, repeater hubs are capable of supporting many Ethernet connections using only a single repeater. For example, if one of the modules is a 12-port ThinWire board, then this one board can support 12 separate 185-meter ThinWire segments. Since 30 devices can be connected to one ThinWire segment, this board can support $12 \times 30 = 360$ nodes.



IGURE 6.9 Stackable hubs are daisy-chained together using an external cable, which enables them o function as a single hub.

nected, the entire chain of hubs becomes a single logical unit that functions as a single repeater. Stackable hubs are less expensive than chassis-based devices. An illustration of a stackable hub is shown in Figure 6.9.

18. So what you are saying then is that when I hear the term "Ethernet hub" I shouldn't assume it is a repeater.

Correct. You need qualification. At one point, several years ago, an Ethernet hub was a repeater. Today, however, *switches* have replaced repeaters in most Ethernet/802.3 LANs

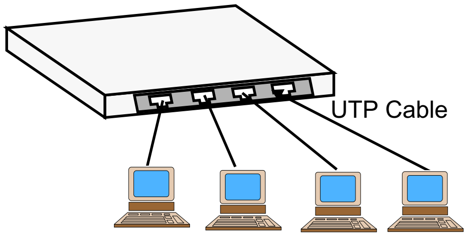


FIGURE 6.10 A four-port "hub." Without any type of qualification, we do not know if this is a repeater hub or a switch hub.

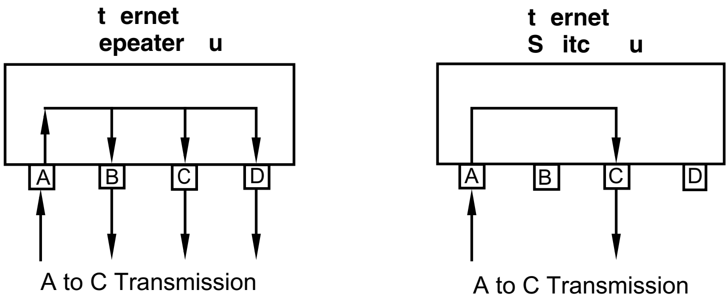


FIGURE 6.11 In a repeater hub, if data on port A are destined to port C, all ports receive the data. In a switch hub, if data on port A are destined to port C, only port C receives the data.

and the term "hub" refers to a switch instead of a repeater. Physically, repeater and switch hubs look the same. For example, consider the illustration shown in Figure 6.10. Without any type of qualification, this 4-port device could be a repeater hub or it could be a switch. We don't know. Functionally, repeater and switch hubs are not the same. A repeater hub takes an incoming signal, amplifies it, and then repeats (i.e., broadcasts) it to all of the hub's ports regardless of the destination port. A switch, on the other hand, only transmits data between the hub's sending and receiving ports (Figure 6.11). We will talk more about switches later in this chapter.

19. Is there anything else I need to know about repeaters?

There are two more pieces of information we should mention. Although repeaters are layer 1 devices, several vendors incorporate some intelligence into them. For example, many Ethernet/802.3 repeater hubs are capable of detecting Ethernet "collisions" (discussed in Chapter 8), and temporarily shutting down any segment that exhibits an excessive number of collisions. You should also know that in an Ethernet/802.3 LAN, Ethernet rules restrict the number of repeaters allowed. We discuss this in more detail in Chapter 8. For now, though, just realize that repeaters are layer 1 devices that regenerate signals.

20. Are there any other layer 1 devices I should know about?

We'll mention just one more: *media converters*, which enable different network media to be connected to one another. For example, using a media converter you can connect a coaxial cable to UTP cable; coaxial cable to fiber-optic cable; a 100 Mbps Ethernet UTP segment to 100 Mbps Ethernet fiber-optic segment; and half or full-duplex UTP segments to 10 Mbps Ethernet fiber-optic segments. (See Chapter 8 for a discussion of 100 Mbps Ethernet and full-duplex Ethernet.) Some people like to think of transceivers as media converters because transceivers support various types of media via a standard AUI connection. Nevertheless, media converters operate at the physical layer and provide a simple mechanism for extending the distance between two devices by mixing copper and fiber cable. (*Note:* Media converters are non-standard devices and are not covered under the IEEE 802.3 standard.)

21. What devices operate at the data link layer?

Several network devices operate at the data link layer (layer 2 of the OSI model), including *network interface cards*, *PC cards*, *bridges*, and *switches*.

22. I am familiar with network interface cards. These are Ethernet cards, right?

Well, yes and no. An *Ethernet card* is a network interface card used in Ethernet/802.3 networks. Not all network interface cards are Ethernet cards, though.

23. This is confusing.

A network interface card is known by many names. Some of the more common ones are *LAN adapter, network adapter, network card,* and *network board.* Generally, we prefer to call them *NIC* (pronounced "nick"). A NIC can support different types of networks and media. For example, an Ethernet/802.3 NIC is designed specifically for Ethernet/802.3 networks and can be purchased with connectors supporting UTP, BNC, AUI, and fiber (see Figure 6.5). An Ethernet NIC is often called *Ethernet card* or *Ethernet adapter.* Similarly, a token ring NIC is designed specifically for token ring networks, can support various media, and is called a *token ring card* or *token ring adapter.*

24. I see. It's a semantic thing. You have the generic name—network adapter—and the specific name—Ethernet card. But don't all NICs do the same thing?

We must equivocate yet again. In a generic sense, all NICs are functionally equivalent As layer 2 devices they perform typical layer 2 functions, including organizing data into frames, transferring frames between the ends of a communication channel, and managing the link by providing error control, initialization, control termination, and flow control (see Chapters 2 and 4). However, it is the network's architecture (Chapter 2) that determines the manner in which these functions are implemented.

25. Can you give me an example?

Sure. Consider the function of framing. The format of Ethernet frames is not the same as that of token ring frames (see Chapters 8 and 9). Hence, an Ethernet NIC does not frame data in exactly the same manner as a token ring NIC. A token ring NIC is also responsible for token-passing and includes chips for monitoring and reporting network errors.

26. Earlier you said that transceivers are usually built directly into a network interface card. Doesn't this make a NIC a layer 1 device also?

Good question. NICs do indeed have layer 1 components as part of their construction and hence perform layer 1 activities in addition to layer 2 activities. For example, on a sending node the NIC performs framing (layer 2) and converts bit values into electrical signals using an appropriate coding scheme (layer 1). At the receiving node, the NIC monitors the medium for transmissions, captures data from the medium if the frame's destination address matches the NIC's address (or if it is a broadcast or multicast), and then passes the data to the node for processing. The NIC also checks the integrity of a captured frame (error control). So, a NIC can be regarded as a combination layer 1/layer 2 device.

In addition to performing layer 1 and layer 2 functions, new "smart" NICs off-load packet data processing tasks from the host's CPU leaving the host to manage only session-level data. Smart NICs are only capable of processing nonfragmented TCP connections. This implies that the local host still must take care of exceptions such as error control (e.g., retransmissions), reassembly of out-of-order segments, and establishing/terminating TCP sessions. Nevertheless, given that TCP operates in LANs nearly 100% of the time without experiencing any exceptions, smart NICs are more efficient than traditional NICs and hence can improve a system's overall performance. Candidates for these new adapters include servers or other hosts that process large amounts of network data (Figure 6.12).

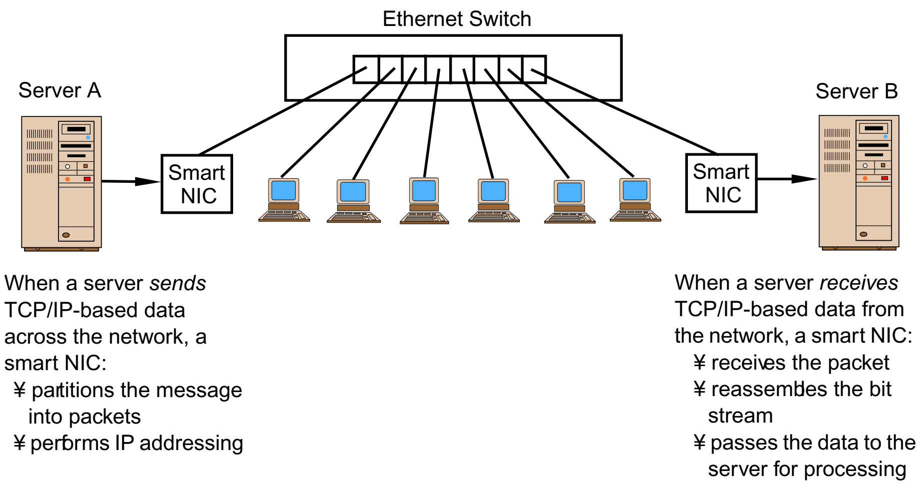


FIGURE 6.12 A smart NIC helps its host preserve CPU cycles. Assuming there are no errors during transmission, a smart NIC performs TCP/IP packet processing in hardware and leaves exception handling to the host software. This strategy improves host processor efficiency, increases data throughput, and reduces latency. Source: Adapted from Gervais, 2000.

27. Are NICs part of a computer system, or do they have to be installed separately?

It depends. Some are built-in as an integral part of the computer's design. However, many desktop PCs, including Intel-based and Macintosh units, are not always manufactured with built-in network support. In such computers a network adapter is needed in order to connect the PC to a LAN. Add-in NICs exist for nearly all expansion slot types used in computers manufactured today.

28. What does a NIC installation involve?

Installing a NIC is relatively straightforward—insert the card into an existing expansion slot, install special network "driver" software to enable the NIC to communicate with the networking software, reboot the machine, and test the card. On Intel-based (or compatible) machines, a NIC also must be assigned an I/O address and an interrupt request (IRQ). These assignments must be made to prevent the NIC from conflicting with other installed boards. (This is not necessary for Macintosh computers.) Many of the reputable NIC manufacturers do this via software; others require you to set DIP switches on the card. Many NICs support Plug-n-Play (PnP)—that is, they comply with the PnP specifications—which enables them to be configured automatically without user intervention.

29. What do you mean by "special network 'driver' software"?

A NIC is a network device that must communicate with its host's network operating system. This communication occurs via software called a *LAN driver*. A driver is critical to the operation of a NIC and must be written so the card supports the appropriate network protocols and operating system. Incorrect LAN drivers are invariably the source of network performance problems. Normally, a NIC requires a separate driver for each combination of network protocol and operating system. However, two specification standards—the Network Driver Interface Specification (NDIS) and the Open Data-link Interface (ODI)—provide generic interfaces that reduce the number of drivers required. Thus, a NIC that supports either NDIS or ODI only needs a single driver and can support multiple network protocols including TCP/IP, IPX/SPX, and NetBIOS. Similarly, standard driver specifications exist for MacOS's Open Transport networking.

30. Are all NICs the same? In other words, does it matter which brand I purchase?

Although most NICs, regardless of vendor, have a high degree of interoperability (e.g., an Ethernet card from one manufacturer most likely will communicate with an Ethernet card from a different manufacturer), all NICs are not the same.

31. So what do I need to consider when purchasing a NIC?

Several things. First of all, make sure the NIC's data bus is appropriate for the host system. For example, you don't want to use an 8-bit NIC in a Pentium server. At the same time you don't want to use a 32-bit NIC in a 386 workstation. Second, some NICs include an on-board processor, on-board RAM, or both. A NIC with an on-board processor is able to do more work than one without this feature and consequently relieves the host system's processor from performing certain functions. The on-board RAM provides a NIC with additional buffer space, which improves communication between the host system and the network. Assuming all things are equal and you can afford it, a 32-bit NIC with on-board processing and RAM will perform faster than a NIC without these features. Third, some NICs include on-board LEDs. For example, a UTP-based Ethernet card might contain LEDs for link status, collisions, and activity. These LEDs can provide valuable diagnostic information about the card's state or network activity. Another consideration is whether the card supports *auto-sensing* for 10/100 Mbps Ethernet/802.3 LANs, or if the card supports full-duplex networking. You should also make sure that the card you purchase is compatible with the host system's bus architecture. Thus, if your PC is an ISA bus machine or a PCI bus machine, make sure the NIC you purchase is ISA- or PCI-compatible. Finally, the drivers that come with the card must support your network and operating system. Ideally, the drivers should be either ODI- or NDIS-compliant, or both, for Intel/Windows systems.

32. I have a laptop computer and use a PC card for network connectivity. Is this considered a NIC?

A *PC card* (Figure 6.13) serves the same purpose as a NIC, namely, to effect communication between a node and the network. So, yes, it is a type of NIC.

33. Can you tell me a little more about PC cards? Are they same as PCMCIA cards?

PCMCIA cards—PCMCIA stands for Personal Computer Memory Card International Association—were originally designed to serve as memory cards (and thus their name). They have since evolved into multipurpose plug-in devices and are today called PC cards. These devices are small (about the size of a credit card, only thicker) plug-in adapters used in portable or laptop computers. Three different "types" are available. *Type 1* cards, the earliest, are only 3.3 millimeters thick and enhance the memory capabilities of a device. Memory support includes ROM (read-only memory) and flash memory (a special form of ROM that can be reprogrammed). Some manufacturers use Type I cards for software upgrades. *Type II* cards are 5 mm thick and used for modems and network adapters for both Ethernet and token ring. These devices are similar to NICs and support various media

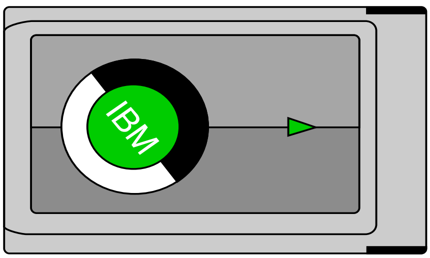


FIGURE 6.13 PC cards are multipurpose plug-in devices for portable computers and can be used as memory devices, modem and network adapters, or hard disk drives.

types including RJ-11 for a modem connection and RJ-45 for UTP-based LANs. Both Ethernet/802.3 and Token Ring adapters are supported. *Type III* cards are 10.5 mm thick and are generally either miniature hard disks or wireless NICs. As of this writing, a *Type IV* card is being considered that will be approximately 16 mm thick and support hard disk drives that have a capacity greater than what is currently available from Type III cards. PC cards are installed into an appropriate PCMCIA slot within a portable computing device. Type III slots also accept Type I or Type II cards, and Type II slots also accept Type I cards. Most laptops manufactured today contain at least one Type III or Type II slot.

34. Let's move on. From the list of layer 2 devices given earlier we still haven't talked about bridges and switches. What are they?

Bridges and switches are similar devices. We'll defer our discussion of switches until later, preferring to cover bridges first because many of the concepts and principles that apply to bridges also apply to switches. A network *bridge* interconnects two or more individual LANs or LAN segments. Unlike repeaters, which are layer 1 devices—bridges connect networks that have different physical layers. This is what makes them layer 2 devices—the physical layer is transparent to bridges. Bridges also can connect networks using either the same or different type of architectures (e.g., Ethernet-to-Ethernet, token ring-to-token ring, or Ethernet-to-token ring) (Figure 6.14). Two bridge standards have been defined by IEEE. The first is a *transparent bridge* and is used in IEEE 802.3 ("Ethernet") and 802.5 (token ring) networks. The second is a *source routing bridge*, which was introduced by IBM and used exclusively in token ring networks.

35. What do bridges do?

Bridges pass frames between LANs and provide filtering. They allow frames from a node on one network to be forwarded to a node on another network, but discard any frames destined for the same network from which the frames originated. Thus, bridges keep local traffic local, but forward traffic destined for a remote network. Since bridges operate at the data link layer, they check the hardware (i.e., the MAC-level) address of a particular network interface card to determine whether to forward or discard a frame.

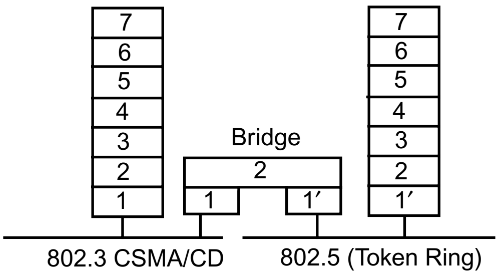


FIGURE 6.14 OSI representation of a bridge. Bridges can connect networks using different architectures such as Ethernet and token ring.

36. How does a transparent bridge work?

A transparent bridge is a "plug and play" unit—you connect it to your network and power it on. Operating in *promiscuous mode*, a transparent bridge captures every frame that is transmitted on all the networks to which the bridge is connected. The bridge examines every frame it receives and extracts each frame's source address, which is then added to a "learned address" table maintained by the bridge. Eventually, this table contains an entry for each unique source address and the port on which the frame was received. For example, in Figure 6.15, when node 1 (with a hardware source address of 1) transmits a frame, the bridge "learns" that node 1 is on channel A (i.e., port A) and adds a corresponding entry to its address table. The bridge also examines the destination address of the frame to determine if the frame should be forwarded or filtered. All broadcast and multicast frames always get forwarded. If the destination address is not a broadcast or multicast address, and it is not found in the bridge's address table, the frame is forwarded by default. This is what the bridge does when it is first powered on and its address table is empty. This procedure is referred to as "flooding." The only condition under which a frame does not get forwarded (i.e., it is discarded or filtered) occurs when the frame's destination address is found in the address table and corresponds to the same channel on which the frame was received. So if node 1 transmits a frame to node 3 in Figure 6.15, the bridge does not forward the frame to Channel B; the destination address (3) is a source address on the same channel on which the frame originated. On the other hand, if the destination is node 5, or if the frame is a broadcast or multicast, then the bridge forwards the frame.

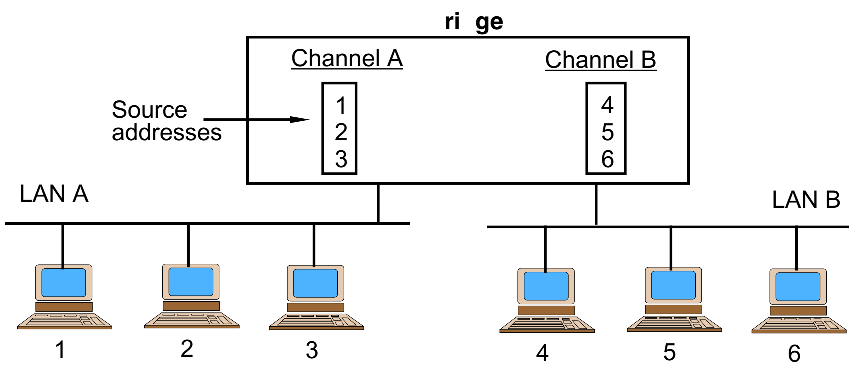


FIGURE 6.15 Bridges interconnect separate networks, making them appear as a single network. Operating at the data link layer, a bridge builds a table of hardware addresses that identifies the address of each node and the segment to which a node is attached. Using this table, a bridge either forwards a frame from one network to the other or discards the frame depending on whether the destination node is local or remote. For example, a frame from node 3 to node 1 does not get forwarded to channel B. Thus, none of the nodes on LAN B "sees" node 3's transmission. However, a frame from node 3 to node 5 is forwarded from channel A to channel B. A bridge will, however, forward any broadcast frames from LAN A to LAN B, and vice versa.

37. How does a source-routing bridge differ from a transparent bridge?

As we mentioned earlier, the source routing bridge was introduced by IBM for use in token ring networks. With source routing, the sending machine is responsible for determining whether a frame is destined for a node on the same network or on a different network. If the frame is destined for a different network, the source machine designates this by setting the high-order bit of the group address bit of the source address to one. It also includes in the frame's header the path the frame is to follow from source to destination. Source routing bridges are based on the assumption that a sending machine will provide routing information for messages destined for different networks. By making the sending machine responsible for this task, a source routing bridge can ignore frames that have not been "marked," and forward only those frames with their high-order destination bit set to one. An illustration of source routing bridges is given in Figure 6.16.

38. I have heard the term "spanning tree" used with bridges. What does this mean?

For reliability, some networks contain more than one bridge, which increases the likelihood of *networking loops*. A networking loop occurs when frames are passed from bridge to bridge in a circular manner, never reaching their destination. To prevent networking loops when multiple bridges are used, the bridges communicate with each other and establish a map of the network in order to derive what is called a *spanning tree* for all the networks. A spanning tree consists of a single path between source and destination nodes

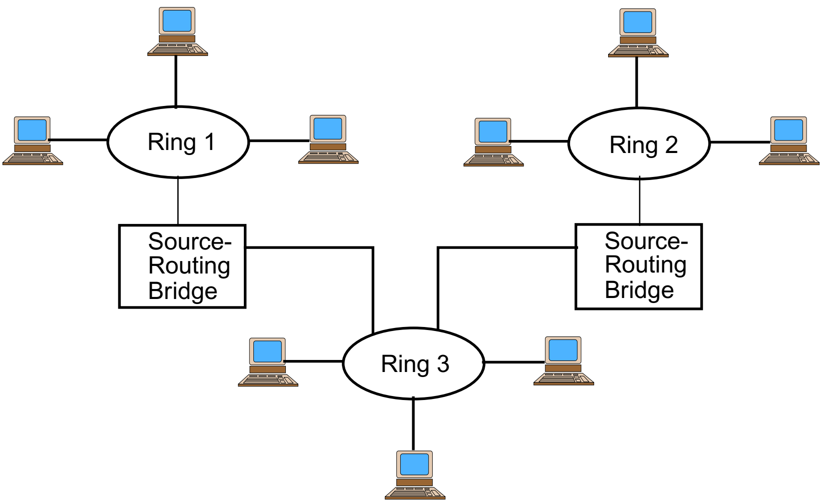


FIGURE 6.16 Bridges used to interconnect separate token ring networks are called source-routing bridges. As is the case with transparent bridges used in Ethernet/802.3 networks, source-routing bridges copy frames from one ring to another. They also retransmit frames to the next station on the same ring. Unlike transparent bridges, source-routing bridges depend on the sending station to provide routing information for frames destined for a different network.

that does not include any loops. Thus, a spanning tree can be considered a loop-free subset of a network's topology. The spanning tree algorithm, specified in IEEE 802.1d, describes how bridges (and switches) can communicate to avoid network loops.

39. Can you give me an example of what you mean by a network loop?

Sure. Look at Figure 6.17, which has four network segments interconnected by four bridges. LAN 1 is connected to LAN 2 by B1; LAN 2 is connected to LAN 3 by B2; and LAN 3 is connected to LAN 4 by B4. Note also that LAN 1 is connected to LAN 4 by B3. Thus, multiple bridges are being used on LAN 1. Let us assume that a frame originates on LAN 1 and none of the bridges (B1-B4) has an entry for the frame's destination. Here is an example of how a network loop can develop:

1. A frame originates on LAN 1; neither B1 nor B3 has the destination address as an entry so neither know on what network the destination node resides. Hence, B1 and B3 must forward the frame to their respective LANs. That is:
   - B1 forwards the frame to LAN 2
   - B3 forwards the frame to LAN 4

2. The frame is now on LAN 2 and LAN 4. Once again, neither B2 nor B4 knows on what LAN the destination node is located since they do not have the destination address in their tables:
   - B2 forwards the frame to LAN 3
   - B4 forwards the frame to LAN 3

3. The frame is now on LAN 3, having come from different LANs, and B2 still does not have the destination address as an entry in its table:
   - B2 forwards the frame to LAN 2
   - B4 forwards the frame to LAN 4

4. The frame is now on LAN 2 and LAN 4, and neither B1 nor B3 knows the location of the destination node's LAN:
   - B1 forwards the frame to LAN 1 (We have a network loop.)
   - B3 forwards the frame to LAN 1 (We have a network loop.)

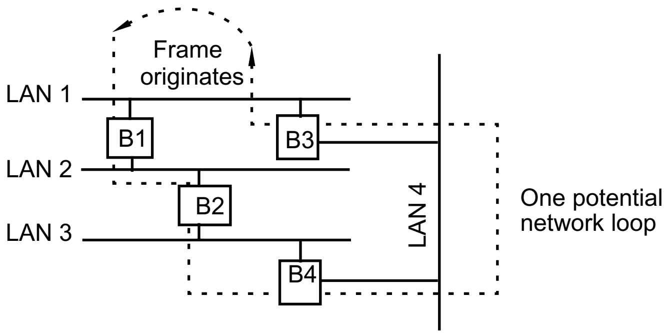


FIGURE 6.17 A possible network loop.

40. So how does the spanning tree algorithm help?

The spanning tree algorithm would disable the port on B3 that connects LAN 4 with LAN 1. If B4 were to fail, the algorithm would then automatically enable the LAN 1-to-LAN 4 connection via B3.

41. Does it matter what network protocol you are using when you install a bridge?

No. Bridges are transparent to protocols operating at higher layers. This means that regardless of the network protocol being used, bridges will pass or discard frames independent of the protocol. Thus, two networks based on different protocols connected to the same bridge are viewed as a single logical network. Bridges also force the repeater count in an Ethernet/802.3 network (see Chapter 8) to return to one. Bridges, however, are highly susceptible to *broadcast storms* since they will pass broadcast frames from one network to another. A broadcast storm occurs when several broadcasts are transmitted at the same time. Broadcast storms can use up a substantial amount of network bandwidth, and in many cases, can cause a network to crash or shut down.

42. I have heard the term "store-and-forward" used when discussing bridges. What does this mean?

Bridges are indeed *store-and-forward* devices. What this means is they capture an entire frame before deciding whether to filter or forward the frame. This provides a high level of error detection because a frame's CRC checksum can be computed by the bridge and matched to that of the frame. If the CRC checksum is not correct, the frame is dropped. This prevents the propagation of corrupt frames.

43. If I had to choose between using a repeater or a bridge, which should I select?

Neither. You would want a switch (see the discussion below). In the rare instance where you do need to select between a repeater or bridge, though, the answer is, "it depends." The purpose of a repeater is to extend a network. The purpose of a bridge is to segment network traffic. In choosing between a repeater and a bridge you need to determine the manner in which the device will be used. Repeaters and bridges also provide electrical isolation. That is, if the device fails, then nodes on the same physical segment can still communicate with each other, but nodes on different segments cannot. So, if the purpose is to electrically isolate segments, then either device will do the job.

Bridges, unlike repeaters, also can connect networks in different geographical locations that require a telecommunications link for connectivity. Wireless bridges also have become available for limited-distance remote connections. Remote bridges that use radio waves can be placed up to 25–30 miles (40–48 km) apart, provided the terrain and weather allow for it, and if the bridges have directional antennas. Remote bridges using laser communication techniques can be spaced approximately 3500 feet (1050 m) apart.

44. Does replacing a repeater with a bridge improve Ethernet LAN performance?

Not necessarily. The performance of a bridge is measured by the number of frames per second it can forward (the *forwarding rate*) and the amount of delay that is introduced due to forwarding (the *forwarding delay*). In an ideal setting, an Ethernet/802.3 bridge forwards 14,880 64-byte frames per second. This is referred to as *wire speed*. (See Chapter 8.) Furthermore, an ideal Ethernet/802.3 bridge has a 60 μs forwarding delay for 64-byte frames. The propagation delay of a repeater is less than 3 μs. Also keep in mind that a repeater propagates errors but a bridge does not.

Before replacing a repeater with a bridge, you must design the network properly based on traffic patterns. For example, in a repeater-based LAN, it does not matter where a server is located. However, in a bridge-based LAN, if a server is placed on one physical segment, and all nodes that must communicate with the server are on a different segment, then performance will be worse than the repeater-based configuration. The concept of network segmentation and partitioning is discussed in further detail in Chapter 8.

45. What's a brouter?

A *brouter* is a combination bridge-router; it is basically a bridge that has routing capabilities. A brouter is considered a legacy component and is briefly addressed in Chapter 7.

46. OK. Let's move on to switches. What are they and how different are they from bridges?

We assume you are interested in learning about Ethernet switches in particular.

47. Yes. Are there others?

Yes. Although LAN switches were initially designed for Ethernet/802.3 networks, the concept of switching has since been extended to other LANs. We'll discuss the other types in later chapters.

48. Back to Ethernet switches, please.

Conventional Ethernet switches are layer 2 devices that are essentially modified multiport bridges. Like bridges, each port on an Ethernet switch supports a separate LAN segment, and each port can accommodate different media including ThinWire, UTP, and fiber-optic cable. Furthermore, each switch port filters traffic sent over its attached segment. Thus, traffic destined for a node on the same segment does not cross the switch's port boundary; it remains local to that segment. Furthermore, if a node on one segment sends frames to a node connected to a different segment, that is, a different switch port, the frames are forwarded across the port boundary and through the switch to the appropriate destination port without any other port seeing the transmission.

49. It sounds like an Ethernet switch is the same as a bridge except for its name. What's different about switches?

What makes switches different from bridges is their architecture. Repeaters and bridges are designed for shared media LANs; the architecture of switches, however, permits multiple, simultaneous data transmission paths between ports. Each switch port is assigned a specific MAC address, with data paths between ports being hardwired and part of the switch's internal circuitry (called the *switch fabric* or *switch matrix*). When a data frame enters a switch port, the port's network adapter translates the MAC destination address of the frame to a specific switch port address and then transfers the frame to that MAC-specified destination port. Thus, the data transmission in a switch is based on a static port-to-MAC address association. Some switches support only one MAC address per port; others support more than one MAC address per port. The bottom line: Nodes connected to bridges share bandwidth; nodes connected to switches do not share bandwidth—they have "private" connections.

50. What are the various types of switches?

Ethernet switches have three basic design architectures: *store-and-forward, cut-through,* and *hybrid*.

51. Is a store-and-forward switch the same thing as a store-and-forward bridge?

Pretty much so. A switch that incorporates the store-and-forward design (sometimes referred to as a *buffering switch*) operates exactly like an Ethernet bridge—it waits until it receives an entire data frame before forwarding it. When the switch receives a frame, it first performs an integrity check to ensure that the frame does not contain any errors. As a result, data reliability is excellent in this type of switch since "bad" frames (e.g., incorrect CRC checksums) are never forwarded. After checking for errors, the switch extracts the destination address from the frame's address field, performs an address table lookup to identify the destination port to which the frame should be sent, and forwards the frame to the destination port if it is different from the port at which the frame arrived. (If the port is the same, the switch discards the frame.) An illustration of a store-and-forward switch that contains individual port buffers is given in Figure 6.18.

52. What's different about the cut-through architecture?

The cut-through architecture is what really separates switches from traditional store-and-forward bridges. A *cut-through switch* operates in the following way: If a frame arriving at one port in the switch is to be transmitted to a different port, the switch begins this transmission as soon as it reads the destination address of the frame. This technology improves Ethernet performance considerably by reducing delays.

Cut-through switches can be implemented using either a crossbar or backplane design. A *crossbar design* identifies the frame's destination address and the path within the switch the frame must follow to get to the destination port. Once these have been determined, the switch transfers the part of the frame it has already received (the preamble, start frame delimiter and destination address) to the destination port. All remaining parts of the frame (source address, length count, data, pad, and checksum) are immediately transferred as they are received by the switch via this same data path. An illustration of a crossbar switch is shown in Figure 6.19. Note that this design can introduce delay if the data path is not clear for transmission. For example, suppose a frame arrives from segment 2 destined for segment 3. If the path to segment 3 is busy (e.g., a transfer might be occurring from segment 1 to 3), then the frame must remain in segment 2's buffer until the path is clear. This can "back up" traffic on segment 2.

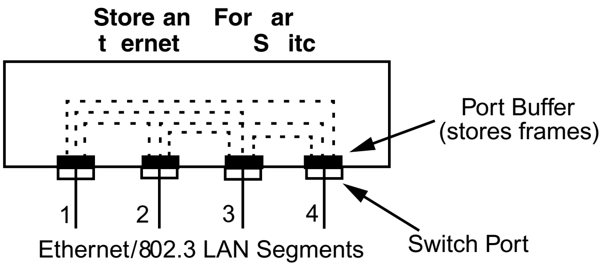


FIGURE 6.18 A store-and-forward Ethernet switch has buffers at each port. When a frame enters a port from a segment, the switch stores the frame in that port's buffer until the entire frame is received and checked for errors. If the frame is error-free, the destination address is identified and the frame is placed at the destination segment's port without any other port seeing the transmission. Source: Adapted from Majkiewicz, 1993, and Sharer, 1995.

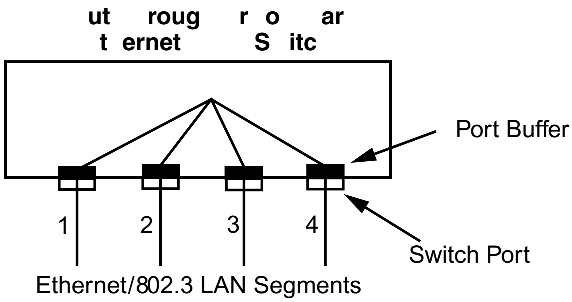


FIGURE 6.19 Cut-through Ethernet switches transmit frames as soon as the destination address is known. In the crossbar design, the data paths connecting the ports are all interconnected. If this path is busy with a current transmission, delays will occur. Source: Adapted from Majkiewicz, 1993, and Sharer, 1995.

In contrast to the crossbar approach, the *backplane design* places frames on a high-speed backplane, which interconnects all ports. If the destination port is free, frames are immediately transferred. If the destination port is busy, frames are buffered onto the backplane until the port is clear. This eliminates the potential congestion problem of the crossbar approach. The key to this design is the switch's backplane, which requires a data rate greater than the aggregate throughput of the switch. For example, an eight-port 10 Mbps Ethernet switch has a total throughput of 80 Mbps. Given that a transmission involves at least two nodes, it is possible to have as many as four simultaneous transmissions occurring in parallel. To avoid bottlenecks, a 10 Mbps switch must be able to handle at least 40 Mbps of aggregate data flow. Typically, a cut-through switch based on the backplane design has a backplane that is at least equal to the total aggregate throughput of the segments. In our illustration, this would be 80 Mbps. Some switches have gigabit per second backplanes. Viewed from this perspective, switches do not actually increase the speed of a 10 Mbps segment. Rather, they increase the aggregate throughput capability of a network. Thus, switches simply are high throughput devices that provide the capacity for multiple segments to operate concurrently. An illustration of the backplane design is shown in Figure 6.20. Cut-through switches generally are more expensive than store-and-forward switches because of their more sophisticated circuitry. (*Note:* Ethernet cut-through switches, which are discussed in Chapter 8, neither check CRC checksum values nor minimum frame lengths. Thus, a cut-through switch will propagate "bad" frames throughout an Ethernet/802.3 network.)

53. What's a hybrid switch?

A *hybrid switch* integrates the best features of store-and-forward (reliable frame transmission) and cut-through (low latency) designs. A hybrid switch can be configured on a per-port basis to change automatically from cut-through switching to store-and-forward switching if error rates exceed a user-defined threshold. When error rates fall below this threshold, the switch reverts to cut-through switching. An additional capability is a "runt-free" mode in which the switch discards frames smaller than the mandated 64 bytes minimum size for Ethernet. This ensures the filtering of collision fragments while maintaining the low latency characteristics of cut-through switching.

54. How does the performance of bridges and switches compare?

Switches provide high throughput with low or fixed *latency.* Used in this context, latency is the amount of delay a network device introduces when data frames pass through it. Thus, latency is the amount of time a frame spends inside a network device such as a bridge or switch. Switch latency is usually measured from the instant the first bit of a

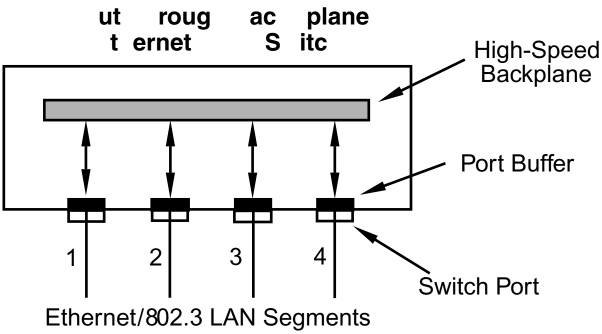


FIGURE 6.20 Backplane-based cut-through Ethernet switches also transmit frames as soon as the destination address is known. In the backplane design, though, data frames are placed on a high-speed backplane for transmission. The speed of this backplane is greater than the aggregate throughput of the switch. If a destination port is busy, the frame remains on the backplane; this eliminates the kinds of delays inherent in the crossbar design. Source: Adapted from Majkiewicz, 1993, and Sharer, 1995.

frame enters the device to the time this bit leaves the outbound (i.e., destination) port. Depending on a switch's architecture, its latency is usually less than 100 μs. In comparison, bridge latency of 400 μs is common, and routers have latency as high as 1500 μs (i.e., 1.5 ms). Low latency is good because the lower it is the faster a device processes a data frame. If latency is too high, then time-sensitive network protocols such as SNA or IPX can time out. Latency also is an issue for time-sensitive applications such as full-motion video.

55. How does the performance of different switch architectures compare?

Store-and-forward switches provide both traffic isolation and error immunity to any destination port. The trade-off for error immunity, though, is high latency, which can be in milliseconds rather than microseconds. Since store-and-forward switches wait until an entire frame has arrived before forwarding it, latency, which is dependent on frame size, can range from 61 μs to 1200 μs over a 10 Mbps channel. This could be a serious concern for some applications.

Cut-through switches have extremely low latency, usually on the order of 20 to 40 μs. Furthermore, since cut-through switches begin forwarding frames the moment the destination address is known, latency is independent of frame size. Thus, a maximum-size Ethernet/802.3 frame (1518 bytes) has the same latency as a minimum-size Ethernet/802.3 frame (64 bytes). This is not the case with store-and-forward switches where latency varies directly with frame size. Considering that the total round-trip bit propagation speed of a standard 10 Mbps Ethernet/802.3 network is approximately 51.2 microseconds, the speed of a cut-through switch approximates cable or wire speed, which is 14,880 packets per second using 64-byte packets. (See Box 8.3 in Chapter 8.) The trade-off, though, is there is no opportunity for the switch to check the integrity of a frame before it is forwarded to its destination port. Thus, if corrupt data messages are contained in the fields that follow the destination and source address fields of a data frame, the frame is still forwarded to the destination port. Errors on a specific port are propagated to other ports on the switch, or throughout the network. This reduces bandwidth and can delay forwarding error-free frames.

56. Besides the switching fabric, is there anything else I need to know about switches to increase my switch-knowledge coefficient?

Yes. Two two other switch-related components, which we informally addressed in our discussion of the switching fabric, are queuing and implementation. A switch's *queuing model* deals with congestion management, which is needed when more than one input port contends for the same output port within the switch. Some switches use fixed buffers, which are aligned to a specific maximum transmission unit (MTU) size; other switches use dynamic buffers, which partition individual buffers into smaller sizes. A dynamic buffer architecture is more efficient than a fixed buffer architecture because there is less wasted memory. With dynamic buffers, a switch can allocate memory commensurate with a frame's size. A fixed buffer approach, however, wastes memory because frames that are less in size to the fixed buffer space occupies the entire buffer. Buffers can be placed at input ports (called input queuing), output ports (called output queuing), or centrally accessible by all ports (called shared-memory queuing). Each location strategy has its advantages and disadvantages. Finally, some switches also have multiple queues per port, which are needed to address QoS issues (see Chapter 5).

As for a switch's *implementation model*, this refers to the manner in which switching decisions are made. For example, a centralized switch model relies on a central forwarding table that is accessible by all ports, whereas a distributed switching model enables ports to maintain their own lookup tables, which are synchronized with a master address table.

57. I have heard and read about "layer-3 switching" and "layer-3 switches." Frankly, I am a little confused by these concepts. Could you shed some light on this?

Sure, but not here. This is discussed in Chapter 7 when we examine the differences between routers and switches. For the moment, though, suffice it to say that Layers 2 and 3 of the OSI model are merging and it is becoming difficult to distinguish between traditional layer-2 devices such as LAN switches and layer-3 devices such as routers.

END-OF-CHAPTER COMMENTARY

This concludes our discussion of layer 1 and layer 2 network components. In subsequent chapters some of the concepts discussed in this chapter are extended. For example, in Chapter 7, we discuss WANs, internetworking, and network layer (layer 3) concepts and components. Two topics that are discussed in Chapter 7 that were referenced in the current chapter are routers and switches. Another topic of this chapter, Ethernet switches, is expanded on in Chapter 8. The topic of switching is also extended in later chapters. For example, token ring switching is discussed in Chapter 9, FDDI switching is discussed in Chapter 10, frame relay switching is discussed in Chapter 11, and ATM switching is discussed in Chapter 15. Finally, for a review of layer 1 or layer 2 concepts, refer to Chapters 4 and 5, respectively.

Chapter 7

Internetworks and Network Layer Concepts and Components

In Chapters 4, 5, and 6, we introduced many of the concepts and hardware components related to layers 1 and 2 of the OSI model. Our discussion in these chapters also was relative to local area networks. In this chapter, we extend our previous discussions to wide area links. As part of this presentation, we introduce the concept of internetworking from a WAN perspective and discuss the third layer of the OSI model—the network layer. An outline of the main topics we address follows.

- The Concept of Internetworking (Questions 1–4)
- WAN Circuits (Questions 5–12)
- SONET (Questions 13–15)
- Layer 3 Concepts and Issues (Questions 16–18)
- Router Protocols and Routing Algorithms (Questions 19–32)
- Routing vs. Switching (Questions 33–36)
- Virtual Private Networks (VPNs) (Questions 37–45)
- Multiprotocol Label Switching (MPLS) (Questions 46–49)

1. What is an internetwork?

In Chapter 1, we defined a computer network as a collection of computers and other devices that use a common network protocol to share resources with each other over a network medium. We also indicated that if a collection of computer networks are connected to one another, then we have what is known as a network of networks, which is formally called an *internetwork*. Thus, an internetwork refers to a collection of interconnected networks that function as a single network. Furthermore, the individual networks comprising an internetwork are called subnetworks, the devices connected to a subnetwork are called end nodes (or end systems), and the devices that interconnect subnetworks are called intermediate nodes (or intermediate systems). An internetwork can involve local networks (e.g., LAN-to-LAN or LAN-to-main-frame connections), long-distance connections between networks requiring WAN connections (e.g., LAN-to-WAN connections), and WAN-to-WAN connections. (See Chapter 1 for additional information about LANs and WANs.) For example, a network located in an office on one floor of a building can be con-

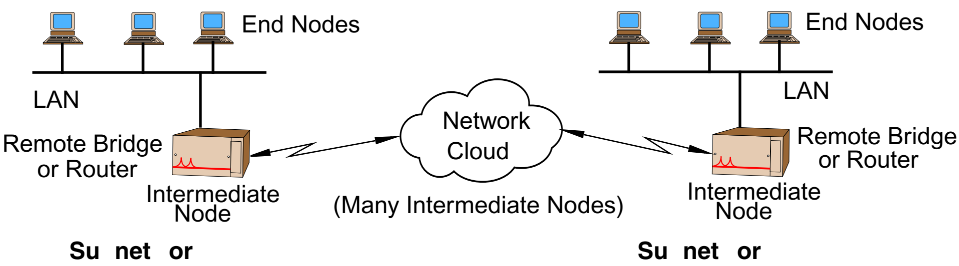


FIGURE 7.1 A wide area network interconnects geographically separated LANs using a point-to-point topology. A WAN transmission facility consists of remote bridges or routers and a data communication circuit. What happens inside the WAN cloud is a function of the network protocol and technology used.

nected to another network located on a different floor of the same building. Collectively, these two interconnected networks represent an internetwork. Figure 7.1 shows a simple internetwork involving a LAN-to-LAN connection. A good illustration of an internetwork is a college campus network, which typically consists of several autonomous departmental LANs interconnected via a campuswide backbone. In this configuration, the departmental LANs are considered subnetworks, and the overall college network is considered an internetwork. Recall also from Chapter 1 that the Internet is the world's largest internetwork and is sometimes thought of as a *wide area internetwork* (WAI).

2. How are internetworks formed?

The manner in which networks are connected to one another depends on the type of networks involved. To help guide us in determining which method to use, networks are usually classified in one of three ways relative to the concept of internetworking: identical, similar, and dissimilar. *Identical networks* employ exactly the same architecture and cabling; *similar networks* have different architecture or different cabling; and *dissimilar networks* employ different hardware, software, protocols, and often support different functions or applications. Dissimilar networks are sometimes referred to as *heterogeneous networks*. It should be obvious that the internetworking strategy for interconnecting identical networks is different from interconnecting similar or dissimilar networks.

There are three general ways in which internetworks are formed, and each method is closely aligned with the bottom three layers of the OSI model. More specifically, identical networks can be interconnected at the physical layer (layer 1), similar networks can be interconnected at the data link layer (layer 2), and dissimilar networks can be interconnected at the network layer (layer 3).

3. What about the hardware devices used for these connections?

The hardware devices used to link identical, similar, and dissimilar networks also operate at the respective layer. For example, in Chapter 6, we learned that repeaters are used to extend the length of a LAN. Since repeaters operate at the physical layer, the LANs that are interconnected must be exactly the same. Thus, a repeater provides connections at the physical layer and works with only a specific LAN architecture, as shown in Figure 7.2(a).

We also learned in Chapter 6 that a bridge and a layer-2 switch provide connections at the data link layer. These devices support different physical layers and can interconnect different LAN architectures, as in Figure 7.2(b). Bridges and layer-2 switches also are network-protocol independent, which enables them to interconnect networks using different network protocols (e.g., TCP/IP and IPX). Unlike routers (discussed below), which operate at layer 3 and do protocol translations, though, bridges and layer-2 switches ignore layer-3 information. Interconnectivity is achieved by forwarding frames between networks by using hardware addresses, not network addresses. In this context, an internetwork consisting of two dissimilar networks connected by a bridge or switch can be viewed as a single logical network. (*Note:* Different LAN architectures require different bridging techniques. For example, transparent bridging for Ethernet/802.3, source routing bridging for token ring or FDDI, and transparent/source routing bridging for mixed environments. Review Chapter 6 for more information about these different bridging methods.)

Internetworks that use layer-3 devices can have different physical and data link layers, as in Figure 7.2(c). Layer 3 devices also can support different network protocols (e.g., IP, IPX, AppleTalk). Interconnectivity is achieved by forwarding packets from one network to another using network layer information (e.g., network addresses). In a heterogeneous networking environment in which dissimilar networks are interconnected, layer-3 devices perform *network protocol translation*. One device that provides connections at the network layer is called a *router*. (*Note:* Another device that operates at the network layer is a layer-3 switch, which we discuss later in the chapter.) Thus, a router's job is to interconnect physically different networks and route packets from one network to another. It determines a path to a destination node and then starts the packet on its way. Routers that support more than one network layer protocol are called multiprotocol routers. So if a packet originating from an IP network is passed to an AppleTalk network, a multiprotocol router rebuilds (or reformats) the packet to the proper form so it can be interpreted by a node on the AppleTalk network. Since routers operate at the third layer of the OSI model, they do not forward broadcast packets unless they are configured to do so. Routers also employ routing protocols that determine the least-cost path a packet is to travel from source to destination nodes.

4. I think an example would help me understand these concepts a little better.

As an example, consider a simple WAN that interconnects two LANs. A typical scenario involves two remote bridges or routers (one at each end) interconnected via a WAN data communications circuit similar to that shown in Figure 7.1. WANs use either circuit-switching or packet-switching techniques. (See Chapter 2 and Table 2.1 for more information about packet- and circuit-switching.) In a circuit-switched WAN, a fixed connection is established between source and destination nodes prior to transmission, each packet takes the same path, and all packets arrive in sequence. ISDN (see Chapter 12) is one example of a circuit-switched WAN. In a packet-switched WAN, connections are established during the transmission process. Thus, packets do not necessarily travel the same route, and they might arrive out of sequence at the destination node. Examples of a packet-switched WAN

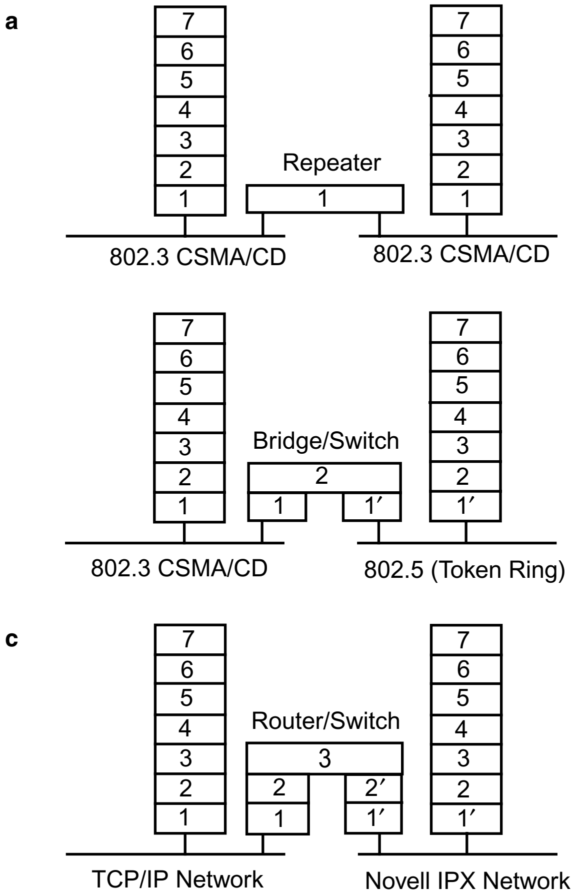


FIGURE 7.2 An OSI representation of internetworking. In a repeater-based internetwork (a), the LAN architecture must be the same. A bridge- or switch-based internetwork (b) has different physical layers and connects different types of LAN architectures. (*Note:* MAC-based bridges can only interconnect similar LAN architectures.) Bridges and layer-2 switches are also network protocol independent. A router- or layer-3 switch-based internetwork (c) operates at layer 3 of the OSI model and interconnects networks using different network protocols. These devices perform protocol translations.

include frame relay, SMDS, and ATM. As a result, what happens inside the network cloud shown in Figure 7.1 depends on the technology used. In later chapters, we will peek inside this cloud and examine exactly what is taking place when we discuss some of these technologies.

5. My company's Internet connection is via a leased T1 line. Is this a WAN link?

Yes it is. A leased line is a dedicated connection between two sites.

6. Can you tell me something about T1 lines? I know they are 1.544 Mbps but I don't really know anything else about them.

We'll be glad to. The T1 terminology was originally defined by AT&T and describes the multiplexing of 24 separate voice channels into a single, wideband digital signal. A T1 frame consists of 193 bits—eight bits per channel plus one bit for framing. Bits 1 through 8 are dedicated to channel 1, bits 9 through 16 are dedicated to channel 2, and so forth.

Each voice channel is rated at 64 kbps. When multiplexed into a digital signal, a voice channel is referred to as a *digital signal at level 0* (DS-0). Thus, DS-0 represents a single, digital voice channel rated at 64 kbps. A T1 circuit carries a DS-1 signal, which consists of 24 DS-0 channels plus one 8 kbps channel reserved for framing. This results in an aggregate bandwidth of 1.544 Mbps.

The T1 concept eventually evolved to what is known as the North American Digital Hierarchy (NADH), which consists of multiplexed T1 lines. For example, two T1 lines are combined to form a T1C circuit rated at 3.152 Mbps. (In DS terminology, T1C is known as DS-1C.) This is more than twice 1.544 Mbps because NADH uses bit-stuffing (see Chapter 5), which increases the aggregate bandwidth. A T2 circuit (DS-2) consists of four multiplexed T1 circuits and has an aggregate bandwidth of 6.312 Mbps; a T3 link (DS-3) consists of 28 multiplexed T1 circuits with an aggregate bandwidth of 44.736 Mbps; and a T4 channel (DS-4), rated at 274.176 Mbps, consists of 168 multiplexed T1 circuits. Table 7.1 provides a summary of this hierarchy.

7. What's the difference between T1 and Digital Signal terminology?

In practical terms and everyday usage, T1 and DS-1 are considered synonymous.

8. What terminology is used outside of North America?

T1 service has the same meaning in Australia and Japan as it does in North America (U.S. and Canada). However, in Europe, South America, Africa, parts of Asia, and Mexico,

TABLE 7.1 Summary of NADH Line Rates

Digital Signal	T-Carrier	Data Transmission Rate (Mbps)	Number of Multiplexed DS-0 Channels	Number of Multiplexed DS-1 Channels
DS-0	—	0.064	1	—
DS-1	T1	1.544	24	1
DS-1C	T1C	3.152	48	2
DS-2	T2	6.312	96	4
DS-3	T3	44.736	672	28
DS-4	T4	274.176	4032	168

TABLE 7.2 Summary of European E-Carrier Line Rates

E-Carrier	Data Transmission Rate (Mbps)	Number of 64 kbps Channels	Number of E-1 Channels	Number of E-2 Channels	Number of E-3 Channels	Number of E-4 Channels
—	0.064	1	—	—	—	—
E-1	2.048	30	1	—	—	—
E-2	8.448	120	4	—	—	—
E-3	34.368	480	16	4	—	—
E-4	139.264	1920	64	16	4	—
E-5	565.148	7680	256	64	16	4

T1 is meaningless. Instead, an analogous service called E-1 (the "E" for European) is used in these locations. An E-1 carrier is normally supplied as an ISDN PRI circuit, which comprises thirty 64 kbps voice channels plus two 64 kbps channels for control ($30B + 2D$). The aggregate bandwidth of an E-1 carrier is 2.048 Mbps. (See Chapter 11 for more information about ISDN.) As with T1 service, E-1 links can be multiplexed into higher-capacity lines. Table 7.2 summarizes this hierarchy.

9. Why was 64 kbps selected as the basic building block of T1/DS-1 circuits?

During the early stages of developing copper-based analog telephone networks, it was discovered that the normal range of frequencies generated by the human voice is from 300 Hz to 3300 Hz. The difference between these frequencies, 3000 Hz, is the amount of bandwidth telephone companies allocated to support the transmission of voice signals. (See Chapter 4 for more information about analog circuits and bandwidth concepts.) In practice, though, telephone companies actually allocated 4000 Hz channels and installed filters at 300 Hz and 3300 Hz. Thus, voice signals that generated frequencies less than 300 Hz or greater than 3300 Hz were discarded.

As digital technology and data applications emerged, analog technology was unable to separate voice or data from noise in a satisfactory manner. This led to the introduction of digital signaling, which requires converting analog signals to digital signals and vice versa. This analog-to-digital conversion process involves two steps: *sampling* and *coding*. Digitizing an analog signal requires regular samples of the amplitude of the signal's waveform to be taken over time so that the generated digital signal matches its corresponding analog signal. According to Nyquist's rule (Box 7.1), analog-to-digital conversions should be done by sampling the analog signal at twice the highest frequency on the line to avoid harmonic distortion. Thus, $3300 \times 2 = 6600$ samples per second are required. However, since telephone companies partitioned their circuits into channels of 4000 Hz, 8000 samples per second are actually taken. This higher sampling rate also provides support for higher voice frequencies. This equates to 125 μsec per sample (1 divided by 8000).

Once a sample is taken, it is then converted into an eight-bit digital code: 00000000, which represents the absence of voltage, or 00000001, which represents the presence of voltage. By using eight bits, 256 (2^8) possible points can be used to partition the wave for sampling (Figure 7.3). Determining whether a sampled point gets coded 0 or 1 depends on where along the wave the sample is taken. This coding process is called *pulse-code modulation* (PCM). The PCM process, which involves a device called a *codec* (coder-decoder), is summarized in Figure 7.4. Multiplying 8000 samples per second by eight bits per sample yields 64,000 bps. Thus, the 64-kbps rate is derived from the early development of analog-to-digital conversions—the digital representation of a single analog voice call requires 64,000 bits.

It is instructive to note the construction of a T1's line rate. As stated earlier, each channel is sampled at a rate of 8000 times per second. This produces an eight-bit number for each sample. Seven of these bits represent data and one bit is used for control. This yields per channel transmission rates of 56,000 bps for data and 8000 bps for control. Since a T1 channel can support 24 simultaneous voice channels via TDM (Chapter 2), we have 1,344,000 bps for data and 192,000 bps for control. Added to this is a separate 8000 bps channel for frame synchronization. This is summarized here.

Data: 56,000 bps per channel at 24 channels = 1,344,000 bps

Control: 8000 bps per channel at 24 channels = 192,000 bps

Framing: 8000 bps for frame synchronization = 8000 bps

This line rate can be further confirmed by observing that a T1 frame contains 193 bits—168 for data, 24 for control, and 1 for synchronization; 193 bits at 8000 samples per second yields 1.544 Mbps. A T1 service is sometimes referred to as a "bit-robbing" service because circuit control is in-band; T1 steals a proportion of the available bandwidth for control purposes. (*Note:* The European standard is based on a 256-bit frame that consists of thirty-two 8-bit time slots. Similar calculations can be done using these values. E-1 circuits are also controlled out-of-band.)

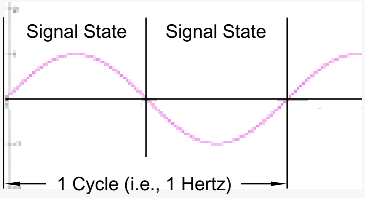
10. OK. I have also heard about fractional T1 service. Does this involve just multiplexing individual DS-0s?

Yes. Fractional T1 service (FT1), as its name implies, provides a fraction of a T1's capacity. This is achieved by combining multiple DS-0 (i.e., 64 kbps) channels. For example, 128 kbps is two DS-0 channels, 256 kbps consists of four DS-0 channels, and 512 kbps consists of eight DS-0 channels. When ordering FT1 service from a telco provider, you actually receive a full T1 channel but only pay for the number of DS-0 channels you order. (*Note:* Some U.S. carriers provide 256 kbps channels as part of their ISDN services, but these are being dropped in favor of 64 kbps channel services.)

A 64 kbps FT1 line is less efficient than a 64 kbps ISDN channel in terms of the amount of bandwidth available for data communication. This is because T1 control frames are in-band. Thus, you only get 56 kbps for data with every channel; the remaining 8 kbps are for control. ISDN, on the other hand, uses a separate channel for control and thereby provides a full 64 kbps for data. Similarly, a 128 kbps FT1 line only provides 112 kbps for data, but a 128 kbps ISDN channel provides a full 128 kbps. FT1 service is attractive to customers who do not require full T1 service but need more capacity than an ISDN 64/128 kbps line.

BOX 7.1: Nyquist's Theorem

In the early-to-mid 1920s, Henry Nyquist discovered that the maximum signaling rate of a noiseless channel was twice the number of samples. This discovery can be seen by observing the following standard sine wave carrier.



Observe that the natural form of a sine wave makes it possible for every half cycle to represent one signal state since the two half cycles are mirrored images of each other. This leads to the logical conclusion that the sampling rate must be twice the highest frequency because each cycle of the waveform corresponds to two values—one for the positive amplitude level and one for the negative amplitude level. Thus, if we have w cycles per second (i.e., Hertz), then we can have $2w$ signal states. Generalizing this concept, if a noiseless communications channel uses N values per signaling state, then the channel's maximum data transmission capacity in bits per second is given as

$$\text{Maximum Data Rate} = 2w\text{Log}_2 N \text{ bps}$$

where w = the number of cycles expressed in Hertz and
N = the number of discrete signaling states used

This relationship, known as *Nyquist's Theorem*, provides the maximum data rate of a noiseless communications channel. For example, if we have a 4000 Hz channel transmitting binary signals—and thus two signaling states—then the maximum channel capacity is:

$$\begin{aligned} C &= 2w\text{Log}_2 N \\ &= 2(4000)(\text{Log}_2 2) \\ &= 8000(1) \\ &= 8000 \text{ bps} \end{aligned}$$

Note that Nyquist's Theorem is the basis for pulse code modulation (PCM), which specifies that an analog signal be sampled at regular intervals and at twice the highest frequency on the line in order to get a sample that is an exact representation of the original signal. Nyquist's Theorem (and PCM) also has applications in CDs and digital audio tapes. For example, the upper limit of audio frequencies that can be reproduced digitally is a function of the sampling frequency. Thus, in order to capture a 20 kHz audio bandwidth, a sampling rate of 40 kHz is required. This is the sampling rate on which CD quality audio is based.

In addition to remembering that Nyquist's Theorem applies to noiseless channels, it is equally important to remember that the theorem also applies only to the signaling rate. Thus, if we increase the amount of data a signal carries via a modulation technique, then the data transmission rate also increases without violating the theorem. For a demonstration of Nyquist's Theorem see http://www.cs.brown.edu/people/dlg/gfxnotes/signal/nyquist.

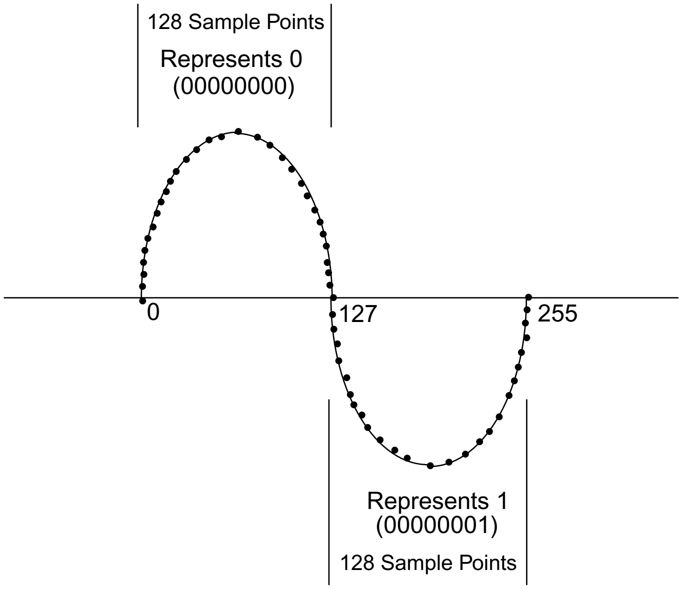


FIGURE 7.3 By using an 8-bit code to convert an analog signal into a digital signal via pulse-code modulation, an analog signal is partitioned into $2^8 = 256$ possible sample points: 128 points represents 0, and 128 points represent 1. Source: Adapted from Bates & Gregory, 1998.

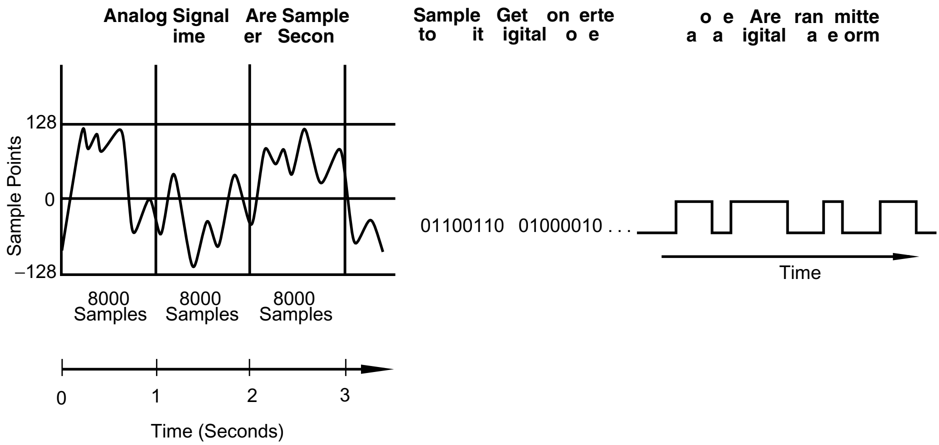


FIGURE 7.4 Example of the pulse-code modulation (PCM) process. Source: Adapted from Parkinson, 1988.

11. How much does T1 service cost and what kind of equipment is needed?

T1 service costs vary widely and are based on a customer's location relative to the telco provider and the type of circuit desired. Consideration also must be given to end equipment purchases and monthly service charges, which include the cost of the circuit itself and any *local loop* charges. The local loop is essentially the cable that connects the telephone central office (or exchange) with the customer's location. Local loop charges usually vary directly with the distance of the customer's location from the central office. For example, to interconnect PBXs, a *channelized* circuit is used, which requires no special equipment. (*Note:* A PBX stands for *Private Branch Exchange* and is a telephone exchange used within an organization to provide internal telephone extensions and access to the public telephone network—the modern day equivalent of what used to be called a *switchboard*.) The trade-off is that you cannot partition the T1 link. If, on the other hand, you need to partition the link, then a nonchannelized circuit is needed, requiring two T1 multiplexers—one at each end of the circuit.

A T1 circuit also requires special termination equipment called a *CSU/DSU*. A *Channel Service Unit* (CSU) performs many functions. It regenerates the signal, monitors the line for electrical anomalies, provides proper electrical termination, performs framing, and provides remote loopback testing for diagnosing line problems. Some CSUs also support the Internet's Simple Network Management Protocol (SNMP). A *Data Service Unit* (DSU), which is also referred to as *Digital Service Unit*, provides the interface (usually V.35, a type of serial interface) for connecting a remote bridge, router, or switch to a T1 circuit. The DSU also provides flow control between the network and the CSU. A CSU and DSU are usually combined to form a single unit—a CSU/DSU or DSU/CSU— which is sometimes described as the digital equivalent of a modem (i.e., a "digital modem") for a T1 line. Although the functions of a CSU/DSU and modem are similar, describing a CSU/DSU as a digital modem is misleading because a CSU/DSU does not perform any type of modulation or demodulation. Instead, a CSU/DSU works exclusively with digital signals; it provides an interface between a digital computing device and a digital transmission medium. Figure 7.5 shows a typical T1-based WAN link.

Although you didn't ask the question, an E-1 circuit is terminated using a Network Termination Unit (NTU), which provides broadly similar CSU/DSU functionality. Unlike the U.S. market, where the provision of a CSU/DSU is normally the responsibility of the end user, NTUs are always supplied by the telco in Europe. You should also note that line coding is different between T1 and E-1 circuits. In the United States, a technique called "bipolar with 8 zeros substitution" (B8ZS) is used; in Europe, a technique called "high density bipolar—3 zeros" (HDB-3) is used.

Prices for T1 multiplexers range from a few thousand dollars to as much as $100,000 or more depending on configuration (e.g., number of ports, bandwidth capacity). CSU/DSU units cost between $200 and $1000. Routers are priced from around $1000 to more than $50,000 depending on configuration needs. As we indicated earlier, circuit costs fluctuate. Factors that influence these costs include: geographical location; the telco provider (e.g., WorldCom, AT&T, British Telecom, BellSouth); whether a local loop fee is required, which varies depending on the distance a customer's site is located from the telco's nearest point of presence (POP); the service contract; whether or not you commit to a short-term (one year) or long term (two to five years) subscription; the manner in which the circuit is

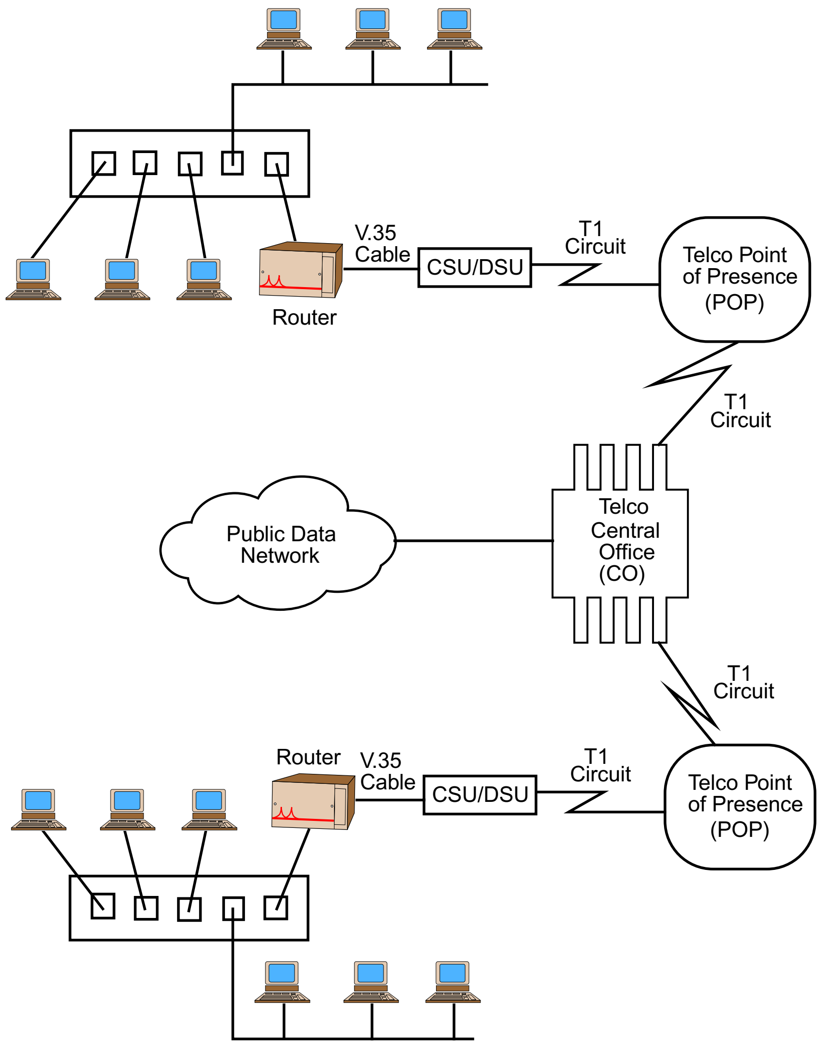


FIGURE 7.5 A typical configuration scheme of a leased T1 WAN connection between two sites involves a V.35 link between a router's V.35 port and a CSU/DSU. The CSU/DSU provides the interface to the T1 circuit. This circuit terminates at the telco's CO either directly or via a POP located near the customer's premises. The CO then provides connectivity to the network.

routed; whether you will be running a WAN technology like frame relay over the link or simply using it as a dedicated leased circuit; and many others. Given the many parameters on which circuit costs are based, we opt to not provide any estimates of these costs.

12. How do T1 and E-1 compare to ISDN's Primary Rate Interface?

ISDN PRI is an ANSI and an ITU-T standard. As an ANSI standard ISDN PRI signaling consists of twenty-three 64 kbps channels for voice or data—called *B* channels—and one 64 kbps control channel—called a *D* channel. This format, known as 23*B* + *D*, is based on the North American DS-1 service and provides 1.544 Mbps. Compared to T1 service, ISDN PRI provides more bandwidth for data because it uses a separate channel for control. Twenty-three channels at 64 kbps each yield an aggregate bandwidth of 1.472 Mbps. T1 service provides only 1.344 Mbps for data because its control is in-band. As an ITU-T standard, ISDN PRI supports a 30*B* + 2*D* format that is based on the European E-1 standard and provides 2.048 Mbps. (See Chapter 11 for more information about ISDN.)

13. I am constantly reading and hearing about SONET and OC rates like OC-3. Where does this terminology fit into all of this?

Excellent question. Before we answer it, though, let's review a little history. The development and deployment of NADH led telco carriers to develop their own methods of providing T-carrier service based on NADH. This acceptance of NADH established it as a de facto standard. There was one problem, though. The telcos' T-carrier services were incompatible, which made it difficult for customers to exchange data between different carriers. As a result, subscribers became locked into one solution and NADH eventually became a de facto proprietary standard. (See Chapter 1 for additional information about standards.) In addition to proprietary intercarrier circuits, NADH was also incompatible with the way the Europeans multiplexed their signals. In the early days of networking, when proprietary standards ruled and communications were usually restricted to within a country's own borders, these issues were not significant. In an era of open systems, standards, and global telecommunications, however, these issues are of paramount importance.

To address the difficulties inherent in intercarrier circuits globally, two transmission technology standards were developed: *Synchronous Optical Network (*SONET*)* and the *Synchronous Digital Hierarchy (*SDH*)*. SONET is an ANSI standard; SDH is an ITU-T standard. SDH was drafted after SONET and incorporates it. SONET and SDH (frequently written as "SONET/SDH") are international physical layer standards that provide a specification for high-speed digital transmission via optical fiber. At the source interface, signals are converted from electrical to optical form. They are then converted back to electrical form at the destination interface.

In the ANSI world, SONET's terminology includes Optical Carrier level (OC-*n*) and Synchronous Transport Signal level (STS-*n*). STS rates represent electrical signals and their optical equivalents are expressed as OC rates. In the ITU-T world, the official term used is Synchronous Transport Module (STM-*n*). Table 7.3 contains a summary of OC-*n*, STS-*n*, and STM-*n* line rates. (*Note:* The designation, OC-*nc* indicates that multiple smaller circuits are concatenated to form a circuit. For example, OC-3 denotes a single 155.52 Mbps circuit, but OC-3c denotes three OC-1 circuits are concatenated to provide this bandwidth.)

14. The OC rates listed in Table 7.3 are fast. Also, isn't OC-3 what ATM uses?

You are correct in noting that OC rates are fast. The recent advances in LAN technologies (e.g., 100 Mbps and Gigabit Ethernet) rendered WAN links as the bottleneck of an

TABLE 7.3 Comparison of SONET and DS Line Rates

OC-n	STS-n	STM-n	Data Transmission Rate (Mbps)	Number of DS-0 Channels	Number of DS-1 Channels	Number of DS-3 Channels
OC-1	STS-1	—	51.84	672	28	1
OC-3	STS-3	STM-1	155.52	2016	84	3
OC-9	STS-9	STM-3	466.56	6048	252	9
OC-12	STS-12	STM-4	622.08	8064	336	12
OC-18	STS-18	STM-6	933.12	12,096	504	18
OC-24	STS-24	STM-8	1,244.16	16,128	672	24
OC-36	STS-36	STM-12	1,866.24	24,192	1008	36
OC-48	STS-48	STM-16	2,488.32	32,256	1344	48
OC-96	STS-96	STM-32	4,976.64	64,512	2688	96
OC-192	STS-192	STM-64	9,953.28	129,024	5376	192

internetwork. Consider, for example, a WAN that interconnects two Gigabit Ethernet LANs by a T1 circuit. Although data transmission locally is occurring at a rate of 1000 Mbps, the transfer rate between LANs is 1.544 Mbps, nearly 650 times slower. SONET resolves this bottleneck. You also are correct in noting that OC-3 is the rate used for ATM networks. In fact, 155 Mbps (OC-3) ATM and 622 Mbps (OC-12) ATM were designed specifically to use SONET as their carrier service. Also, higher rate ATM can run only over SONET.

15. Can you tell me a little more about SONET and SDH?

Sure. As we indicated earlier, SONET was developed by the telcos to address the need for a fiber-optic based standard for broadband transmissions within the telecommunications industry. Its roots are from synchronous transfer mode (STM), which is used in U.S. digital telephone networks. The basic building block of the SONET signal hierarchy is STS-1 (51.84 Mbps). This line rate is derived from the STS-1 frame, which consists of 810 eight-bit bytes transmitted at 8000 Hz. Multiplying 810 bytes per frame, at 8 bits per byte, by 8000 Hz yields 518,400,000 bps or 51.84 Mbps.

As a fiber-based medium, SONET offers several advantages over the copper-based T1 hierarchy. First, hundreds of thousands of simultaneous voice and data transmissions are possible using fiber. This is not feasible with copper cable. Second, fiber is immune to EMI (see Chapter 4). Third, fiber is available in either single or multimode and thus can be used for LAN connections or as the backbone of a WAN. As a synchronous transmission facility, SONET again offers several advantages over its asynchronous T1 counterpart. Bandwidth can be allocated on an as-needed basis and routes can be dynamically reconfigured. As a carrier service, SONET can serve as the transport facility for any type network technology or service, including ATM, FDDI, SMDS, and ISDN. Finally, SONET can support various topologies including point-to-point, star, and ring.

SDH has its roots in an early transport mechanism called *plesiochronous digital hierarchy (PDH)*. (*Note:* The word "plesiochronous" means "partially synchronized.") PDH is similar to STM in that both use time-division multiplexing. The difference is STM is applied to T-carrier circuits (e.g., T1 and T3), and PDH is applied to E-carrier circuits (e.g., E-1 and E-3). Aside from some minor differences, SDH is essentially the same as SONET, and at OC-3 rates and higher, the two are virtually identical.

16. Besides the physical layer, what is involved in transmitting data across a WAN?

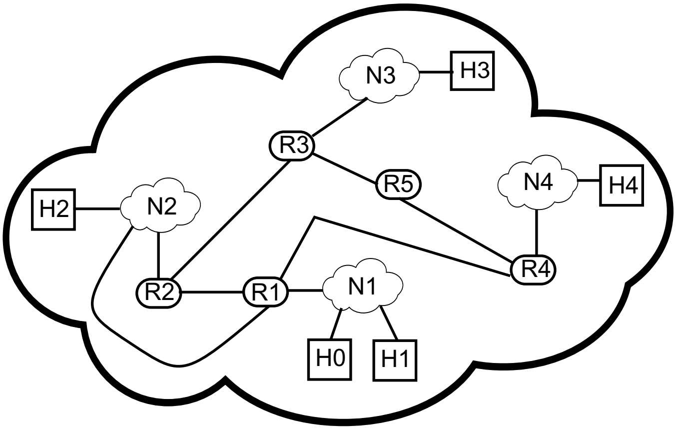
The single, most important issue related to data transmission across WAN links is *routing*, which involves directing data packets from source to destination. Routing is a network layer function (layer 3 of the OSI model). This layer also provides services to the transport layer (layer 4) and performs congestion control.

17. Can you give me an example of how the network layer works?

Sure. We will use the TCP/IP model instead of the OSI model. (See Chapter 2 for a comparison between the TCP/IP and OSI models, and Chapter 3 for additional information about the TCP/IP protocol suite.)

The TCP/IP layer that functions similar to OSI's network layer is the Internet layer which also is called the network layer. The heart and soul of this layer is the Internet Protocol (IP)—the IP of TCP/IP. IP is a connectionless datagram service and is responsible for routing packets between nodes. (A *datagram* is an IP network layer packet. See Chapter 3 for additional information.) In short, IP receives data bits from the upper layer, assembles the bits into packets (i.e., IP datagrams), and then selects the "best" route based on some criterion, called a "metric" (e.g., distance, number of router "hops," bandwidth) the packets should take to reach their destination. Since IP is connectionless, every packet must contain the address of the destination node. This address, called an Internet or IP address, is assigned to an Internet node by a network administrator or by an automated protocol (such as Microsoft's DHCP) as part of the node's initial network configuration. An IP address uniquely identifies a host similar to the way a street address uniquely identifies a residence, or the way a social security number, driver's license number, or student ID number uniquely identifies a person. It is used by the network layer as a road map to locate a host within the Internet by determining what path a packet is to follow en route to its final destination.

Packets destined for a host connected to the same LAN as the sending host are generally delivered directly by the sending host. To transfer packets destined for a host connected to a remote network, however, dedicated routers are usually used. Routers are also referred to as *gateways*. For example, consider Figure 7.6, which contains four interconnected networks (N1 through N4), five hosts (H0 through H4), and five routers (R1 through R5). If H0 sends a packet to H1, no special router is needed to route the packet. H0 effectively serves as its own gateway for locally destined packets. However, if H0 sends a packet to H2, then at least one router is involved in the transfer. The packet could go through any of the following router paths: R1 only; R1-R2; or R1-R4-R5-R3-R2. Although routers are used to deliver packets from one network to another, IP does not guarantee that a packet will indeed be delivered to its destination. If an intermediate router,



IGURE 7.6 A network and associated subnetworks (subnets) are typically pictured as clouds. hown in this figure are four interconnected subnets (N1 through N4), five hosts (H0 through H4), nd five routers (R1 through R5). As an example of routing, note that data packets originating on H1 nd destined for H2 can take several paths through the network. One path is through R1 only. A econd path is R1-R2. Still a third path is via R1-R4-R5-R3-R2. If the network layer is IP-based, hich provides connectionless datagram service, all of these routes are possible.

for example, contains incorrect or stale routing information, packets might get lost. IP does not take any action to retransmit undelivered packets. This is done by higher-level protocols, specifically TCP. Additionally, IP fragments and reassembles packets when necessary so they do not exceed the maximum packet length (called the *maximum transmission unit*, or MTU) a physical network is capable of supporting. If a packet's size is greater than a network's MTU, the packet is broken into smaller units (called *fragmenting*) and sent to the destination in the form of several separate packets. The complete packet is then reassembled at the destination node before it is passed to the higher levels. Reassembly of an IP datagram can only occur at the destination node and not at any of the intermediary nodes the packet traverses. This is because IP is a connectionless datagram service—datagrams can take different paths en route to their destination and hence an intermediary node might not receive all of the fragmented datagrams.

When too many packets are present in a subnet, performance degrades. This situation is called *congestion*. Congestion occurs when routers are too slow, causing queues to lengthen. Additionally, if routers are too fast, queues will build up whenever input traffic is greater than the capacity of output lines (e.g., three input lines delivering packets at top speed and all need to go out on the same line). The ultimate level of congestion is known as *deadlock*, a concept with which you might be familiar if you have studied multitasking operating systems. Deadlock occurs in this context when one router cannot proceed until a second router does something, and the second router cannot proceed because it is waiting for the first router to do something. The network layer is responsible for providing congestion control, which deals with making certain that a subnet can carry the offered traffic. It is global in scope and involves all hosts, routers, and other factors. Note that congestion control is not the same as flow control, which applies to point-to-point traffic between a sender node and a receiver node.

18. How does a router determine a particular path?

The path a packet takes through a network from source to destination is a function of routing protocols. Examples include RIP (Routing Information Protocol), RIP version 2 (RIP-2), OSPF (Open Shortest Path First), and IS-IS (Intermediate System to Intermediate System). The first three are part of the TCP/IP suite and can only route IP packets. IS-IS can route both IP and OSI Connectionless Network Layer Protocol (CNLP) packets.

Routing protocols are a function of network protocols. For example, if your network protocol is TCP/IP, then several routing protocol options are available including RIP, RIP-2, and OSPF. If your network protocol is OSI's CNLP, then your routing protocol is IS-IS. Most network protocols, however, can be encapsulated in TCP/IP and routed using TCP/IP-based routing protocols. *Protocol encapsulation,* also called *tunneling*, "wraps" packets from one network protocol in a packet for another protocol. This wrapped packet can then be routed through a network using a routing protocol that is supported by the "wrapper" network protocol. AppleTalk packets, for instance, can be routed through the Internet by wrapping them into TCP/IP packets. Tunneling also is used for transporting nonroutable protocols across a WAN. For example, the former Digital Equipment Corporation's LAT and Microsoft's NetBEUI protocols do not provide network layer service and hence cannot be routed. These protocols can be sent across a WAN, though, by encapsulating them within a routable protocol such as IP. IP tunneling is commonly used within the Internet especially for virtual private networks (VPNs), which are discussed later in the chapter, as well as for secure Web connections. Tunneling effectively removes network protocol restrictions inherent in a particular network. The concept of tunneling is demonstrated in Figure 7.7.

19. What's the difference between a *router* protocol and a *routing* protocol?

The term "router protocol" formally specifies three different types of router-related protocols—those that provide a service, those that greet neighbors, and those that do routing. For example, since IP provides a service to the transport layer, it is considered a network layer *service* protocol for TCP/IP. *Neighbor-greeting protocols* are those that enable nodes and routers to find each other so they know which nodes and routers are accessible. These protocols also provide address-translation capabilities. One example is the Address Resolution Protocol (ARP), which is part of the TCP/IP suite (Chapter 3). Finally, examples of routing protocols include RIP and OSPF.

20. While we are on the subject of terminology, what do IGP and BGP mean?

IGP stands for Interior Gateway Protocol. To understand IGP we need to first introduce the concept of an *autonomous system* (AS). Adjunct to the various routing protocols like RIP and OSPF is something known as *protocol areas*, which are also called *routing domains*. In the Internet, these routing domains are referred to as autonomous systems. An

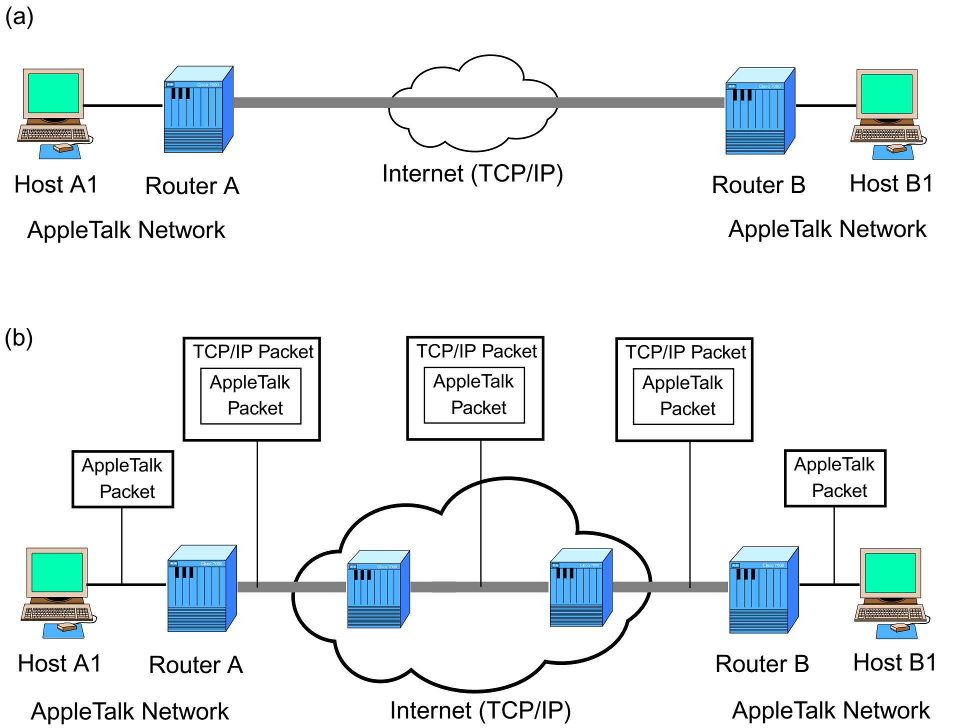


FIGURE 7.7 In the context of protocol encapsulation, tunneling involves "wrapping" packets constructed in one protocol format within some other protocol. A logical view of tunneling is shown in (a), where the thick line represents the tunnel to be established between routers A and B. In (b), a physical view of tunneling is shown. Here, host A1 transmits a data packet formatted in AppleTalk to host B1 across the Internet. The packet is first sent to router A, which wraps the packet into a TCP/IP-based packet. That is, router A takes the entire AppleTalk packet and places it into the data field of an IP datagram (see Figures 3.15 and 3.22). This IP datagram is assigned router B's address as the packet's destination address. This newly formed packet is then transmitted across the tunnel from router to router as a full IP packet. When router B receives the packet, it deencapsulates it (i.e., it removes the IP wrapper) and then transmits the preserved AppleTalk packet to host B1. In addition to protocol encapsulations, tunneling is commonly used for secure transmissions across the public Internet by wrapping an encrypted data packet in an IP packet.

AS is a collection of networks controlled by a single administrative authority. The networks within an AS also share a common routing strategy. That is, the routers connecting the networks within an AS trust each other and exchange routing information using a mutually agreed upon routing protocol. The network and associated subnets shown in Figure 7.6 can be regarded as an AS if all the routers employ the same protocols.

An IGP is an Internet protocol used to exchange routing information within an AS. Examples of these *intra*domain protocols include RIP, RIP-2, OSPF, IGRP, and Enhanced IGRP. (IGRP and Enhanced IGRP are proprietary Cisco Systems' Interior Gateway Routing

Protocols.) Each AS must also support a router that can exchange routing information with other autonomous systems. Routing protocols used for this purpose are known as exterior gateway protocols. Examples of these *inter*domain protocols include EGP, the Exterior Gateway Protocol, defined in RFC 904, and BGP, the Border Gateway Protocol, defined in RFC 1105 and RFC 1771. EGP and BGP are part of the TCP/IP protocol suite. Of the two, BGP has evolved into a robust Internet routing protocol and the term "Border Gateway Protocol" is used in favor of the term "Exterior Gateway protocol." BGP enables a network to make optimal IP routing decisions across multiple administrative domains (i.e., ASs).

AS, IGP, and BGP are fundamental to the way in which the Internet is designed. As a global internetwork, the Internet is partitioned into autonomous systems, which enable different areas of the Internet to be administered separately from one another. Within an AS, routers run the same interior gateway protocol. By keeping an AS administratively separate, different autonomous systems can also run different IGPs within their respective areas. For example, one AS might run RIP, another might run OSPF, and a third AS might support IGRP. These separate autonomous systems can then be interconnected via routers that run a border gateway protocol. Thus, routers within an AS communicate via an IGP, and "border" routers—those between autonomous systems—communicate via a BGP. This concept is expanded later in the chapter during our discussion of OSPF.

21. Let's get back to routing protocols. What functions do they perform?

Routing protocols perform two primary functions. First, they determine the "best" path a packet should take when it travels through a network from source to destination, and second, they maintain routing tables that contain information about the network's topology. In the Internet world, IP is used to transport packets through the Internet based on the information contained in routing tables.

22. Could you explain how these functions work?

Certainly. Network routing protocols rely on routing algorithms (discussed later in the chapter) to calculate the least-cost path from source to destination. A routing algorithm is that part of the network layer software responsible for deciding on which output line an incoming packet should be placed. If the subnet is packet-switched, then this decision is made for every incoming packet. If the subnet is circuit-switched, then routing decisions are only made when the virtual circuit is being set up.

Routing algorithms use a "least-cost metric" to determine the best path. For example, RIP uses a "hop-count" metric, which is the number of router-to-router connections a packet passes through en route to its destination. OSPF's metric is bandwidth. IS-IS supports user-assigned metrics. IGRP and EIGRP metrics include propagation delay, bandwidth, packet size, and load. Other metrics might include time, channel utilization, and esoteric measures such as error rates. As an example, consider the network and associated subnets shown in Figure 7.8. The number of hops between H1 and H2 is two if the packet travels via the path R1-R2. Similarly, if the packet takes the path R1-R3-R2 or R1-R4-R5-R3-R2, then the number of hops is three and five, respectively. Given these various paths, the best or least-cost path is R1-R2 since its hop count is the smallest. Hop-count metrics ignore line speeds or delays. Thus, in Figure 7.8, packets will always take the path R1-R2

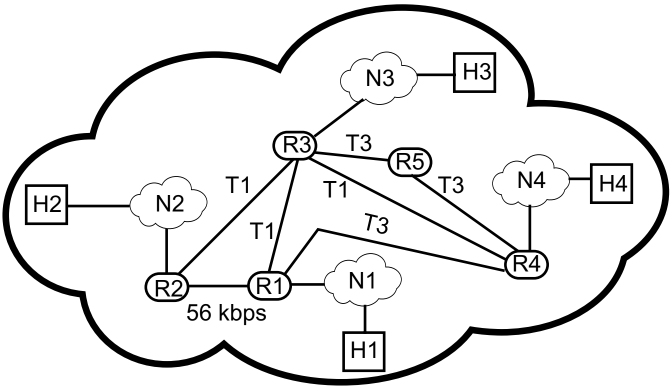


FIGURE 7.8 The determination of the "best" path a packet should take is based on metrics. One metric is number of hops. Another is bandwidth. If the best path is determined by number of hops, then the route a packet takes from H1 to H2 is through routers R1-R2 because this represents the least number of hops: two. On the other hand, if a bandwidth metric is used, then the best path is either R1-R3-R2 or R1-R4-R5-R3-R2.

```
traceroute -s www.fit.edu www.ucf.edu
traceroute to www.ucf.edu (132.170.240.131) from www.fit.edu, 30 hops max, 40 byte
packets
1 bsport.fit.edu (163.118.2.1)  4 ms  3 ms  2 ms
2 172.17.16.53 (172.17.16.53)  8 ms  8 ms  8 ms
3 172.17.16.30 (172.17.16.30)  86 ms  77 ms  61 ms
4 campusgw3.cc.ucf.edu (132.170.60.1)  67 ms  33 ms  16 ms
5 132.170.12.2 (132.170.12.2)  60 ms  15 ms  22 ms
6 www.ucf.edu (132.170.240.131)  13 ms * *
```

FIGURE 7.9 The UNIX traceroute program traces the route an IP packet follows from a source node to a destination node. In this figure, we show the path of a packet from www.fit.edu to www.ucf.edu. The number of router hops is five (the last entry is the destination node, not a router). Each line shows the logical name of the router, its IP address, and the round-trip time in milliseconds of three separate 40-byte packets (called probes) sent between intermediate routers.

(assuming the links are "up") even though it might be a "slower" path than either R1-R3-R2 or R1-R4-R5-R3-R2. A UNIX program that displays the path a packet traverses is called *traceroute* (the Windows NT equivalent is called *tracert*.) The output of a trace made between two web servers—from www.fit.edu to www.ucf.edu—is shown in Figure 7.9. The numbers within parentheses are the Internet addresses of the nodes, and the numbers at the end represent the round trip time (in milliseconds) it takes a packet to reach a gateway. Three separate probes are sent between each intermediate node. Note that the source node is five hops away from the destination node. (The last address is that of the destination host,

TABLE 7.4 Sample Routing Table Information

Destination	Gateway	Flags[a]	Ref[b]	Use[c]	Interface
localhost	localhost	UH	0	33106	lo0
215.103.16.227	187.96.25.13	UGHD	29	102	le0
215.103.16.141	187.96.25.35	UGHD	116	16128	le1
default	187.96.25.1	UG	0	2888304	
187.96.25.0	187.96.25.2	U	210	29024	le0

[a] U = Route is up and operational; G = Packet must pass through at least one router; H = Route is to a specific host and not a network; D = Route was created dynamically
[b] Current number of routes that share the same link layer address
[c] Number of packets sent using this route

not a router.) As packets are transmitted from one router to the next, each router independently examines the contents of each packet's header and runs the specified routing algorithm. Based on its analysis of the header and the results of the algorithm, each router then determines the next hop for the packet.

Routers also maintain routing tables that contain, among others, the destination address of a node or network, known router addresses, and the network interface associated with a particular router address. Routers exchange routing table information with neighbor routers periodically. The type of information exchanged and the frequency of routing table updates are a function of the routing protocol used. When a router receives a packet it looks at the packet's destination address to identify the destination network, searches its routing table for an entry corresponding to this destination, and then forwards the packet to the next router via the appropriate interface. For example, Table 7.4 shows sample routing table information generated on a UNIX system with IP address 187.96.25.2. The command used to generate this table is *netstat -r*. Similar output can be generated on a Windows NT system using the command *route print*. First note the table's last entry, which illustrates "local-host routing." This entry indicates that any packets destined for the local network (187.96.25.0) will be forwarded via gateway 187.96.25.2, which is the IP address of the host. In this context, the local host acts as a simple router. Now look at the second entry of the table. This entry indicates that all packets with destination address 215.103.16.227 are forwarded to the router whose address is 187.96.25.13 via interface *le0*. This router in turn will have information on where to forward the packet so that it will ultimately reach its destination. Similarly, the third entry indicates that packets with destination address 215.103.16.141 are to be forwarded to the gateway whose address is 187.96.25.35, which is accessible via interface *le1*. Finally, the second to last entry of the table references a *default route*, which is a special route that contains the address of a default router. When a router receives a packet that contains an unknown destination address (i.e., there is no entry for the address in the routing table), the router forwards the packet to the default router. As a result, if the host system receives a packet with the destination address 212.133.65.3, the host will forward the packet to the router whose destination address is 187.96.25.1 since there is no entry in the host's routing table (Table 7.4) for 212.133.65.3.

23. I have heard the term "static route" used. What does it mean?

A *static route* is a fixed route that is entered into a router's routing table either manually or via a software configuration program. The selection of the route is determined by a network manager. Although static routes can be beneficial in some instances, they cannot be changed dynamically to compensate for changes in a network's topology.

24. What types of routing algorithms are there and how do they work?

Two general algorithms are available for computing metric information: *distance-vector* and *link-state*. The goal of both types of algorithms is to route a packet from one point in the network to another point in the network through some set of intermediate routers without "looping," a situation in which a packet is forwarded across the same link several times. The primary difference between distance-vector and link-state algorithms is the manner in which they collect and propagate routing information throughout the network. Let's examine these two algorithms separately.

Distance-Vector Algorithms A *distance-vector routing algorithm* determines the distance (hence the name) between source and destination nodes by calculating the number of router hops a packet traverses en route from the source network to the destination network. An example of a distance-vector algorithm is the Bellman-Ford algorithm, which is described in Box 7.2. Two distance-vector-based routing protocols are RIP and RIP-2, which exchange routing tables with their neighbors every 30 seconds. RIP and RIP-2 also support a maximum of 15 hops. Thus, if the number of router-to-router hops between source and destination nodes is greater than 15, then the network to which the destination node is connected is considered "unreachable." This limitation restricts the size of an internetwork to 15 consecutively connected networks.

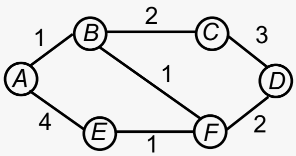
Link-State Algorithms In a *link-state routing algorithm* every router of a network does not send every other router its routing table. Instead, routers send each other information about the links they have established to other routers. This information is sent via a link-state advertisement (LSA), which contains the names and various cost metrics of a router's neighbors. LSAs are flooded throughout an entire router's domain. An example of how this is done is described later in our discussion of OSPF. Routers also store the most recent LSA they receive, and destination routes are calculated using LSA information. Thus, rather than storing actual paths, which is the case with distance-vector algorithms, link-state algorithms store the information needed to generate such paths. An example of a link-state algorithm is Dijkstra's shortest path algorithm, which iterates on length of path to determine a shortest route. Link-state-based routing protocols include OSPF, OSI's IS-IS, and Netware's Link Services Protocol (NLSP). Box 7.3 illustrates Dijkstra's shortest path algorithm.

25. Please tell me more about RIP.

OK. The Routing Information Protocol Version 1 was derived from the Xerox Network System's (XNS) routing protocol, which was also called RIP. RIP was bundled with BSD UNIX in 1982 as part of the TCP/IP protocol suite and became the de facto standard for IP routing. As mentioned earlier, RIP uses a distance-vector algorithm that determines

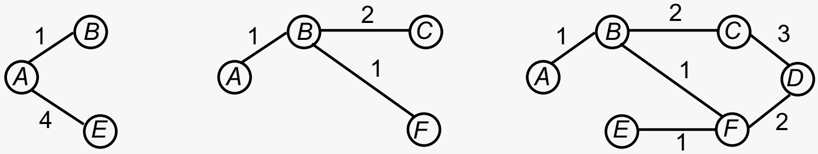
BOX 7.2: Bellman-Ford Algorithm

The Bellman-Ford routing algorithm is distance-vector-based and iterates on the number of hops a source node is from a destination node. To illustrate this algorithm, consider the following undirected graph, which depicts a sample network. The vertices *A, B, C, D, E,* and *F* may be thought of as routers, and the edges connecting the vertices are communication links. Edge labels represent an arbitrary cost. Our goal is to find the shortest path from *A* to *D* using the number of hops as the basis for our path selection.



We examine the costs of all paths leading from *A* to each node on a hop-by-hop basis.

ne op | **o op** | **ree op**



Path *AB* = 1
Path *AE* = 4

Choose path *AB*

Path *ABC* = 3
Path *ABF* = 2

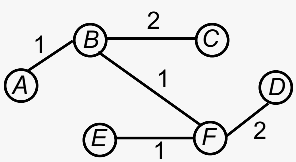
Choose path *ABF*

Path *ABCD* = 6
Path *ABFD* = 4
Path *ABFE* = 3

Choose path *ABFD*
Choose path *ABFE*

In the last step (three hops), two paths are selected. The first path, *ABFD,* represents the least-cost path from *A* to *D* based on the hops metric. The second path, *ABFE,* is selected because it represents the least-cost path from *A* to *E.*

The final result of the Bellman-Ford algorithm yields a tree that represents the least-cost incurred from the source node to every node of the network. Similar trees can be generated for every node of the network. Node *A*'s least-cost tree for our example is as follows:



From Node *A*:

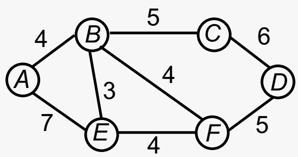
- the least-cost path to *B* is *AB* = 1
- the least-cost path to *C* is *ABC* = 3
- the least-cost path to *D* is *ABFD* = 4
- the least-cost path to *E* is *ABFE* = 3
- the least-cost path to *F* is *ABF* = 2

BOX 7.3: Dijkstra's Shortest Path First (SPF) Algorithm

Dijkstra's SPF routing algorithm is link-state-based and iterates on the distance metric. The algorithm uses a "closest nodes" concept and is based on the following principle:

> *Given a source node,* n*, the shortest path from* n *to the next closest note,* s*, either (a) is a path that directly connects* n *to* s *or (b) includes a path containing* n *and any of the previously found intermediate closest nodes plus a direct link from the last intermediate closest node of this path to* s.

To illustrate this algorithm, consider the following undirected graph, which depicts a sample network. The vertices *A, B, C, D, E,* and *F* may be thought of as routers, and the edges connecting the vertices are communication links. Edge labels represent an arbitrary cost metric. Our goal is to find the shortest path from *A* to *D* based on distance.



To implement this algorithm, it is helpful to maintain a running record of the successive closest nodes to the source node. We will let *k* represent the *n*th closest node. Thus, node *A* corresponds to *k* = 0. That is, the zero closest node to *A* is itself. This is the initialization step of the algorithm. We now begin our search for the successive closest nodes to *A*.

<u>First Closest Node</u> (*k* = 1)

The first closest node to *A* is either *B* or *E* since they are both directly connected to *A*. Since the *AB* path has a smaller cost, we select it. Thus, *B* is the first closest node to *A*.

k	Node	Path
0	*A*	—
1	*B*	*AB*

<u>Second Closest Node</u> (*k* = 2)

The second closest node to *A* must either be (a) a direct link from *A* or (b) via a path that includes the first closest node. The possible paths and related costs are: *ABC* = 9, *ABF* = 8, *ABE* = 7, or *AE* = 7. There are two shortest paths: *ABE* and *AE*. Thus, *E* is the second closest node to *A*.

k	Node	Path
0	*A*	—
1	*B*	*AB*
2	*E*	*ABE*
		AE

<u>Third Closest Node</u> (*k* = 3)

The third closest node to *A* must be via a path that includes nodes *B* or *E*. (There are no more direct links to *A*.) The possible paths and related costs are: *ABC* = 9, *ABF* = 8, *ABEF* = 11, or *AEF* = 11. The shortest path is *ABF*. Thus, *F* is the third closest node to *A*.

k	Node	Path
0	*A*	—
1	*B*	*AB*
2	*E*	*ABE*
		AE
3	*F*	*AEF*

<u>Fourth Closest Node</u> (*k* = 4)

The fourth closest node to *A* is via a path that includes nodes *B, E,* or *F*. The possible paths and related costs are: *ABC* = 9 or *ABFD* = 13. The shortest path is *ABC*. Thus, *C* is the fourth closest node to *A*. Note that neither *ABEF* nor *AEF* is considered at this stage of the algorithm because *F* was previously found to be the third closest node.

k	Node	Path
0	*A*	—
1	*B*	*AB*
2	*E*	*ABE*
		AE
3	*F*	*AEF*
4	*C*	*ABC*

(continued)

BOX 7.3: Dijkstra's Shortest Path First (SPF) Algorithm (continued)

Fifth Closest Node (k = 5)

The fifth closest node to *A* is via a path that includes nodes *B, E, F,* or *C.* The possible paths and related costs are: *ABCD* = 15, *ABFD* = 13, *ABEFD* = 16, and *AEFD* = 16. The shortest path is *ABFD.* Thus, *D* is the fifth closest node to *A*.

k	Node	Path
0	*A*	—
1	*B*	*AB*
2	*E*	*ABE*
		AE
3	*F*	*AEF*
4	*C*	*ABC*
5	*D*	*ABFD*

Since *D* is the destination node, the shortest path from *A* to *D* is *ABFD.*

the best route by using a hops metric. RFC 1058 contains a good description of this algorithm. When used in small homogeneous networks, RIP is a very efficient protocol and its operation is fairly simple. RIP keeps all routing tables within a network updated by transmitting routing table update messages every 30 seconds. After a RIP-enabled device receives an update, it compares its current information with the information contained in the update message. Current routing table entries are replaced with updated information when the update message contains any of the following entries: (a) a route and corresponding metric to a previously unknown destination (this information is added to the existing table), (b) a new route to an existing destination with a smaller metric (the old route is replaced by the new route), and (c) a new metric for the same route to an existing destination (the old metric is replaced by the new one).

RIP uses timers to handle link or neighbor router failures. If a router does not hear from any of its neighbors within 180 seconds, the protocol assumes the node or link is dead. Once this determination is made, the router sends out a special message to its responding neighbors about this failure. Routing table entries that include routes via the dead link or node are then altered accordingly. RIP also imposes a 15-hop maximum—if a destination network is more than 15 hops away, RIP classifies the network "unreachable." In router jargon, the destination network's cost goes to infinity. Thus, the network becomes too expensive to reach and hence is unavailable.

Routers implementing RIP occasionally misinterpret old routing information as new, which can then cause routing loops. To resolve this situation, RIP employs several strategies. These include *split-horizon, split-horizon with poisoned reverse,* and *hold-down*.

Split-Horizon The split-horizon strategy ensures that a router never sends routing information back in the direction from which it came. For example, let's assume router *A* receives a routing table update from router *B*. Once *A* receives this information, *A* updates its routing table to reflect the routes listed in *B*'s routing table. Now, when *A* is ready to send its own routing table update to *B*, split-horizon prevents *A* from sending any updates back to *B* that were made based on the information *B* sent to *A*. Generalizing, with split-horizon, routing information provided by a neighbor is eliminated in any updates sent back to that neighbor.

Split-Horizon with Poisoned Reverse This strategy is similar to split-horizon with one exception: Routing information provided by a neighbor is included in any updates sent back to that neighbor. Such routes, however, are assigned a cost factor (i.e., metric) of infinity. This means that a network is unreachable. For example, consider the network shown in Figure 7.8. Note that R2 and R3 are neighbor routers, and each can claim a route to R4 through each other (e.g., R2-R3-R4, R2-R3-R5-R4, R2-R1-R3-R4, and R3-R2-R1-R4). Assume R2 receives a routing table update from R3 that includes the R3-R2-R1-R4 path. With poisoned reverse, R2's update to R3 indicates that R4 is unreachable (i.e., its metric is infinity). This prevents R3 from claiming that a path to R4 exists through R2. Thus, any path to R4 from R3 must either be a directly connected link (R3-R4), or through other routers (e.g., R3-R5-R4 or R3-R1-R4). The poisoned reverse update is also very effective in eliminating routing loops because when two routers have routes pointing at each other (as was the case in our illustration), the update will immediately make that link unreachable.

Hold-Down The hold-down strategy requires routers not to update their routing tables with any new information they receive for a prescribed period of time called the *hold-down time*. To illustrate this, let's assume that a link within a network fails. Let's further assume that two routers, *A* and *B*, exchange routing table updates and that *B* has been informed of the link failure from one of its other neighbors but *A* has not. If *A* does not receive information about the failed link before it sends *B* a routing table update, *B* will receive *A*'s update, see that the "failed" link is active, and incorrectly reinstate it. With hold-down implemented, though, *A* will receive information about the failed link before it sends an update to *B*. Thus, hold-down enables routing table updates to be propagated to all routers in a timely manner, thereby ensuring that new routes are indeed new and not old ones. By making the hold-down time longer than it takes to "count to infinity," routing loops can be avoided. Hold-down is not standardized and hence should be considered implementation specific.

The format of a RIP packet is shown in Figure 7.10(a). The command field identifies the packet as either a request (a router asks its neighbor to send its routing table) or a response (a routing table update). The version number field contains the version of RIP being used. The AFI field identifies the protocol family of the address contained in the address field. Thus, if the address is an Internet address, the AFI field is coded to represent IP. RIP updates can be up to 520 bytes long and hence multiple entries, from the AFI, can be placed in the datagram. The address field is the destination address of the network being advertised. The metric field contains the hop count to the address listed in the address field. Using only half of its 24 bytes, RIP carries the least amount of information necessary for routers to route messages through a network. Thus, overhead is minimal. RIP is defined in RFC 1058.

26. What's the difference between RIP and RIP-2?

As originally designed, RIP does not support the concepts of autonomous systems, subnetting, or authentication. RIP also cannot interpret BGP or EGP routes. To address some of these issues, several extensions to the original protocol were incorporated to extend its usefulness. This protocol, RIP Version 2, which is shown in Figure 7.10(b), maintains RIP's

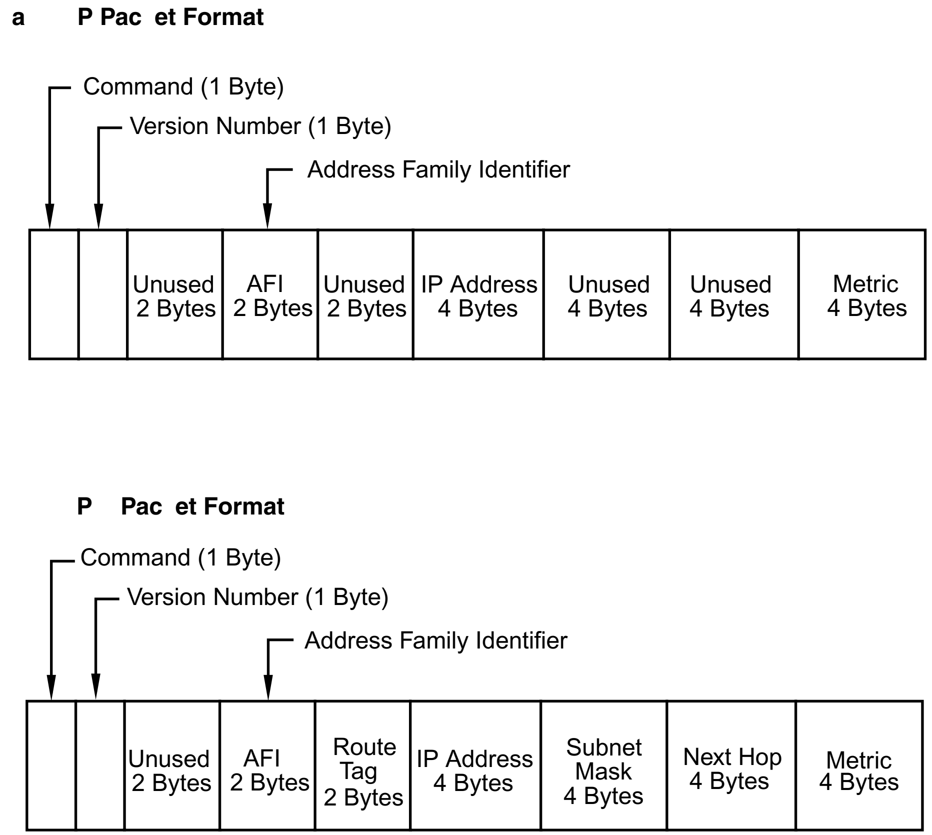


FIGURE 7.10 RIP Version 1 packets (a) contain several unused fields. These fields are defined in RIP-2 (b), thereby extending the usefulness of the RIP protocol. RIP-2 is completely compatible with RIP. The command, version number, AFI, and metric fields have exactly the same meaning in both versions. The version number field for RIP-2 will have a 2 for any RIP message that carries information in any of the unused fields from Version 1. The content of the 2-byte unused field in both versions is ignored. (*Note:* RIP updates can be up to 520 bytes long. Thus, multiple entries, from the AFI, can be placed in the datagram.) Source: Adapted from RFC 1058 and RFC 1723.

command, version number, AFI, IP address, and metric fields. RIP-2 messages that carry information in any of the unused fields from Version 1 will have a 2 in the version number field. The content of the two-byte unused field in both versions is ignored. Thus, a RIP-2 packet does not alter the contents of RIP.

The new features of RIP-2 include the following:

- **Authentication.** If the AFI field of the first entry in a message is FFFF, then this signifies that the message is an authentication packet. Presently, RIP-2 only supports simple passwords for authentication.

- **Interpretation of IGP and BGP Routes.** The route tag field enables RIP to distinguish between intradomain RIP routes and interdomain RIP routes, which are usually imported from an exterior gateway protocol such as BGP or another interior gateway protocol such as OSPF or IGRP.
- **Subnet Masks.** The original version of RIP assumed that all networked devices used the same subnet mask and hence did not carry subnet mask information. RIP-2 removes this assumption by supporting multiple-length subnet masks. This is an extremely important feature because today's network addresses are partitioned into subnets, and routers need a subnet mask to determine routes. The subnet mask field contains the subnet mask, which is applied to the IP address to yield the network address. If this field is 0, then no subnet mask is included for a particular entry. (See Box 3.2 for an example of subnet masks.)
- **Next Hop Field.** This field contains the IP address of the immediate next hop to which packets are to be forwarded based on the packet's destination address. This field eliminates packets being routed through extra hops in a system.
- **Multicasting.** Another advantage is that, instead of broadcasting updates, RIP-2 supports multicasting. This reduces the load on hosts that are not listening to RIP-2 packets.

RIP-2 is specified in RFC 1723. Another version of RIP, *RIPng*, was also developed (see RFC 2080) to support an IPv6-based network.

27. Given its improvements, is RIP-2 now the de facto Internet routing protocol?

Not really. Although RIP-2 improves RIP's capabilities, it still has several deficiencies. The biggest problem with RIP is that it was never designed for large heterogeneous networks. As a network grows, destinations that are more than 15 hops away are classified unreachable in RIP. Furthermore, RIP's hops routing metric is not always the most efficient one to use. Unfortunately, the hops metric cannot be changed to any other metric. Finally, RIP is not as resistant to network changes (e.g., handling link failures), and it requires a greater amount of bandwidth for routing updates than other routing protocols because the entire routing table is sent. Many organizations still use RIP for server-based routing, which enables servers to make dynamic routing decisions with little additional CPU or network overhead. However, the overall internetwork infrastructure today relies on OSPF.

28. Tell me about OSPF. How different is it from RIP?

As noted earlier, the Open Shortest Path First protocol is an interior gateway protocol based on a link-state algorithm; RIP is distance-vector-based. Many of RIP's limitations are resolved with OSPF. For example, OSPF is specifically designed for large heterogeneous IP networks. OSPF supports a 16-bit routing metric, which enables network managers to design least-cost routing schemes based on traffic load, propagation delays, line speed, and bandwidth, instead of relying solely on hops. Routing updates with OSPF are also very efficient and can be authenticated via passwords, digital signatures, and the like. As a link-state-based protocol, OSPF updates routes only when the status of a link changes. Furthermore, OSPF does not broadcast entire routing tables to update neighbor routers. Instead, small link-state packets called link-state advertisements containing specific information about a router's network links are transmitted. OSPF also employs the concept of *areas*. Thus, updates are not bandwidth intensive because, except for area summary updates, they only occur within a prescribed area. "Area routing" also insulates intradomain routing from external routing problems. Other features of OSPF include quick recovery after changes are made in a network's topology, resistance to routing loops, and the capability to interpret and redistribute EGP and IGP routes independently.

There are currently two specifications for OSPF. The first, *OSPF version 2*, is for IPv4 and specified in RFC 2328. The second, *OSPF version 3*, is for IPv6 (see Chapter 3) and is specified in RFC 2740. The contents of each version's packet header are shown in Figure 7.11. Note that there are two primary differences between the two specifications. The first is header size: OSPF for IPv4 specifies a 24-byte header, whereas OSPF for IPv6 specifies a 16-byte header. The second difference is the manner in which OSPF packets are authenticated. In OSPF for IPv4, a two-byte *authentication type* field is used to identify the procedure for packet authentication. This field is followed by an eight-byte *authentication* field that contains the data used by the specified scheme. The types of authentication schemes used include null authentication (type 0), simple password (type 1), and cryptographic authentication (type 2). Definitions of all other authentication schemes are reserved. In OSPF for IPv6, though, there is no need for separate authentication fields because the IPv6 header itself provides this support (see Figure 3.23). As a result, authentication was removed from the OSPF version 3 protocol. In its place is the *instance* field, which enables multiple instances of OSPF to be run over a single link. With the exception of these two differences, both headers contain similar fields. A brief description of these common fields follows.

- The **version number** field contains the version of OSPF that is being implemented. In Figure 7.11(a) the version number is 2 and in Figure 7.11(b) the version number is 3.
- The **type** field specifies the type of OSPF packet that is being transmitted. Several types are possible:
  - — *Hello* packets are used to establish and maintain neighbor relationships and are sent periodically on all interfaces.
  - — *Database descriptions* packets describe the contents of the link-state database and are exchanged between routers when an adjacency is being initialized.
  - — *Link-state request* packets are used to update out-of-date descriptions a router has in its link-state database after exchanging database descriptions with a neighboring router.
  - — *Link-state update* packets are used to flood LSAs.
  - — *Link-state acknowledgment* packets are used to explicitly acknowledge link-state update packets.
- The **packet length** field specifies the length of the packet.
- The **router ID** field identifies the router of the packet's source.
- The **area ID** field indicates the OSPF "area" to which the packet belongs. As indicated earlier, all OSPF packets are associated with a single area.
- The **checksum** field is a standard CRC checksum (see Chapter 5).

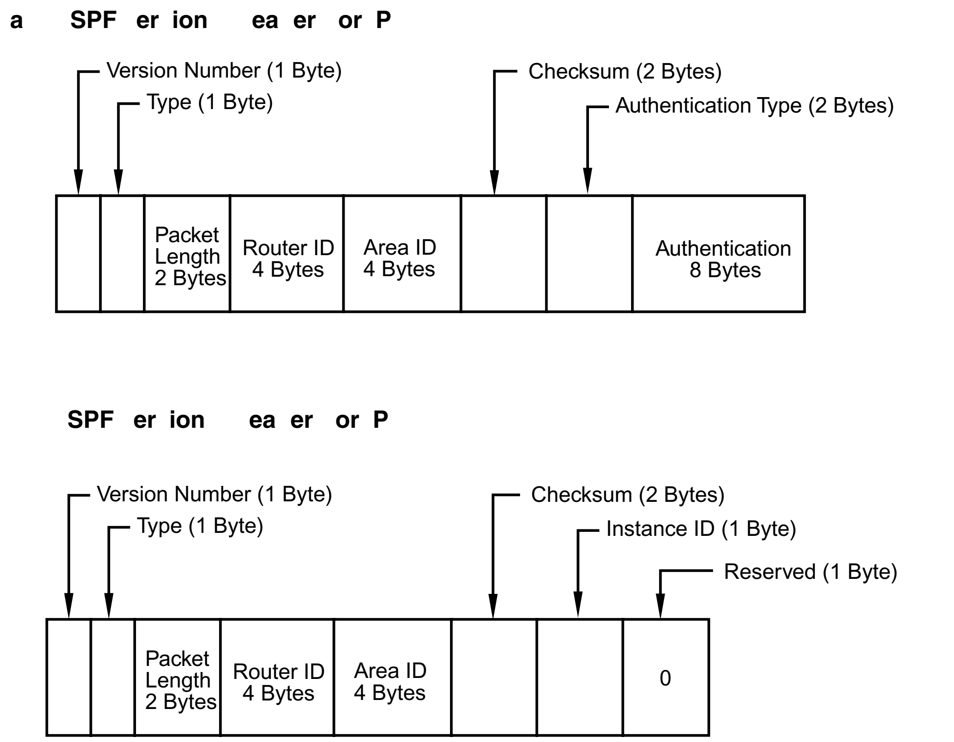


FIGURE 7.11 OSPF version 2 (a) specifies a 24 byte header and is used for IPv4 and OSPF version 3 (b) specifies a 16-byte header and is used for IPv6. With the exception of their size and the absence of authentication fields in OSPF for IPv6, both headers contain similar fields as discussed in the text. Source: Adapted from RFC 2328 and RFC 2740.

29. How does OSPF work?

In an OSPF environment, a collection of networks and hosts (i.e., an internetwork) is grouped together to form an area. Routers within an area, called intraarea routers, route packets among the networks of that area. Intraarea routers maintain identical topological data. OSPF areas are interconnected via area border routers, which keep separate topological data about the areas to which they are connected. These areas can then be interconnected to form an autonomous system (AS). Thus, in an OSPF environment, routers are connected together to form networks. These networks can in turn, be connected to form areas. Autonomous systems are then formed by interconnecting areas. To illustrate this concept, consider Figure 7.12, which shows an OSPF environment that consists of three areas. Routers R1 and R2, R4 and R5, and R8 and R9 are intraarea routers for areas 1, 2, and 3, respectively. Furthermore, R1 is an area border router for area 1, R4 acts as an area border router for area 2, and R7 is the area border router for area 3. Each area is a separate autonomous system, and the intraarea routers only carry information about the networks within their areas. For example, packets originating on network N1 and destined for N3 are routed

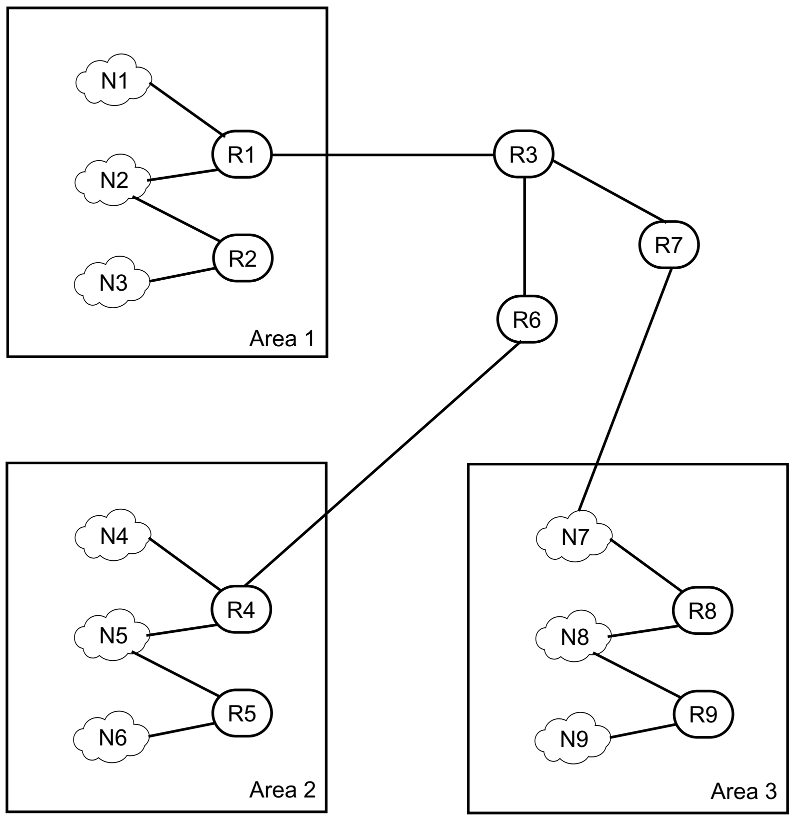


Figure 7.12 A sample OSPF routing environment. Source: Adapted from Moy, 1990.

internally via R1 and R2. Packets originating on N1 and destined for N7, though, must be directed to area border router R1. These packets are then forwarded to R3 and R7.

Note that with areas in place, OSPF routing occurs at two different levels. The lower level is *intra*area routing. The higher level is *inter*area routing, which consists of traffic that traverses the backbone. The backbone in Figure 7.12 is comprised of routers R3, R4, R6, and R7. To minimize the amount of routing updates on the backbone, each area has a *designated router* and a backup designated router. Within an area, each router exchanges link-state information with the designated router. This router, or its backup if the designated router fails, is then responsible for generating link-state advertisements (LSAs) on behalf of that network.

When a new router is first added to a network, it sends a “hello” message to each of its neighbors. Hello messages are also sent periodically by all routers to inform their neighbors that they are still alive. Using a link-state algorithm such as Dijkstra’s shortest path algorithm (see Box 7.3), OSPF routers build a topological database consisting of their view of the network. This information is then transmitted via link-state advertisements to neighbor routers. For intradomain routers, LSAs are exchanged only with routers within their area. Area border routers exchange LSAs with other area border routers.

30. Are there any other routing protocols that I should know about?

Three in particular are IGRP, EIGRP, and IS-IS, which we briefly mentioned earlier. All three are interior gateway protocols, and like OSPF are alternatives to RIP. IGRP and EIGRP are proprietary Cisco Systems protocols, and hence appropriate for Cisco environments. IS-IS is an OSI routing protocol. A brief description of each follows.

IGRP The Interior Gateway Routing Protocol is a proprietary protocol developed by Cisco to address some of the problems associated with routing in large, heterogeneous networks. The key difference between IGRP and RIP is the routing metric. A mathematical formula that takes into consideration factors such as bandwidth, delay, reliability, maximum packet size, and load is used to calculate a metric value. The path with the smallest calculated metric is determined as the least-cost (i.e., "best") path. Another difference between IGRP and RIP is IGRP employs the concept of "trusted neighbor," which is a neighbor router from whom a router will only accept routing updates. Trusted neighbors are defined as part of the configuration process. Similar to RIP, IGRP addresses issues related to routing loops by implementing the concepts of split-horizon and hold-downs.

EIGRP Enhanced IGRP is another proprietary routing protocol designed by Cisco and optimized for networks using Cisco routers. It combines the best features of distance-vector and link-state routing protocols. For example, EIGRP uses "hello" messages to learn of neighbor routers. It also uses a specially designed protocol, the Reliable Transport Protocol, to transmit routing updates instead of using broadcasts. Subnet masking is also supported by EIGRP. Routing metrics are distance-vector based and are calculated via Cisco's diffusing-update algorithm (DUAL).

IS-IS The formal title of this protocol is "Intermediate System to Intermediate System Intradomain Routing Exchange Protocol. It is an OSI protocol designed to run within an AS, which is called a "routing domain" in the OSI world. IS-IS uses a link-state routing algorithm to calculate least-cost paths. Overall, its operation is similar to OSPF: IS-IS maps a network's topology by flooding the network with link-state information.

31. I have heard the term "cider" used when discussing routing. Is this a protocol invented by Apple Computer? You know, like Apple cider?

Cute. You are referring to *Classless Interdomain Routing* (CIDR), which allows sites to advertise multiple Class C networks by using a single prefix. CIDR, which is also discussed in Chapter 3, was developed as a solution to the routing table explosion caused by the rapid growth of the Internet. As the Internet grew, the folks in charge of assigning Internet address began issuing multiple class C addresses to organizations. This meant that more individual networks needed to be announced for routing purposes and maintained by routers. With more networks needing to be announced, coupled with RIP's "feature" of updating routing tables every 30 seconds, most of the available Internet bandwidth was being used for routing table updates rather than user data. CIDR alleviated this problem by summarizing Class C network prefixes and using them as routing entries instead of the actual network addresses. For example, in a CIDR-less environment, an organization that is issued 96 Class C network addresses must have an entry for each address in its router's routing table. With CIDR, though, a special prefix is assigned to these 96 networks that indicates they all belong to the organization's routing domain. Thus, only one routing entry is needed instead of 96. OSPF, BGP, and RIP support CIDR. More specific details about CIDR are given in Chapter 3.

32. Now that I have some understanding of routing and routing protocols, could you tell me the difference between a router and a brouter?

Brouter, huh? There's a term we haven't heard in a while. As indicated earlier, routers operate at the network layer. Their job is to interconnect physically different networks and route packets from one network to another. They determine a path to a destination node and then start the packet on its way. Many routers support more than one network-layer protocol (e.g., IP, IPX, AppleTalk). Thus, if a packet originating from an IP network is passed to an AppleTalk network, a router rebuilds the packet to the proper form so it can be interpreted by a node on the AppleTalk network. Since routers operate at the third layer of the OSI model, they do not forward broadcast packets unless they are configured to do so. Routers also employ routing protocols that determine the least-cost path a packet is to travel from source to destination nodes.

A *brouter* combines the features of a bridge and router. Its name comes from "bridging router." It has the forwarding capabilities of a router and the network protocol independence of a bridge. Brouters can process packets at either the data link or network layers. When used as a bridge, brouters forward frames and can be used to filter local traffic. When used as a router, brouters transfer packets across networks. Brouters had their 15 minutes of fame in the early 1990s. They have since been replaced by switches.

33. Speaking of switches, what's the difference between a switch and a router?

Before we examine the differences between switches and routers, let's briefly review the history of switches and switching. The concept of switching in the telecommunications industry has its roots in circuit-switching. Recall from Chapter 2 that a circuit-switched network employs connection-oriented links. In a circuit-switched network, a dedicated hardware switch first establishes a physical circuit between the source and destination nodes before communication takes place. The switch also keeps this circuit in place for the duration of the transmission. Furthermore, the switch is responsible for managing the link and addressing all related end-to-end connection issues (e.g., flow control, circuit path).

In a packet-switched network, instead of using a dedicated physical circuit for every node-to-node communication, nodes share a communications channel via a virtual circuit. Bandwidth in a packet-switched network is dynamically established and released on an as-needed basis. Thus, a packet-switched network uses bandwidth more efficiently than a circuit-switched network. Furthermore, if the link is connectionless, then messages are partitioned into packets and sent to the destination one at a time and not necessarily in the correct order. By doing so, the switch does not have to concern itself with end-to-end communication issues.

One of the key differences between a connection-oriented and a connectionless link is that the former requires knowledge of the state of the network, whereas the latter does not. Prior to the development of local and wide area data networks, switching was always connection-oriented. In the data network world, however, switching became connectionless. For example, when a router routes a packet through a network, no advance path is established, there is no concern about flow control, and heck, the router doesn't even care if the packets are arriving in the correct order. All of these issues are left for the upper layers. Thus, a router acts like a connectionless switch. So, the primary difference between a switch and a router is really one of semantics. Switches historically infer connection-oriented links; routers use connectionless links.

34. OK. But you answered my question from a conceptual perspective. What I want to know is what distinguishes a layer-2 LAN switch from a layer-3 router?

Sorry about that. *In the strictest sense,* LAN switches are layer-2 devices that examine and use layer-2 data (e.g., MAC addresses) to forward or filter traffic. Since they operate at the data link layer, these devices are supposed to be transparent to protocols operating at higher layers. This implies that regardless of the network protocol being used, switches pass or discard frames independent of network protocols and cannot filter broadcasts. Similarly, and again *in the strictest sense,* routers are layer-3 devices that use layer-3 data (e.g., network addresses) to forward or filter traffic. Routers filter packets based on network protocols, and they can filter broadcasts. Most routers also incorporate some sort of bridging capability. Hence, they can operate at either layer 2 or layer 3.

In the real world, though, LAN switches have encroached on the router's territory. Many LAN switch vendors now incorporate traditional router functionality into their products. For example, some Ethernet switches are capable of examining layer-3 header information, which is then used to filter network protocols or broadcasts. Some switches are capable of creating *virtual local area networks* VLANs (see Chapter 8) using either MAC sublayer addresses (layer 2) or network addresses (layer 3) as the basis for forwarding/filtering frames or packets. As a result, switches have begun replacing routers at the local area network level.

The replacement of routers with switches in LAN configurations did not go unnoticed by router manufacturers. Cognizant of the LAN switch's encroachment, router manufacturers began modifying their products to incorporate a switch's primary feature—wire speed operations. To provide routers with this capability, router manufacturers designed and implemented *application-specific integrated circuit* (ASIC) chips that perform traditional router table lookups and packet forwarding at hardware speeds. ASIC-based routers still examine every packet and calculate network routes as traditional routers do; they just do it much faster. Thus, a layer-3 switch is simply a wire speed router. Physically, these routers provide switchlike performance, but logically, they perform the same operations as legacy routers. Furthermore, unlike layer-2 switches, which establish connection-oriented links between source and destination nodes, routers still route packets on a connectionless basis. Nevertheless, ASIC-based routers are referred to as layer-3 switches or routing switches. Along this same line, because LAN switches have layer-3 functionality, they are referred to as layer-3 devices. Regardless of what you call them (is a router a switch, or is a switch a layer-3 device), the bottom line is this: Layers 2 and 3 of the OSI model are merging, and it is becoming difficult to distinguish between switches and routers.

35. While we're on this topic, what is layer-4 switching?

The concept of layer-4 (or higher) switching is based on using information from the upper layers (layers 4, 5, 6, or 7) to make routing decisions. For example, Web-based packets might be routed in one direction and e-mail packets in another even if both packets have the same destination address. The Web-based packets might take a route that has higher bandwidth capability; the e-mail packets' route might have more delays. Another illustration is Web-browsing traffic, which might take a different path than an electronic commerce-based packet.

The idea behind layer-4 switching is to examine each packet and use its upper-layer information to make routing decisions. Routers that have this capability are being touted as layer-4 switches. This terminology, however, is a malapropism, however, because switches are connection-oriented; routers are connectionless. Switching implies establishing a connection between source and destination ports, which does not occur at layer 4. Regardless of the type of information being used to determine a path or the layer that is providing this information, packets are still being forwarded on a connectionless basis. It is more appropriate to refer to layer-4 switches as either layer-2 or layer-3 *application switches* because application information from upper layers is being used for routing decisions.

36. How different is layer-2, layer-3, and layer-4 switching from IP switching?

"IP Switching" initially was the name a network vendor (Ipsilon Networks) coined to describe its proprietary layer-3 switching strategy. Ipsilon has since been acquired by Nokia, and the term now refers to different strategies for speeding up the processing of IP traffic. The focus of each of these strategies is to apply layer-2 switching technology to layer-3 routing. The primary reason for doing this is to improve the packet forwarding performance of routers and to enable routers to provide sufficient network guarantees to support a specific quality of service (QoS) level. Several strategies follow without explanation. Additional information about these strategies can be found from the references. See also Chapters 5, 8, and 14 for more information about QoS and class of service, CoS.

One strategy is to run IP over ATM (Chapter 14). This strategy itself has several approaches. These include ATM LAN Emulation (LANE), Classical IP Over ATM (IPOA), Next Hop Resolution Protocol (NHRP), and Multiprotocol Over ATM (MPOA), which is an extension of LANE and uses NHRP. Of these approaches, MPOA is receiving considerable attention. A second strategy is Ipsilon's IP switching technology, which employs a specially designed IP switch that can distinguish between "flow-oriented" traffic (e.g., ftp, http, multimedia) and "short-lived" traffic (e.g., e-mail, name server lookups). The IP switch processes all short-lived packets just like traditional routers. Flow-oriented packets, however, get switched to their destinations at hardware speeds. A third IP switching strategy is the IETF's Multiprotocol Label Switching (MPLS) standard, which combines Cisco Systems' tag switching approach with IBM's ARIS (Aggregate Route-based IP Switching) strategy. MPLS is discussed later in this chapter. Other strategies include ASIC-based wire-speed routers that process several million IP packets per second and layer-3 switches as discussed earlier.

37. Speaking about QoS, how is QoS implemented in the Internet?

One way this is done is through the *resource reservation protocol* (RSVP), which is an Internet Engineering Task Force (IETF) standard that operates at layer 3 of the OSI model. RSVP can be thought of as an IP-based quality of service (QoS) protocol that provides a mechanism to control network latency for specific applications. This is done by prioritizing data and allocating sufficient bandwidth for data transmission. QoS is inherent in technologies such as token ring and ATM, but is absent in Ethernet/802.3 and IP. With RSVP, though, Ethernet/802.3 or IP end nodes can reserve a specific amount of bandwidth from the network for a particular transmission. This feature is critical for transmitting data from time-sensitive applications such as real time voice and video. For example, a video conferencing or multimedia application might receive a high priority tag that requires a certain amount of bandwidth, a specific transmission rate, and maximum latency. To effect RSVP across a WAN, every router that is along the path an RSVP data packet traverses must support RSVP. If not, then the application fails. Furthermore, if the WAN cannot support an RSVP request (e.g., there is insufficient bandwidth available), then the application will not run. Another strategy involves using the Multiprotocol Label Switching (MPLS) protocol, which is used in Internet-based virtual private networks (VPNS). Finally, IPv6 (Chapter 3) has provisions for QoS.

38. In your previous response, you mentioned two things I am unfamiliar with: VPN and MPLS. What are these?

We'll begin with VPN, which stands for *virtual private network*. From a conceptual perspective, a VPN represents an IP connection between two sites over a public IP network. A VPN makes it possible for organizations to use a public network such as the Internet as their own private WAN. Examining this concept further, a VPN is first and foremost a virtual network (this is the V and N of VPN), which implies that the paths data travel between a source and destination are being shared by other transmissions. Thus, customers share the network infrastructure instead of using dedicated private lines. Second, it is a private network (this is the P and N in VPN), which implies that the data transmitted between a source and destination are not accessible to unauthorized users. This implies that the network must address security issues and that the network provider as well as the network's users must provide end-to-end data protection. A VPN, then, involves using a public network infrastructure, such as the Internet, as a private WAN where data are transported between source and destination nodes. Furthermore, data transported across a VPN are encrypted so that only the source and destination nodes can decrypt them. (See Chapter 16 for additional information about encryption and decryption.) In this manner, a publicly accessible network like the Internet can be used for moving highly confidential information in a secure manner. Although a shared network can be designed exclusively for VPN customers, VPNs today are mostly IP-based networks such as the Internet.

39. Call me clueless, but what's the big deal about VPNs?

When it comes to providing network resources to its employees and customers, organizations generally have different connectivity requirements. For example, in addition to providing on-site employees with access to network resources via the company LAN, most business organizations must also address the connectivity needs of employees housed at remote sites such as a branch or satellite office, telecommuters, and traveling employees such as salespeople. Furthermore, access to corporate intranets by remote users must be provided in a secure manner, and extranets must be established so that suppliers and customers also have controlled access to selected network resources. Each connectivity type—remote-access, site-to-site intranet, and extranet—has its own set of requirements. VPNs provide a relatively simple manner in which to address these mixed connectivity requirements because the same network infrastructure can be used for all three (see Figure 7.13).

40. How are VPNs kept private?

VPNs maintain private transmissions by using tunnels. Recall earlier in the chapter that we mentioned the concept of tunnels from the perspective of packet encapsulation (Figure 7.7). In the context of VPNs, a tunnel is pretty much the same. A VPN tunnel establishes an end-to-end connection and encapsulates the data with new packet headers so the packets are delivered to the specified destination. What makes the connection private is that network traffic can only enter the tunnel at an endpoint. Thus, access to a VPN is restricted to members of a specific user group. For example, in Figure 7.13, a site-to-site intranet VPN, a remote-access VPN, and an extranet VPN can be established by creating tunnels between each site's respective access point and service provider, and then among the service providers themselves.

41. Does a private transmission imply that the transmission is confidential?

No. As indicated in our previous response, VPN tunnels provide a private pathway to transport data between source and destination nodes. This does not necessarily mean, though, that the transmission is secure. VPNs provides confidentiality by encrypting data. This involves converting sensitive data into a coded form at the source and then decoding the data into a meaningful form at the destination. (Chapter 16 discusses several data encryption techniques.) Encryption is an extremely important component of VPNs that use a publicly accessible network such as the Internet as a private WAN because the underlying technology of the Internet was never designed with security in mind. Several protocols that implement encryption techniques have been developed to help secure VPNs. These include the *Point-to-Point Tunneling Protocol* (PPTP), *Layer 2 Forwarding* (L2F), *Layer 2 Tunneling Protocol* (L2TP), and *IP Security* (IPSec). A brief description of each follows.

The Point-to-Point Tunneling Protocol (PPTP) The Point-to-Point Tunneling Protocol was developed by several organizations, including the Internet Engineering Task Force (IETF), Microsoft, and U.S. Robotics (now part of 3COM). PPTP uses Microsoft's proprietary Point-to-Point Encryption algorithm, which provides encryption and authentication for remote dialup and LAN-to-LAN connections. For dialup users, PPTP can be provided either directly by a PPTP-enabled client, or indirectly via a PPTP-enabled server through an Internet service provider (ISP). Regardless of who provides the service, two connections are established. A control session is responsible for establishing and maintaining a secure tunnel from sender to receiver, and a data session provides data transmission. In

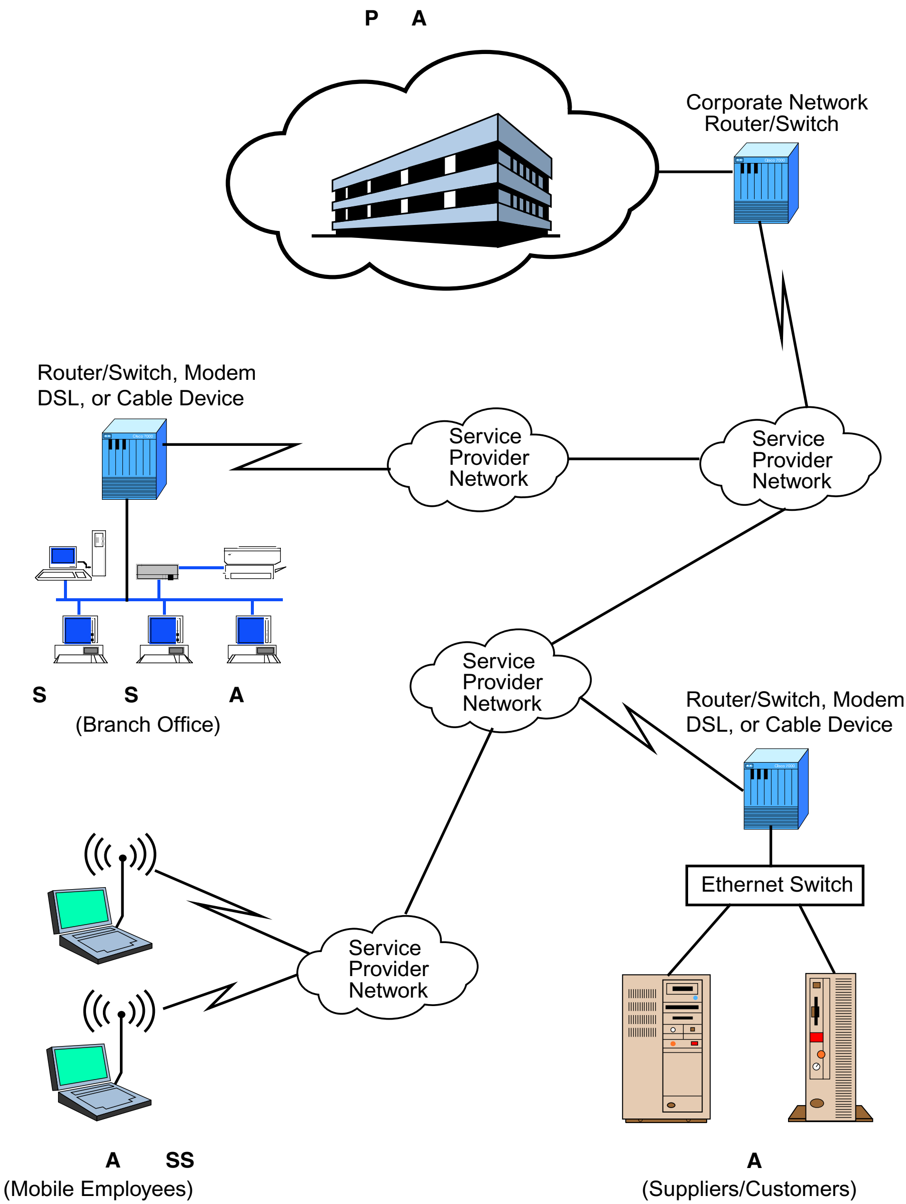


FIGURE 7.13 Most organizations today have mixed connectivity requirements. These include *intranets*, which enable branch offices to access corporate network resources; *extranets*, which provide customers and suppliers with controlled access to select corporate network resources; and *remote access* connections, which provide nomadic employees with access to corporate network resources. Source: Adapted from Dickson, 1999 and Zines, 2000.

LAN-to-LAN applications, a tunnel is established between NT servers. PPTP supports several network protocols including IP, IPX, NetBEUI, and NetBIOS.

The Layer 2 Forward (L2F) Protocol The Layer 2 Forward protocol provides tunneling between an ISP's dialup server and the network. In this application, a user establishes a dialup Point-to-Point Protocol (PPP) connection to the ISP's server. This server then wraps the PPP frames inside an L2F frame, which is then forwarded to a layer-3 device (a router) for network transmission. The router is responsible for user authentication and network addressing. L2F does not provide any data encryption, and its user authentication capability is weak.

The Layer 2 Tunneling Protocol (L2TP) The Layer 2 Tunneling Protocol defines a method for tunneling PPP sessions across a network. It combines PPTP and L2F.

IP Security (Sec) IP Security is a suite of protocols developed by IETF. The suite includes an authentication header (AH), an encapsulating security payload (ESP), and the Internet key exchange (IKE). (*Note:* IKE was originally known as the Internet Security Association and Key Management Protocol with the Oakley key exchange protocol—ISAKMP/Oakley Resolution.) Operating at layer 3, IPSec provides address authentication via AH, data encryption via ESP, and automated key exchanges between sender and receiver nodes using IKE. Although PPTP, L2F, and L2TP support multiprotocol routing and provide some VPN security services, they are more applicable to remote access connections (i.e., dialup). IPSec, on the other hand, provides end-to-end data encryption and authentication for VPNs.

It is important to note that all of the preceding VPN security strategies do not provide the same level of security. For example PPTP, L2F, and L2TP require an additional layer of encryption to provide secure data transmissions. Thus, if a VPN employs any of these security strategies, then the VPN is really only a virtual network and not a true VPN. Of the various security strategies described, only IPSec creates encrypted tunnels. This is why the VPN Consortium supports only IPSec, PPTP with RC4 encryption, and L2TP under IPSec as acceptable VPN security strategies.

42. Tell me more about IPSec.

As indicated above, IPSec is really a suite of protocols that contains an authentication header, encapsulating security payload protocol, and Internet key exchange. The authentication header contains six fields (Figure 7.14) The *next header* field is one byte long and identifies the higher-level protocol that follows the AH. The *payload length* field is also one byte long and specifies the length of the authentication data field. The *reserved* field is just that—it is a two-byte field reserved for future use and always set to zero. The *security parameters index* (SPI) field is four bytes long and identifies the security protocols (called the "security association") being used in the packet. This is followed by the four-byte *sequence number* field, which serves as a counter that identifies the number of IP AH packets it has received that bear the same destination address and SPI data. Although the sending node must include this information in outgoing packets, the receiving node does not have to process this field. The sequence number also protects against the receipt of

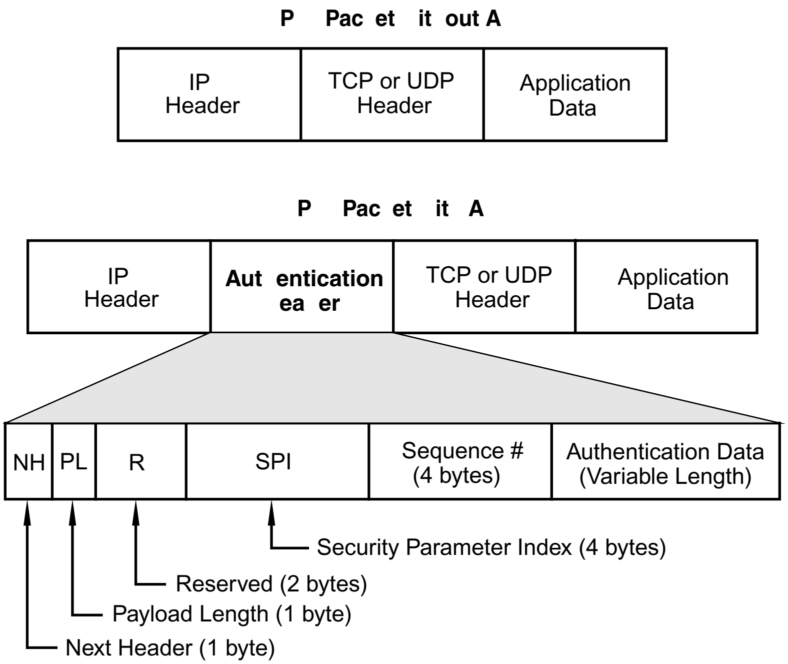


FIGURE 7.14 The authentication header, which is part of the IPSec protocol suite, provides authentication service to IP. AH safeguards data from being altered during transmission. The AH immediately follows the IP header in a standard IPv4 packet, but comes after the hop-by-hop, routing, and fragmentation extension in an IPv6 packet. NH identifies the higher-level protocol that follows the AH. PL specifies the length of the authentication data. R is reserved for future use and is set to zero. SPI identifies the security protocols (called the "security association") being used in the packet. The sequence number serves as a counter that identifies the number of IP AH packets it has received that bear the same destination address and SPI data. The sequence number also protects against the receipt of duplicate packets. The authentication data field contains the integrity check value (ICV), which is a digital signature of the packet. Authentication algorithms used to generate this signature include DES, MD5, the secure hash algorithm (SHA-1), and others (see Chapter 16). Source: Adapted from Thayer, 1997.

duplicate packets. The last field is the *authentication data.* This is a variable-length field that contains the *integrity check value* (ICV), which is a digital signature of a packet. Authentication algorithms used to generate this signature include DES, MD5, the Secure Hash Algorithm (SHA-1), and others (see Chapter 16). The AH immediately follows the IP header in a standard IPv4 packet, but after the hop-by-hop, routing, and fragmentation extension in an IPv6 packet. (The destination options extension header of an IPv6 packet can precede or follow the AH.)

The encapsulating security payload header contains seven fields (Figure 7.15). The first two fields, the *security parameters index* and *sequence number*, are the same as those in the authentication header. Collectively, these two fields are referred to as the control header. The next field, *payload data,* contains the encrypted version of the user's original data. It also contains optional initialization vector (IV) information if the encryption algorithm used to encrypt user data requires any type of synchronization (e.g., DES control information). The fourth field, *padding,* provides for any necessary padding requirements of the encryption algorithm or for byte-boundary alignments. This ensures that the payload data has the correct length. The *pad length* field specifies the number of pad bytes used in the padding field. The *next header* field references the payload data by identifying the type of data contained in the payload data field. These last three fields—padding, pad length, and next header—are called the ESP trailer. The last field, *authentication data,* is optional and similar to the authentication data field of the authentication header—it is a digital signature applied to the entire ESP (sans this field). The ESP header, like the AH, immediately follows the IP header in a standard IPv4 packet, but after the hop-by-hop, routing, and fragmentation extension in an IPv6 packet. (The destination options extension header of an IPv6 packet can precede or follow the ESP header.) If ESP and AH are to be used together, then ESP follows AH.

The Internet key exchange provides a mechanism for automating key exchanges between sending and receiving nodes when authentication (AH) and encryption (ESP) are used together. In such instances, both sender and receiver have to know their respective keys. Furthermore, in order to ensure secure communications across the network, these keys must be known only to the parties communicating. The IKE protocol delivers this service by using the Diffie-Hellman algorithm for key generation, and provides three different methods of key exchange: *main mode, aggressive mode,* and *quick mode.* In main mode, six messages (i.e., three back and forth exchanges) are sent between a sender and receiver. The first two messages establish a specific security policy, the next two messages contain key information, and the two last messages provide authentication information. Aggressive mode is similar to main mode and achieves the same result. The difference is that there are only two exchanges (four messages sent between sender and receiver) instead of three. Quick mode is used to generate new keys after all necessary information has been exchanged between the communicating nodes via main or aggressive modes.

IPSec essentially operates in one of two modes. In *tunnel-mode,* the entire IP packet is encrypted and wrapped within a new IPSec packet (see Figure 7.15). In *transport-mode,* only the data payload is encrypted. For more information about IPSec, see http://www.ietf.org/html.charters/ipsec-charter.html.

43. Let's get back to VPN tunnels and Figure 7. 13. If the connections between the corporate and noncorporate networks (site-to-site intranet, extranet, and remote-access) are VPN tunnels, then how do noncorporate users access the Internet to do web surfing for example? Do they have to go through the corporate network?

Unless other arrangements are made, the answer is yes. One way this can be avoided is for the noncorporate site to have a separate, direct, and independent connection to the Internet. Another way is to implement something called *split tunneling* (Figure 7.16). This option gives noncorporate users access to the corporate network through the encrypted VPN tunnel and direct access to the Internet using an unsecure link, called a *cleartext tunnel*, simultaneously. As noted in your question, without either of these options Internet

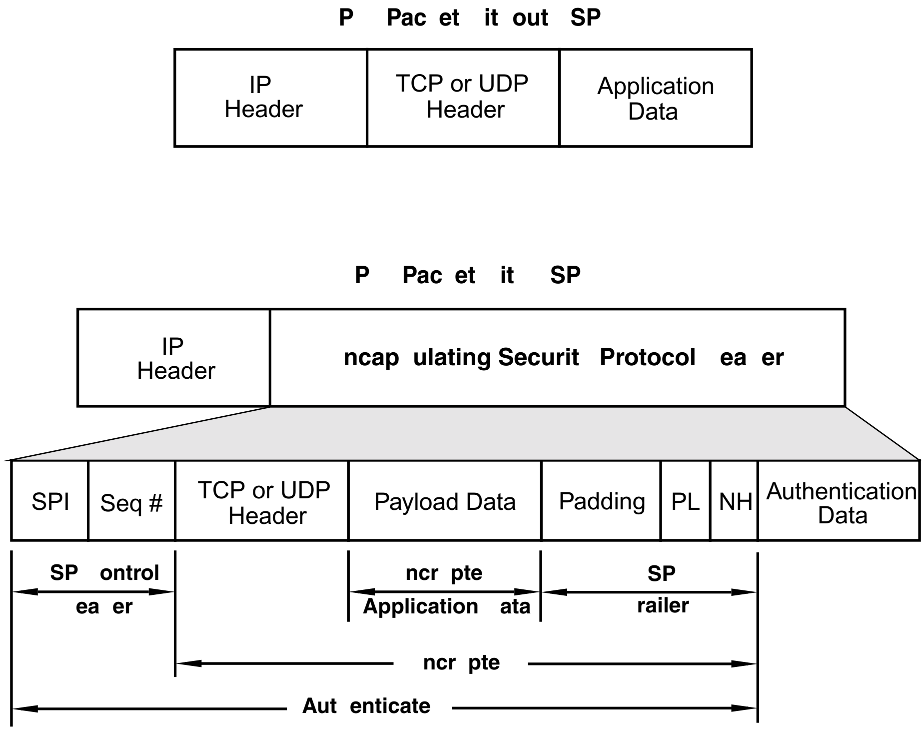


FIGURE 7.15 The Encapsulating Security Protocol header, which is part of the IPSec protocol suite, provides encryption service to IP. The ESP header immediately follows the IP header in a standard IPv4 packet, but comes after the hop-by-hop, routing, and fragmentation extension in an IPv6 packet. The security parameter index (SPI) and sequence number (Seq #) are the same as in the authentication header (see Figure 7.14). The payload data contain the encrypted version of the user's original data as well as optional initialization vector (IV) information if the encryption algorithm used to encrypt user data requires any type of synchronization (e.g., DES control information). The ESP trailer consists of padding, pad length (PL), and the next header (NH) fields. Padding provides for any necessary padding to ensure that the payload data have the correct length; PL specifies the number of pad bytes used in the padding field; and NH identifies the type of data contained in the payload data. The authentication data field is optional and contains a digital signature applied to the entire ESP (sans this field). Source: Adapted from Thayer, 1997.

access is only accessible to noncorporate users indirectly via the corporate Internet link. Thus, the data flow is from the noncorporate user's node through the encrypted VPN tunnel to the corporate network and then out the corporation's Internet gateway, which needlessly increases traffic flow through the corporate network.

44. What about the costs associated with operating a VPN?

A VPN strategy can realize a company considerable cost-savings over competing technologies. For example, competing remote-access solutions are usually 30 percent to

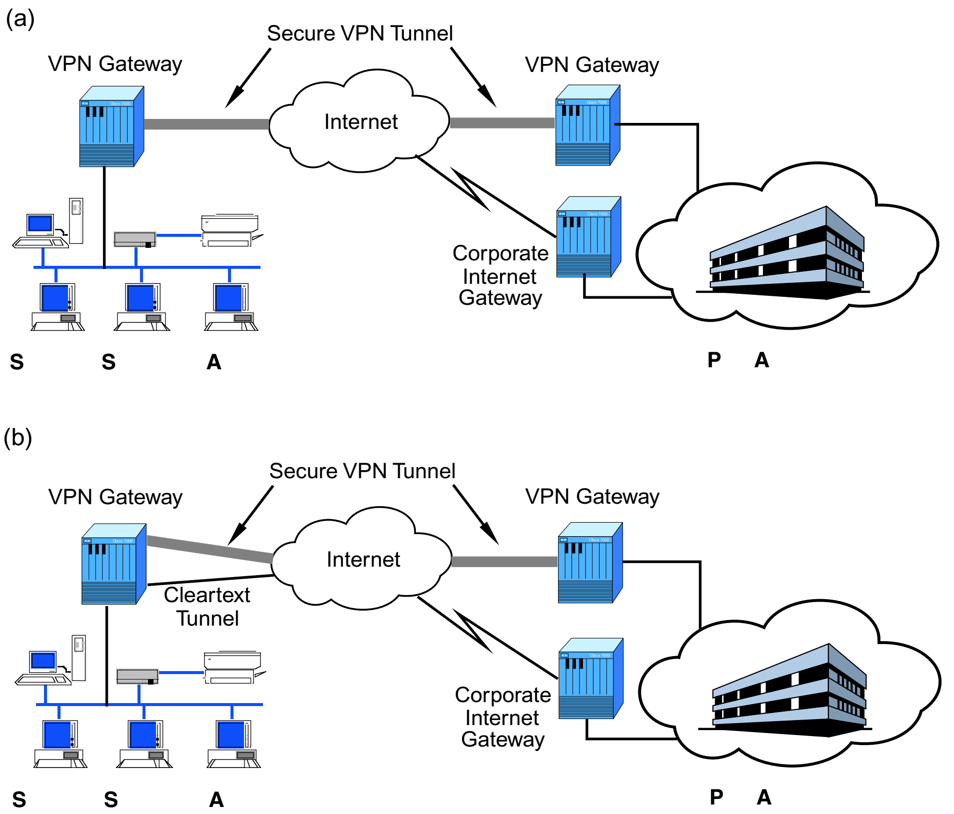


FIGURE 7.16 Without a separate, independent Internet connection, site-to-site intranet VPN users (a) must access the Internet indirectly via the corporate Internet connection, which increases traffic flow through the corporation's network. One way to resolve this is through *split tunneling* (b), which provides site-to-site intranet VPN users with simultaneous access to the corporate network through the encrypted VPN tunnel and direct access to the Internet using an unsecure link called a *cleartext tunnel*. Source: Adapted from Trudel, 2000.

70 percent more than a VPN solution, and site-to-site savings can be as high as 70 percent when compared to private-line solutions. In addition to cost, though, a VPN strategy also provides unparalleled worldwide access. An Internet-based VPN, for example, is unmatched by any other network service when it comes to making network resources accessible globally. Following is a case study of an organization with more than 60 sites located in the United States, Europe, and Asia. This should give you an idea of the costs and benefits associated with VPNs. (*Note:* All costs are given in U.S. dollars.)

The current network, used to interconnect the sites, is an extensive network composed of frame relay, leased line, dial-up, and X.25 packet switching (see Appendix E). In this network, the average connection speed is 56 kbps with an average per-month connection cost of approximately $176,000. This amount is for communications costs only and does not include any provision for the required modem pools and other types of interconnection hardware. By using frame relay and upgrading only 20 of the 60 sites to T1 circuits (1.544 Mbps), the monthly communications costs are expected to exceed $510,000 per month, $360,000 of which is for the 20 T1 sites alone because of their overseas locations. The other 40 sites require the same services, but not necessarily the entire range of connection speeds required at other, larger sites. In the major sites, the need to upgrade from 56 kbps to T1, and in a couple of cases, T3 (45 Mbps), is forthcoming due to a major change in corporate use of network resources to move large amounts of data. Other needs include:

- allowing customers to access specific applications on in-house systems (there may be up to 20,000 customers doing this daily in the future);
- providing extranet connectivity to customer sites using the SNA, AppleTalk, and IPX protocols in addition to IP;
- enabling remote, secure, corporate user access via modems (there could be as many as 2000 users accessing internal resources daily);
- expanding small business sites and office presence worldwide so that a small office in a remote city consisting of two to four users and associated equipment can rapidly interconnect newly acquired companies;
- enabling rapid implementation of high-speed connections due to seasonal changes or promotional issues at offices worldwide;
- managing network resources with minimal or no human utilization from internal resources;
- adding modem, ISDN and other remote low-speed connections without affecting specific in-place network resources from a configuration perspective;
- adding customer and vendor connections quickly for multiple protocol suites with minimal or no internal resource expenditure to add connection capability;
- providing intranet server capabilities to all employees domestically and internationally regardless of network protocol access type used in a user's local area;
- developing consistent configuration rules and conformance criteria;
- implementing internationalized method of standards and performance criteria for all sites so that consistent performance and access reliability are achieved;
- allowing network performance to scale from very small (single system) to millions of session accesses per day at a given site with the same connection methodology for simplicity;
- implementing expandable network architecture that can handle not only multiple protocols and interconnection of same, but multiple versions of the protocols at the same time;
- supporting audited and logged network access via secured facilities at each site;
- implementing very stringent security controls for highly critical systems and network interconnects to ensure that only properly authorized users gain access to critical components; and
- providing a redundant and resilient network environment in the case of performance adversity or network outage by one or more vendors to critical locations.

To solve these problems, a network was designed with the following components:

1. The same public IP provider at all sites will be used whenever possible. For this implementation, two vendors were selected for interconnection of sites because they possess high-speed access facilities at all the customers site locations. Further, both vendors have substantial dial-up and ISDN facilities that allow remote location interconnection at very low or fixed price configurations. Selecting two vendors also allows for diversity and support of user sites that may not have connection facilities to the preferred vendor of the two but do have access to the other.
2. A multiprotocol firewall was selected to provide site-to-site multiprotocol VPN for all protocols, five-layer network security (frame, packet, application, stateful, and proxy in the same product at the same time), scaling (Intel 80486 through Alpha SMP systems), client VPN proxy facilities for remote single system and laptop users, strong authentication for specific systems personnel, remote management of the firewall, and many other facilities that are essential in providing secure networking connectivity. An additional reason for selection is the ability for the product to be used on Internet connections, intranet (there is a server and desktop version) and extranets when connecting to customer or known third party sites.
3. Common network vendor for all interconnection WAN routers and hubs to ensure proper network interconnectivity, management and minimization of manpower and technical expertise requirements.

With this approach, the 20 main sites—which had an average upgrade fee of $18,000 per site per month, and required specific routing topologies to one or two centralized sites—can be upgraded for an average per-site cost of $7000 per month. (This figure is higher in some countries, but substantially lower in others, for a T1 connection.) This per month communications fee is normal for a public IP connection. This means that flat-rate costs per month for the 20 sites are approximately $140,000. This is 39 percent of the original cost estimate for the upgrade at the same speed, but also includes all the other ancillary network connectivity requirements.

Additional costs for this implementation include 20 full-functionality firewall facilities (an average of $20,000 per site for an Alpha-based Windows NT system including firewall software and support). Since the new connectivity method provides a $13,000 per month savings, the firewall and all facilities for its interconnection is paid for in less than two months per site. Additional expected per-user costs for Client VPN is approximately $250 per system, which will grow slowly. Costs for modem and ISDN pools are eliminated as are the maintenance, operations, and network management costs and efforts.

For completeness, the customer examined the potential use of a router-only solution with VPN capabilities, but the solution could not solve the remote laptop and small site VPN problem nor the filtering and security management issues that only a firewall can solve. While a router is required for this type of connection to be feasible, so is the right type of firewall so that all the ancillary controls, audit trail, logging, security services and multiple protocol session control and management are available regardless of how the customer needs to connect to the facility. While this solution is still implemented at this writing, initial results are very positive and the cost savings described in this example are real.

45. What are some typical VPN configurations?

There are many different types of configurations possible. A brief description of the more salient ones follows:

Router-to-Router VPN-on-Demand Tunnel Connections Between Sites In this implementation, a VPN-capable router is set up to know that when a connection is made to a specific IP address on the connected network, it should set up an encrypted linkage for all traffic between the two routers. This is often also called an encrypted tunnel facility because the connection does not individually encipher the sessions as much as it creates a master session between the two routers and channels all user traffic inside the master session (like moving cars through a tunnel). The tunnels are created with the first user connection between the site(s) and are persistent until the last user disconnects from the site pair, which causes the routers to stop the tunnel session. This type of VPN relies on router compatibility. Among others, routers must have compatible VPN capabilities, key exchange, and cryptographic support. This method is also highly vendor specific—usually, two different vendors of routers will not interoperate in a tunneled manner. VPNs via a tunnel implementation may or may not be encrypted depending on vendor offering. A good example of this is the Layer Two Forwarding (L2F) protocol.

Router-to-Router VPN-on-Demand Multiprotocol Tunnel Connections Between Sites over an IP Network Similar to the previous definition, this type of VPN implementation allows the customer to use an IP network between two sites to carry tunneled packets for other protocols besides IP. An IP-based VPN is established between two sites over the public IP network. The routers know that when another protocol, such as Netware's IPX or Apple's AppleTalk, issues a connection request to a specific node on the other side of the IP network, a "transparent" connection needs to be established and the non-IP protocol tunneled to the remote site. This type of connectivity is extremely useful in companies that have small to medium remote sites and want the benefits and cost savings of connectivity to a shared IP network, but are not running IP as the only protocol between sites. It's also a big cost savings method for international network connections. The cost of a public IP network with multiprotocol VPN tunneling is considerably less than a 56-kbps private network connection.

Router-to-Router VPN-on-Demand Encrypted Session Connections Between Sites Like a tunnel, specific routers are defined with each other as to whether they support VPN, encryption, or other security. Unlike a tunnel, each session is encrypted and match-paired with its partner on the other side of the public network. While this is simpler to manage session-wise than a tunnel, it can have a greater amount of overhead for highly connected applications between the same two site pairs in a network.

Firewall-to-Firewall VPN-on-Demand Tunnel Connections Between Sites (See Chapter 16 for more information about firewalls.) Like the equivalent router facilities, this provides an equivalent service. The major difference is the ability to impose security rule restrictions and traffic management, auditing, authentication, data encryption, and other security features that firewalls offer but routers do not. This provides additional security and accounting information useful for management of the facilities. An example of an emerging standard for this is *IP security* (IPSec) from the Internet Engineering Task Force (IETF). IPSec is a suite of protocols that includes an authentication header (AH) and an encapsulating security payload (ESP). AH provides address authentication for IP traffic, and ESP defines IP data encryption. (See Chapter 16 for additional information about IPSec.) IPSec enables the same or dissimilar firewall vendors to negotiate a protocol methodology that provides the described VPN facilities or subsets thereof. Be careful: Some vendors' offering of IPSec do not interoperate with other vendors and only support their own firewall implementations.

Firewall-to-Firewall VPN-on-Demand Multiprotocol Tunnel Connections Between Sites over an IP Network Again, similar in nature to the router approach but with all the firewall facilities as well. For this type of VPN to work with multiple protocol tunneling, the firewall must be capable of handling multiple protocol filtering and security.

Client-to-Firewall IP Tunnel VPN Facilities In some recent implementations, a client VPN tunnel manager and encryptor software package is installed on a client system, such as a laptop. The firewall implements a proxy facility that knows how to deal with the client. The client, upon connecting to the site via the IP network, negotiates a VPN tunnel with the site firewall via the client VPN software. Once the session tunnel is activated, the firewall and client system provide a secure connection over the public IP network. In this approach, VPN client facilities are usually required for a variety of operating system environments to satisfy the remote connectivity facilities. This type of service, although it is becoming more common, is usually not seen implemented on a router-based VPN solution. This is jointly due to the need to maintain database information on the client side and the complexity of key distribution and management, which usually require a disk-based system to deal with the items involved (most routers are diskless). An example of this is the V-One implementation called *SmartGate*, which implements a proxy on the firewall side of the connection and either a soft-token or hard-token software package on the client side to connect to the proxy on the firewall for the VPN facility. Another is the proxy suite from Aventail that provides many equivalent services for NT.

Client-to-Server IP Tunnel VPN Facilities Companies such as Microsoft are implementing a VPN tunneling facility that allows the software on a client to initiate and connect a VPN tunnel between itself and either a local or remote server on a network. This provides the ability for end-to-end VPN services and, with encryption, the opportunity to provide secure VPN facilities from the source of information to the destination of information. Microsoft provides this capability with their Point-to-Point Tunneling Protocol (PPTP) currently available in Windows NT and Windows 98/2000. PPTP works hand-in-hand with IP.

Client and Server Firewall Implementation with Full VPN Capabilities This approach provides the greatest level of complexity and the greatest level of security by implementing a full firewall facility on every system on the network. This provides the VPN facilities previously described but also the ability to support full network security policy management and control on both sides of the connection (client-only VPN facilities do not provide client network access control services). An example of this type of approach is the server and client versions of Network-1 Security Solutions' FireWall/Plus, where the server and desktop machines have full firewall facilities to provide full network access control between the systems and network in addition to VPN facilities.

Dedicated VPN Box Some vendors have come up with dedicated systems that can connect either in front of or behind a router facility to implement VPN facilities between a company and a public IP network such as the Internet. These boxes are simple to implement and usually provide much higher performance than software-based solutions implemented in firewalls or via other schemes. Normally, however, they do not provide an adequate client-level security facility for VPNs and are mostly dedicated for site-to-site access. They also can be expensive for highly connected sites.

As you can see, there is a bewildering array of VPN choices and solutions depending on need and fiscal resources.

46. OK. I now have a far better understanding of VPNs. This leaves the subject of MPLS, which you mentioned earlier but have yet to discuss.

MPLS stands for *multiprotocol label switching*. Initially developed to effect wire-speed routing, MPLS evolved from several earlier technologies, including Cisco's Tag Switching and IBM's Aggregate Route-based IP Switching (ARIS) strategy (see http://www.watersprings.org/links/mlr). Since ASIC-based routers now employ switchlike performance (i.e., wire-speed route lookups and forwarding rates), MPLS is no longer used in this context but instead provides other benefits to IP-based networks, including VPNs.

47. In what way is MPLS beneficial to VPNs?

When VPN technology was initially introduced, VPNs were deployed using one of two basic models: (a) at layer 2 using point-to-point permanent virtual circuits (PVCs) with frame relay (Chapter 12) or ATM (Chapter 14) or (b) at layer 3 using tunneling protocols (as discussed earlier). Although both models have been successful, they still present VPN service providers and customers with several challenges. One challenge is the issue of scalability and flexibility. As customers' connectivity needs increase, it is difficult to add new sites quickly within a provider's network. For example, when using PVCs or tunneling protocols, a new network topology needs to be defined by the service provider for each customer. One way to resolve this is to establish a fully-meshed network in which every node is connected to every other node via a point-to-point tunnel. (Recall from Chapter 2 that a fully-meshed design, also called a closed loop, is one in which every node is adjacent to every other node.) The problem with this design, though, is that as the number of VPN sites increases, the number of tunnels needed to support the fully-meshed network increases dramatically. For example, an organization with 50 sites requires (50)(49)/2 = 1225 PVCs or tunnels for a fully-meshed network, and if full connectivity is required among 100 sites, the number of PVCs or tunnels needed increases to 4950. Furthermore, the amount of routing information that must be exchanged among routers also increases as new sites are added. Another challenge that VPN designers face is the issue of quality of service (QoS). Most VPN service providers frequently need to support various QoS implementations (see Chapter 5). In a PVC-based VPN, managing multiple service levels across multiple PVCs can be an administrative nightmare for large networks. In a tunneling protocol-based VPN, QoS is not a built-in feature and hence must be provided using another protocol such as RSVP. MPLS addresses these two challenges by supporting a fully-meshed design with quality of service (QoS) capability.

48. How different is an MPLS-based VPN compared to a traditional VPN?

As noted above, traditional VPNs involve using point-to-point frame relay/ATM PVCs or tunneling protocols such as PPTP, L2F, L2TP, and IPSec. Of these two basic implementation models, building a point-to-point network within a service provider's network is more expensive than using tunneling protocols. Given the VPN Consortium's support for IPSec, most traditional VPNs deployed today are IPSec-based. As a result, we will compare MPLS-based VPNs to IPSec-based VPNs.

IPSec VPN An IPSec-based VPN relies on tunneling and encryption for secure end-to-end transmissions. When an end node transmits a packet to its VPN gateway (e.g., see Figure 7.16), the gateway encrypts and encapsulates the packet before transmitting it to the network service provider. This newly formed packet is then transmitted to the provider's network where it is ultimately transported to the destination site. As the packet traverses the service provider's network en route to the destination node, point-to-point tunnels are established between the intermediate nodes. Thus, the only access to the tunnel is at one of the end points.

IPSec VPNs operate at TCP/IP's network layer and can be deployed across any IP network, including the Internet. This makes them an excellent choice for securing remote access across an untrusted public network such as the Internet and are attractive to Internet service providers and businesses alike. Furthermore, as an IETF standard, interoperability issues have been greatly reduced enabling customers to select VPN products from different vendors. IPSec-based VPNs also work best at the outer edge of a service provider's network. (*Note:* The gateways along these edges are commonly referred to as *edge routers* because they are at the edge or outer regions of the network.) Data along the outer edge are usually more susceptible to security threats than data within the interior part or core of the provider's network.

Although IPSec-based VPNs provide a highly secure infrastructure, they still have their shortcomings. For example, because they rely on tunneling and encryption, IPSec VPNs are subject to high delays (i.e., latency) because it takes time to encrypt and encapsulate a packet. Furthermore, once an IPSec packet reaches the service provider's network, additional delay is possible if the packet needs to be fragmented so that its length is commensurate with the provider network's maximum transmission unit (MTU). There is also the issue of scalability. As noted earlier, it is both expensive and formidable to provision new links and services to keep up with customers' as noted earlier demands for adding new sites and service requirements to their VPNs.

MPLS VPN The components of an MPLS network consist of two types of routers: Edge routers—called *label edge routers* (LERs)—which connect to routers located at customers' sites, and *label switch routers* (LSRs), which form the core of the provider's MPLS network. By default, LSRs are configured in a fully-meshed topology (see Figure 7.17).

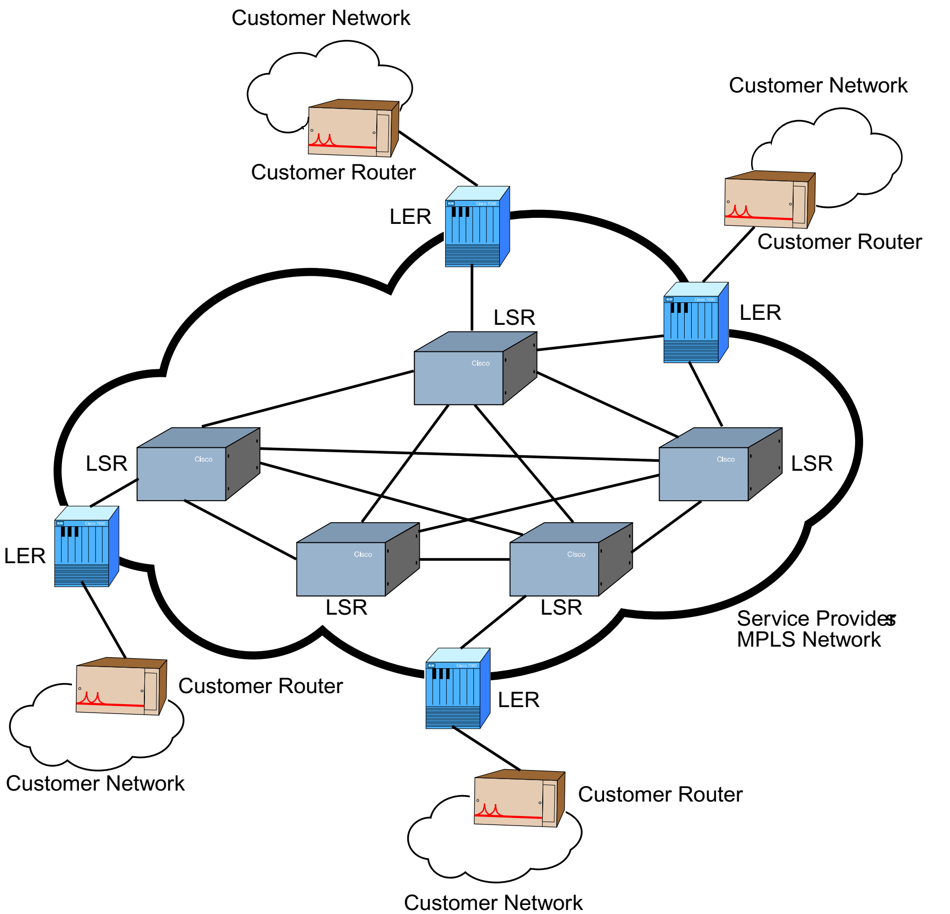


FIGURE 7.17 An MPLS network consists of label edge routers (LERs) that connect to customer routers at the edge or outer part of the provider's network, and label switch routers (LSRs) that form a fully-meshed topology within the core of the provider's network. Packets first enter the network via the LER that connects to the customer's router. This ingress router assigns a label to the packet and transmits it to an LSR. The packet is then transported across the network in a hop-by-hop fashion using label information instead of IP addresses. Source: Adapted from Staats, 2001 and Welcher, 2000.

In contrast to an IPSec VPN, an MPLS-based VPN uses neither tunneling nor encryption. Instead, it relies on "labels," which are small packet identifiers that provide (among others) addressing information about the packet. Incoming packets from a customer's site are first assigned a label by an LER. Since this is the point where a packet first enters the MPLS network, this LER is called the ingress router. The label assigned to the packet contains information that identifies the VPN as well as information about the path the packet is to traverse. Packets are then forwarded throughout the provider's MPLS network along a *label switch path* (LSP) comprising of LSRs. These LSRs make forwarding decisions based solely on the packet label's contents. At each hop, the LSR strips off the existing label and applies a new label that informs the LSR at the next hop how to forward the packet. When the packet arrives at the egress router (i.e., the LER that connects the destination site to the MPLS network), the router removes the path information from the label. In IP networks, the bytes corresponding to an MPLS label are inserted prior to the IP header; in frame relay networks (Chapter 12), the label is the data link connection identifier (DLCI); and in ATM networks (Chapter 14), the label is the virtual path and channel identifiers (VPI/VCI).

The advantages of an MPLS strategy are many. For example, the customer's router only needs to establish a peering relationship with the provider's LER instead of with all the routers that comprise a particular tunnel (see for example Figure 7.7b). This simplifies local management of a VPN because a customer can use the provider's LER as the respective default router for all of its sites. Furthermore, instead of running special MPLS routing software, the customer's routers can use static routing or RIP. Another advantage is that in addition to address information, labels also carry traffic prioritization information, which gives MPLS VPNs QoS capabilities. Latency is also minimized because packets do not have to be encrypted or encapsulated. Encryption is not necessary because each MPLS VPN effectively creates a private network, This is similar to the manner in which frame relay networks provide security (see Chapter 12). MPLS networks are also highly scalable because the core of the network is fully-meshed, which implies that no new tunnels have to be established when customers add new sites. If you would like additional information about MPLS, the MPLS Resource Center (http://www.mplsrc.com) is a good starting point. To learn more about the VPN marketplace, see http://vpdn.com.

49. Thanks for the references. One last question: Is MPLS a standard?

MPLS, per se, is not a standard. Instead, it is a technology that comprises a collection of protocol standards or soon-to-be standards. For example the first IETF standard in this collection is RFC 303, "Multiprotocol Label Switching Architecture." The priority labeling function that provides traffic prioritization information is another protocol that is being standardized by IETF. To give you an idea of the scope of activity related to MPLS, see http://www.mplsrc.com/drafts.shtml.

END-OF-CHAPTER COMMENTARY

In this chapter we examined many aspects related to layer 3 of the OSI model. Several of the topics or concepts developed here are further explored in other chapters. For example, telephone circuits are revisited in Chapter 15 as part of a discussion on modems. Several WAN technologies and services are also discussed in subsequent chapters, including ISDN (Chapter 11), frame relay (Chapter 12), SMDS, (Chapter 13), and ATM (Chapter 14). Finally, in the area of VPNs, we examine several different encryption strategies in Chapter 16. You are also encouraged to revisit key concepts that were discussed in previous chapters, including QoS (Chapter 5) as well as IP and the network layer (Chapter 3).

Chapter 8

Ethernet, Ethernet, and More Ethernet

In this chapter, we focus our attention on Ethernet networks. In previous chapters, we informally examined aspects of Ethernet technology to illustrate various network concepts. For example, in our discussion of the data link layer in Chapters 5 and 6, Ethernet was used as an example to demonstrate frame formats, MAC hardware addresses, and network switches. In this chapter the subject of Ethernet is presented in a more formal and comprehensive manner. An outline of the topics and concepts we discuss follows:

- History of Ethernet (Questions 1–3)
- Ethernet vs. IEEE 802.3 (Questions 4–13)
- The Initial IEEE 802.3 Protocol: 10 Mbps Ethernet (Questions 14–17)
- 10 Mbps Ethernet Performance Issues: Network Diameter and Collision Domain (Questions 17–28)
- Partitioning, Switched Ethernet, and Virtual LANs (Questions 29–40)
- 100 Mbps Ethernet: Fast Ethernet and 100VG-AnyLAN (Questions 41–57)
- 1 Gigabit Ethernet (Questions 58–71)
- 10 Gigabit Ethernet (Questions 72–77)
- IsoEthernet (Question 78)

1. What exactly is Ethernet and what is its origin?

Ethernet is a local area network protocol developed jointly by Xerox, Intel, and Digital Equipment Corporation (DEC) at the Xerox Palo Alto Research Center (PARC) in the mid-1970s. It was designed as a technology that would allow for the interconnection of office devices. Although the concept of Ethernet was originally developed at PARC, Ethernet's genesis is with Norman Abramson of the University of Hawaii in the late 1960s to early 1970s. Abramson developed a network called ALOHA, which was used to connect the main campus site in Oahu to seven other campuses on four of the Hawaiian islands. Using a technique called *contention*, Abramson demonstrated that multiple nodes on a network could use the same channel for communications and they could send data whenever they had data to send. The primary difference between the ALOHA network and Ethernet is that ALOHA permitted any node to transmit data at any time, made no provision to allow a node to detect if another node was sending data, and there was no procedure for dealing with what would come to be known as *collisions*. Collisions occur when two or more nodes attempt to transmit data simultaneously. Without a mechanism for dealing with the eventuality of simultaneous transmissions, ALOHA required many retransmissions. Ethernet, on the other hand, was designed with both carrier sense capability (CSMA) and collision detection (CD) (see Chapter 5).

2. How did Ethernet get its name?

The name Ethernet was derived from the old electromagnetic theoretical substance called *luminiferous ether*, which was formerly believed to be the invisible universal element that bound together the entire universe and all its associated parts. Thus, an "ether" net is a network that connects all components attached to the "net."

3. Is Ethernet a standard?

Yes. Through a consortium organized in the early 1980s, Xerox, Intel, and DEC published a vendor standard now known as the *Ethernet Blue Book*. This book described the methods in which Ethernet would be developed and implemented. It also described how Ethernet hardware and data link services would function. Work on this standard continued to evolve, culminating in the 1982 publication of a cooperative standard titled *Ethernet Version 2.0*.

4. So what happened? Why is there Ethernet and IEEE 802.3?

Although the Xerox-Intel-DEC consortium developed and produced an Ethernet standard (V2.0), it was not an acceptable domestic or international standard for LAN technology. Consequently, the IEEE formed subcommittee 802.3 and produced an IEEE standard for a technology very similar to the Ethernet V2.0 specification. Rumor has it that the IEEE wanted to make Ethernet V2.0 its standard but the Xerox-Intel-DEC consortium wanted to maintain the patent. To avoid any type of patent infringement, the IEEE modified Ethernet V2.0 and produced the IEEE 802.3 specification. These patents have since been given to the IEEE and anyone can now license Ethernet from IEEE for a flat fee of $1000. Due to its influence with U.S. and international standardization authorities, IEEE 802.3 eventually became ISO standard IS88023.

5. How different is IEEE 802.3 from Ethernet V2.0?

The two Ethernet specifications are similar because IEEE used the technological details of Ethernet V2.0 as a basis for the 802.3 standard. However, several serious technical differences were introduced in the IEEE version that make the two standards incompatible. These included differences in cable size, transceiver function, frame formats, and topology.

6. In what way are the cables different?

In the V2.0 standard, the Ethernet cable (thick coax) is prescribed with a 0.395-inch diameter. In the IEEE specification, the cable's diameter was increased to 0.405-inch. The

IEEE's rationale for increasing the cable diameter was that the larger diameter provided better electrical characteristics. The only problem a larger diameter cable presented was that V2.0 compliant transceivers could not be used on IEEE 802.3 compliant cable. This presently is not a serious issue because a "thick" Ethernet network is something you are more apt to inherit than install. Vendors manufacture transceivers capable of connecting to both V2.0 and IEEE 802.3 cables.

7. OK. So cable differences are no big deal. What about transceiver differences?

Recall from Chapter 6 that a transceiver enables a node to communicate with the cable. Specifically, it transmits and receives simultaneously, and it notifies a node if a collision occurred. When a transceiver is connected to the cable there is usually no way to determine if the transceiver is working unless there is a data transmission. In the V2.0 specification, a signal known as *signal quality error* (SQE) is periodically generated by the transceiver and read by the controller of the host to which it is connected. Historically, this is called *heartbeat* and effectively informs the host's controller that the transceiver is "alive." In the IEEE 802.3 standard, transceivers do not generate a heartbeat unless a real signal quality error occurs. Thus, SQE effectively is used for network management. Given this difference in the operation of V2.0 and 802.3 transceivers, if a V2.0 controller is mated with an IEEE 802.3 transceiver, the controller interprets the absence of a heartbeat from the transceiver as a "dead" transceiver. To accommodate both standards vendors incorporated a switch into their transceivers that can enable or disable SQE.

8. Once again, no big deal. In what way are the two frame formats different?

Here is where your string of "no big deal" is broken. The difference in frame formats is a very big deal because it affects how data frames are formatted at the data link layer. Although V2.0 and IEEE 802.3 frames range from a minimum of 64 bytes to a maximum of 1518 bytes (see Chapter 5), they do have one major difference: V2.0 frames contain a two-byte "type" field, which is used to identify the different higher-level protocol types used by DEC, Intel, and Xerox (e.g., IP, IPX). The IEEE 802.3 specification does not support a protocol type field. Instead, the V2.0 "type" field is replaced with a "length" field, which specifies the length in bytes of the bit string that represents user data (see Figure 5.3.) The exclusion of a "type" field in the 802.3 standard means that compatibility cannot be maintained between 802.3 and V2.0. Unlike the previous two differences of cable diameter and SQE function, nothing can be done to compensate for the different frame formats.

9. If there isn't a "type" field in an IEEE 802.3 frame, then how does the network layer know what network protocol is being used for the frame?

This information is provided by the *logical link control* (LLC). Recall from Chapter 5 that the IEEE partitions the OSI data link layer into the MAC and LLC sublayers. In an 802.3 frame, an LLC header containing higher-level protocol information is provided at the beginning of the frame's data field. The combination of LLC header and data field is known as the LLC protocol data unit (LLCPDU). Ethernet V2.0 frames do not have an

LLC component. (*Note:* For completeness sake, the LLC sublayer also contains a source service access point, SSAP, and a destination service access point, DSAP. These SAPs provide the mechanism for source and destination nodes to communicate and are needed for protocol-type identification. See the Internet's Request for Comment, RFC 1340, for additional information.)

10. Are there any other differences?

Yes. Two other differences between V2.0 and 802.3 are topology and cable type. IEEE 802.3 supports both bus and star topologies, but Ethernet V2.0 supports only a bus topology. Finally, 802.3 compliant networks can be either baseband or broadband, but in V2.0 only baseband Ethernet networks are supported.

11. Is IEEE 802.3 considered superior to Ethernet V2.0?

It depends on what you mean by superior. Rather than comment on whether one standard is better than another, we will say this: There is nothing intrinsically wrong with V2.0 other than it is a proprietary standard that does not comply with the prescribed ISO standard for Ethernet-like networks. Furthermore, although the two standards are similar—802.3 was designed after V2.0—they are different enough to be considered incompatible. This incompatibility resulted from political as well as technical issues and is best left to historians and analysts.

12. So what's the bottom line? Do I have to be concerned with V2.0?

The bottom line is that vendor support for IEEE 802.3 overshadows that for Ethernet V2.0; any new "Ethernet" network you install probably will be based on the IEEE 802.3 protocol. Nevertheless, there are still many "old" Ethernets in existence and it is possible that you might inherit one that incorporates both V2.0 and 802.3 compliant products.

13. How important is it to distinguish between V2.0 and 802.3 when discussing Ethernet networks?

In casual usage, IEEE 802.3 is commonly referred to as Ethernet. What you should realize, though, is that technically it is *not* Ethernet—only V2.0 is considered Ethernet. This is similar to the zero versus "oh" issue in mathematics when reading a number such as 206. In casual, everyday usage we say "two-'oh'-six." Technically, though, the number should be read as "two-zero-six" because zero is a number whereas "oh" is a letter. Nevertheless, in this book, to play it safe, we use the notation "Ethernet/802.3" when referring to Ethernet networks.

14. Let's move on to the 802.3 protocol.

OK. At the physical layer (Chapters 4 and 6), the IEEE 802.3 standard addresses issues such as cable type, cable length, and connector types. Ethernet/802.3's physical layer also encodes data prior to transmission using a technique called *Manchester encoding*, whose purpose is to ensure that the end of a transmission (carrier-sense failure) is properly detected. Manchester encoding differs from standard digital transmission in two ways. First, instead of "high" equaling 1 and "low" equaling 0, a timing interval is used to measure high-to-low transitions. Second, instead of the timed transmission period being "all high" or "all low" for either 1 or 0, a state transition is encoded into the transformation. Specifically, a 1 is sent as a half-time-period low followed by a half-time-period high, and a 0 is sent as a half-time-period high followed by a half-time-period low (Figure 8.1) Consequently, the end of the last bit transmitted is easily determined immediately following the transmission of the last bit. The main benefit of this approach is error recovery; part of the signal transition from high to low and from low to high can be clipped or distorted, and there is still "intelligence" in the timing interval to determine if the signal was rising or falling. This allows on-the-fly signal recovery and minimizes single bit errors.

At the data link layer (Chapters 5 and 6), the standard is based on 1-persistent CSMA/CD and uses a binary exponential backoff algorithm to calculate the wait time. The average wait time for a new packet is set to an arbitrary initial value. This value is then doubled each time a collision results when a transmission is attempted with the same packet. The Ethernet/802.3 data link layer also specifies a minimum frame size of 64 bytes and a max-

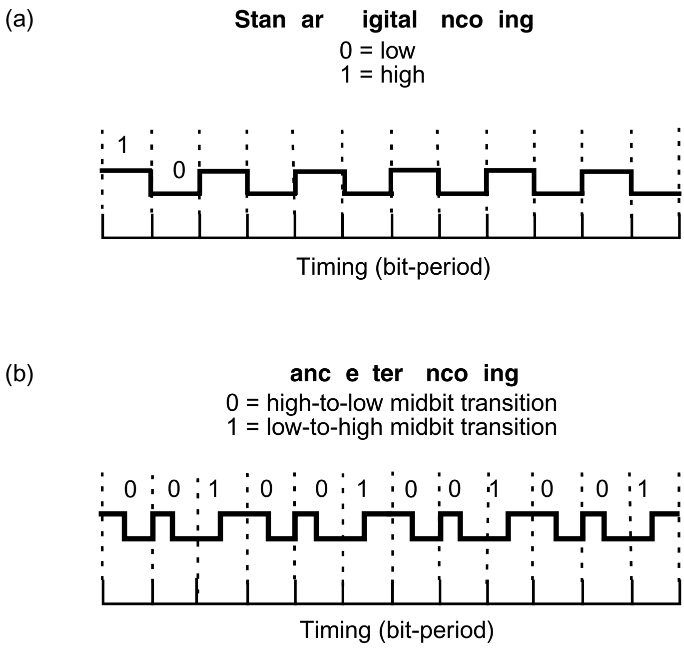


FIGURE 8.1 Part (a) shows a typical digital transmission. When an electrical pulse is "on," the line is at a high level; when there is no pulse during a transmission ("off" state), the line is at a low state. For each bit-period, there is either a pulse or no pulse. Part (b) shows a Manchester encoded digital transmission. Here, each bit-period has two intervals. A 1 is sent as a half-time-period "low" followed by a half-time-period "high," (i.e., a low-to-high midbit transition), and a 0 is sent as a half-time-period "high" followed by a half-time-period "low" (i.e., a high-to-low midbit transition). Instances where no transition occurs at the midpoint of a bit interval are used for control purposes.

imum frame size of 1518 bytes. The frame format consists of a 7-byte preamble, a 1-byte start frame delimiter, a 6-byte destination address, a 6-byte source address, a 2-byte length count, a data field that ranges between 46 and 1500 bytes, a pad field that ranges from 0 to as many bytes is necessary to get the data field to a minimum 46 bytes, and a 4-byte CRC checksum field (see Figure 5.3 and Chapter 5).

An Ethernet/802.3 address consists of 48 bits, represented as 12 hexadecimal digits, and partitioned into six groups of two. The higher-order three bytes (the leftmost six hexadecimal digits) represent the manufacturer of the Ethernet device; the lower-order three bytes represent the serial number of the device. For example, the address 08:00:20:01:D6:2A corresponds to an Ethernet/802.3 device manufactured by Sun Microsystems (as indicated by Sun's code 08:00:20) that has the serial number 01:D6:2A. The IEEE is responsible for assigning Ethernet addresses. (See Appendix A for additional information about Ethernet address prefixes. See also Box 5.2 for additional information about Ethernet/802.3 addresses.)

15. What do Ethernet designations like 10BASE-T mean?

IEEE 802.3 LANs are designated using the general format shown here as well as in Figure 8.2).

Signaling Rate (Mbps)—Band (Base or Broad)—Length (Meters) or Cable Type

Thus, a 10-Mbps baseband LAN that uses UTP cable is designated as *10BASE-T* (the T is for twisted-pair), and a 10-Mbps ThickWire baseband LAN is designated as *10BASE5* (the 5 is for 500-meter segment length). A brief description of the various IEEE 802.3 10-Mbps LAN specifications is given in Table 8.1. Examples of the various types are also provided in Figures 8.3 (ThickWire), 8.4 (ThinWire), 8.5 (UTP), and 8.6 (UTP). We suggest you study Table 8.1 and Figures 8.3 through 8.6 before proceeding further.

16. In Table 8.1, several attributes are given for each LAN type. For example, you list maximum number of nodes per segment, maximum segment length, and so forth. How significant are these?

Very significant. The IEEE 802.3 specifications for the maximum number of nodes per segment and maximum segment length, coupled with minimum and maximum frame sizes of 64 and 1518 bytes, respectively, are necessary due to Ethernet's contentious nature. The maximum number of nodes per segment ensures that an Ethernet network does not get saturated; the minimum packet size and maximum cable-segment length enable Ethernet

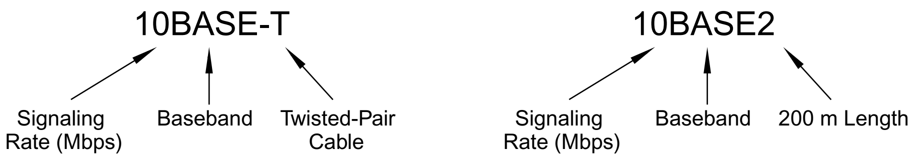


FIGURE 8.2 Baseband and broadband IEEE 802.3 LANs are designated by bandwidth in Mbps followed by either the type of cable it supports or its maximum length in meters.

TABLE 8.1 Summary of IEEE 802.3 Specifications for 10 Mbps Ethernet

Type	Description[a]
10BASE5 ThickWire	**Cable:** Thick Coax (RG-8); **Topology:** Bus; **Connectors:** Transceivers, transceiver cable, 15-pin AUI; uses "vampire" tap; **Maximum Segment Length:** 500 m; **Maximum Nodes per Segment:** 100—spaced in 2.5 m increments; **Maximum Diameter:** 2500 m; **Other:** 50-ohm termination at each end of cable; one end grounded to building ground
10BASE2 ThinWire	**Cable:** Thin Coax (RG-58); **Topology:** Bus; **Connectors:** BNC; **Maximum Segment Length:**185 m; **Maximum Nodes per Segment:** 30—minimum 0.5 m between nodes; **Maximum Diameter:** 925 m; **Other:** 50-ohm termination at each end of cable; one end grounded to building ground
10BASE-T UTP Ethernet	**Cable:** Category 3, 4, or 5 UTP; **Topology:** Star; **Connectors:** RJ-45, patch panels, repeaters; **Maximum Segment Length:** 100 m; **Maximum Nodes per Segment:** 2; **Maximum Diameter:** 500 m; **Other:** Each node is connected directly or indirectly to a hub; indirect connections are via wallplates or patch panels
10BASE-FB[b] Fiber Backbone	**Cable:** Fiber; **Topology:** Point-to-point; **Connectors:** Fiber-optic transceivers, ST; **Maximum Segment Length:** 2000 m; **Maximum Nodes per Segment:** 2; **Maximum Diameter:** 2500 m; **Other:** Backbone-only technology used to interconnect Ethernet repeaters; maximum of 15 repeaters permitted; uses synchronous signaling to retime the optical signals for data transmissions
10BASE-FL[b] Fiber Link	**Cable:** Fiber; **Topology:** Point-to-point or star; **Connectors:** Fiber-optic transceivers, ST; **Maximum Segment Length:** 2000 m; **Maximum Nodes per Segment:** 2; **Maximum Diameter:** 2500 m; **Other:** Can be used to interconnect workstations or repeaters; maximum of five repeaters permitted; replaces fiber-optic interrepeater link (FOIRL); if 10BASE-FL is mixed with FOIRL, max segment length is 1000 m
10BASE-FP[b] Fiber Passive	**Cable:** Fiber; **Topology:** Star; **Connectors:** Fiber-optic transceivers, ST; **Maximum Segment Length:** 500 m; **Maximum Nodes per Segment:** 33; **Maximum Diameter:** 2500 m; **Other:** Used for small installations such as workgroup LANs; specifies a passive hub, which means it uses no electronics (including power) and hence is immune to external noise

[a] All types subscribe to the 5-4-3 repeater placement rule.

[b] Part of the 10BASE-F Standard for fiber-optic Ethernet.

nodes to accurately detect collisions on the network; and the maximum frame size allows nodes to detect the completion of a transmission, which facilitates error detection.

17. Table 8.1 also specifies a maximum diameter as part of a network's description. For example, 10BASE5 has a maximum diameter of 2500 m, and 10BASE-T has a maximum diameter of 500 m. What is a network "diameter?"

Diameter in this context is a term that emanates from the branch of mathematics called graph theory, which is frequently employed by theoretical and applied computer scientists in the study of network design. The diameter of a 10-Mbps Ethernet/802.3 network is the

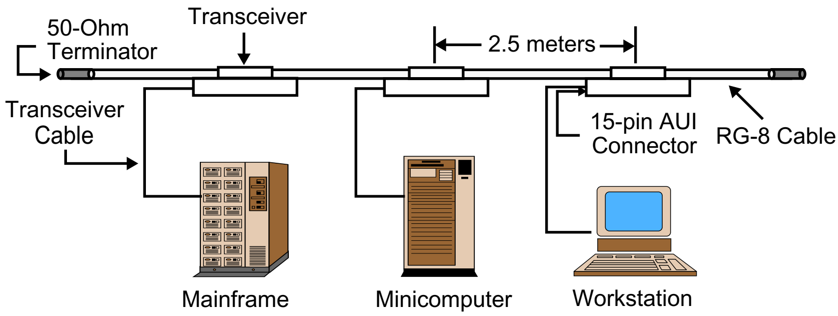


FIGURE 8.3 Example of a 10BASE5 (ThickWire Ethernet) LAN. The topology is a bus configuration, the maximum length of a single segment is 500, and a single segment supports up to 100 nodes. Nodes are connected to the cable via transceivers and transceiver cable; transceivers are spaced 2.5 m apart to prevent signal interference. The actual physical connection of a transceiver involves drilling into the cable using a "vampire tap." Each end of the cable is terminated with a 50-ohm resistor, and one end of the cable must be grounded.

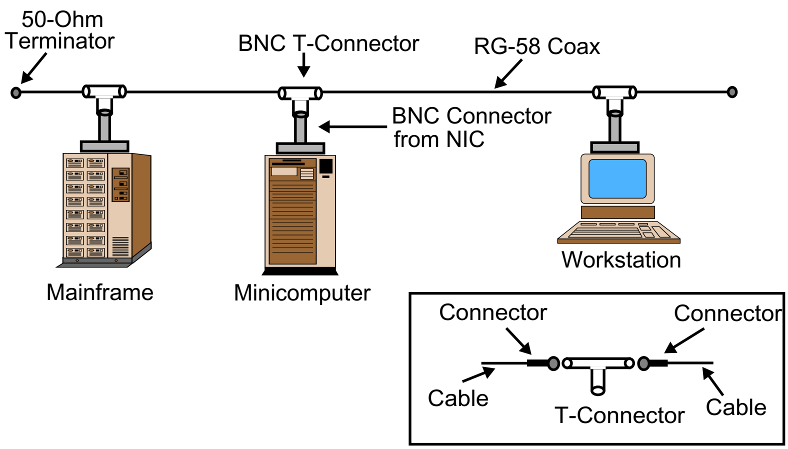


FIGURE 8.4 Example of a 10BASE2 (ThinWire Ethernet) LAN. The topology is a bus configuration, the maximum length of a single segment is 185 m, and a single segment supports up to 30 nodes. Nodes are connected to the cable via BNC T connectors, which must be spaced at least 0.5 m apart to prevent signal interference. Each end of the cable is terminated with a 50-ohm resistor, and one end of the cable must be grounded. A segment is composed of several pieces of cable, with each piece being connected via a T connector (see inset).

overall length between the network's two most remote nodes. For example, consider the 10BASE-T LANs shown in Figures 8.7 and 8.8. The first one has a network diameter of 200 m; the second LAN's diameter is 300 m.

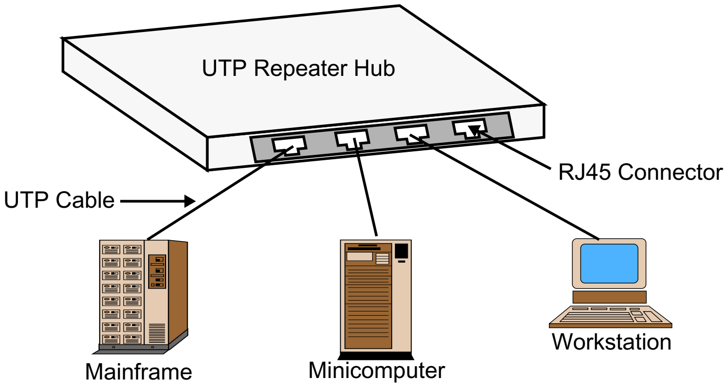


FIGURE 8.5 Example of a typical 10BASE-T (Twisted-Pair Ethernet) LAN. The topology is a star configuration, and the maximum length of a segment is 100 m. Note that only two nodes can be connected per segment—a workstation and a repeater. Both the hub and a device's network interface card (NIC) contain eight-pin modular (RJ-45) connectors. Cable can be Category 3, 4, or 5; higher grade cables provide better performance.

18. Why are the maximum diameters different for the various LAN types?

The maximum diameter of a 10 Mbps Ethernet/802.3 is the maximum distance possible for a *single* Ethernet/802.3 LAN. By *single* we mean an Ethernet/802.3 LAN that consists of either one segment or multiple segments connected by repeaters. Maximum diameters vary among LAN types because they use different cable. For example, the network diameter of a 10BASE5 LAN, which uses RG-8 coaxial cable, ranges from a minimum of 500 meters (for one segment) to a maximum of 2,500 meters (five segments connected by four repeaters). Similarly, a 10BASE-T LAN uses unshielded twisted-pair cable and has a network diameter that ranges from 200 m to 500 m (see Figure 8.9).

19. In Figure 8.9, you mentioned the 5-4-3 repeater placement rule. What is this rule?

In order for a network to function properly, it must comply with its design rules. The *5-4-3 repeater placement rule* is a general rule of thumb to follow when configuring an Ethernet/802.3 LAN to ensure that it follows IEEE specifications. Ethernet/802.3 LANs are based on CSMA/CD (see Chapter 5). Thus, collisions are an inherent part of an Ethernet/802.3 LAN. If a LAN consists of a single segment (Figures 8.3 and 8.4), or a single UTP repeater hub (Figures 8.5, 8.6, and 8.7), then the 5-4-3 repeater placement rule is of no practical concern. However, if a LAN's diameter is going to be extended using repeaters, then the 5-4-3 repeater replacement rule becomes extremely critical. You see, every Ethernet/802.3 LAN has what is known as a *collision domain*. A collision domain consists of a single network where two nodes can cause a collision. In the case of a single-segmented Ethernet/802.3 LAN, the independent segment represents the collision domain; in a multisegmented Ethernet/802.3 LAN, however, the collective segments comprise the collision domain. For example, in Figures 8.3 and 8.4, the collision domain comprises the

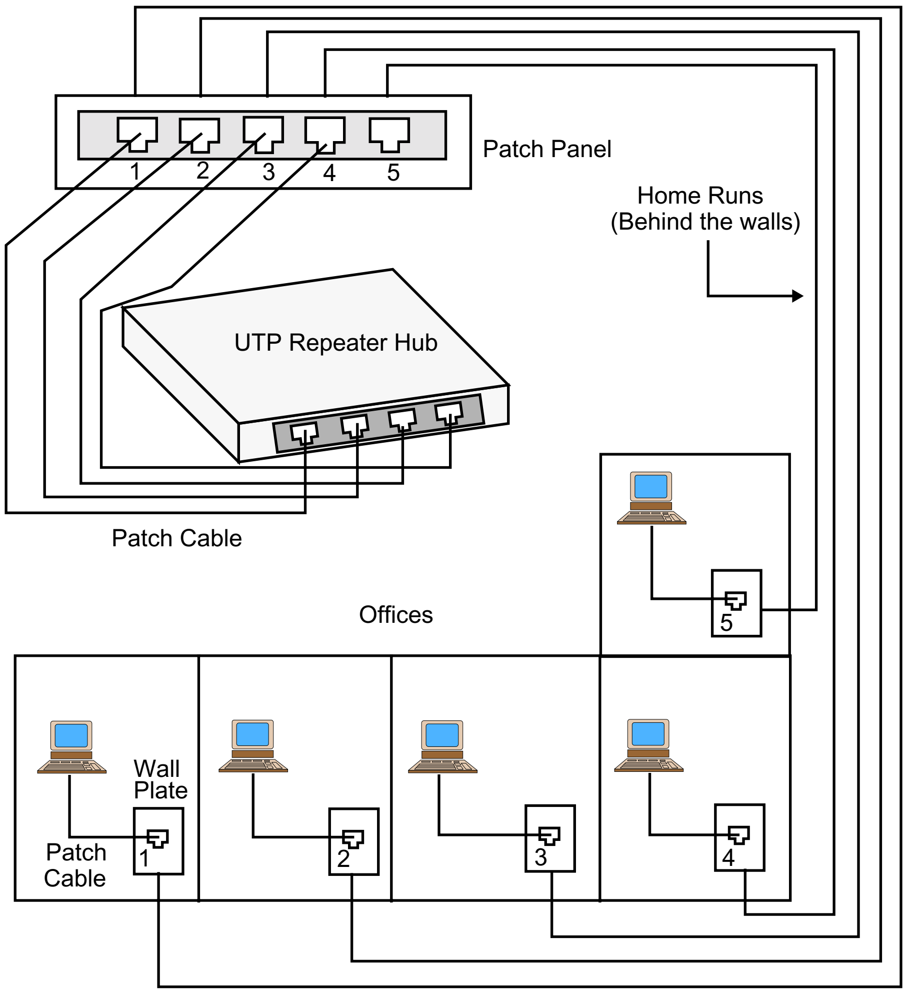


FIGURE 8.6 A typical 10BASE-T wiring scheme involves a centralized hub located in a wiring closet and cable installed between a patch panel and a wall plate located in an office or room. This length of cable is referred to as a "home run." Each end of these home runs is then "punched down" behind the patch panel and wall plates. Patch cable is used to connect ports on the patch panel to the hub and workstations to wall plates. The total length of all cable for one workstation (hub to patch panel, home run, and patch cable from wall plate to NIC) must be no more than 100 m.

Although patch panels provide a certain level of convenience, they also represent a potential source of additional noise. Nevertheless, patch panels are useful in many situations. For example, note that patch panel port 5 is wired to office 5, but the workstation in office 5 does not have a connection to the LAN because patch panel port 5 does not have a connection to the hub. Connectivity can be provided to office 5, though, simply by disconnecting a patch cable from one of the patch panel ports and reconnecting it to port 5.

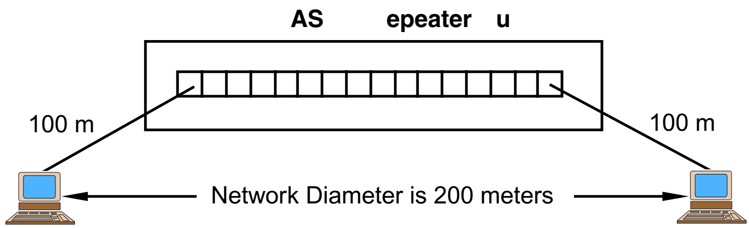


FIGURE 8.7 A network diameter is the distance between the two remotest nodes of a network. In this illustration, we have a 10BASE-T LAN that can connect up to 16 nodes via a UTP repeater hub. If each node is connected to the hub with the maximum segment length permitted (100 m), then the network diameter is 200 m—100 m from sending node to repeater port plus 100 m from repeater port to receiving node.

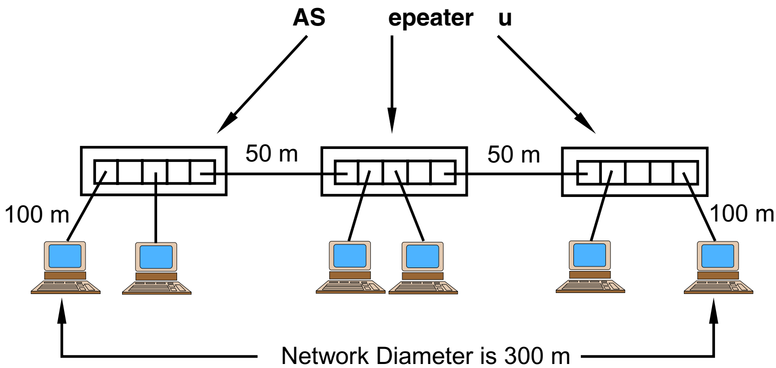


FIGURE 8.8 In this illustration, a 10BASE-T LAN consists of four segments interconnected by three repeaters. If the links between nodes and their respective ports are 100 m each, then the network diameter is 300 m—100 m from sending node to port, plus two 50-m hub links, plus 100 m from port to receiving node. Remember: The network diameter is always the distance between the farthest two nodes.

single ThickWire or ThinWire segment. In Figures 8.5, 8.6, and 8.7, the collision domain comprises the segments connected to the single repeater. In Figures 8.8 and 8.9, though, multiple segments are interconnected via multiple repeaters. This extends the network's diameter, which increases the distance between the two most remote nodes. Physically, the networks in Figures 8.8 and 8.9 appear to be separate. Electrically, though, these physically separate networks belong to one collision domain. The bottom line is a single collision domain contains a maximum of 1024 end nodes, and its diameter cannot be more than 2500 m.

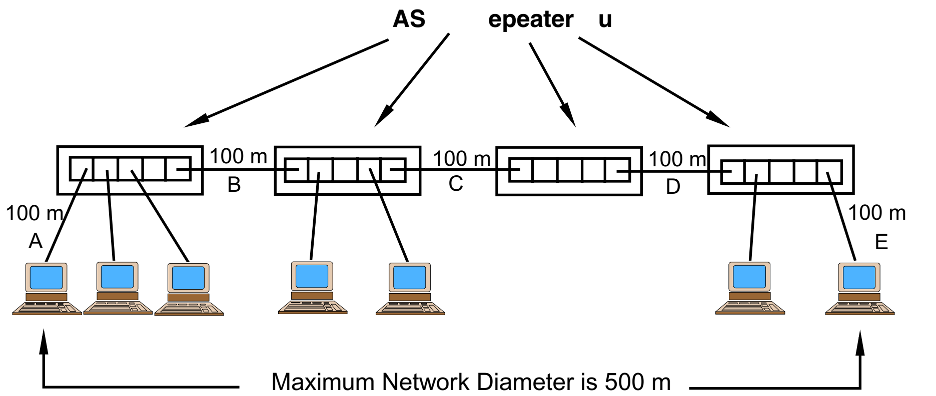


Figure 8.9 The maximum diameter of a 10BASE-T LAN is 500 m. Note how this configuration complies with the 5-4-3 repeater placement rule. There are 5 segments (labeled A through E), four repeaters, and at most only three segments populated with end nodes.

20. What does all of this have to do with the 5-4-3 repeater placement rule?

We're getting there. We want to first set the stage by extending our discussion about collision domains. CSMA/CD design rules specify that if a node's transmission results in a collision, then the node must be able to detect the collision before it stops transmitting. If not, the node will never know that the frame it transmitted was corrupted. For example, look at Figure 8.9 and consider the following scenario involving the nodes at the opposite ends of the collision domain (i.e., the workstations attached to segments *A* and *E*, respectively):

- Node *A* transmits a frame.
- Around the same time, node *E* wants to transmit a frame. It senses the channel but does not "hear" anything because *A*'s transmission has not reached *E*. Sensing an idle channel, *E* transmits a frame.
- The two frames collide.
- *E* must now send a jamming signal to all the nodes connected to the network, including *A*. Furthermore, the jamming signal must reach *A* before *A* stops transmitting. If the collision occurred at *E*'s "front door," that is, just before *A*'s frame reached *E*, the jamming signal must travel a considerable distance.

On an Ethernet/802.3 LAN, timing is everything. Although electrical signals on Ethernet media travel at nearly the speed of light, it still requires a specific amount of time for a signal to travel from one node to another across the network. In Ethernet/802.3 jargon, time is measured using the unit *bit-time*, which is equal to 0.1 μs. Thus, a one bit transmission requires 0.1 μs. Recall that the smallest Ethernet/802.3 frame is 64 bytes long, which is 512 bits. This implies that to transmit a 64-byte frame requires 512 bit-times or 51.2 μs. This means that a 64-byte frame has a collision domain of 512 bit-times or 51.2 μs. Thus, any delay cannot be greater than 51.2 μs. Returning to our scenario above, in order to maintain a round-trip delay of no more than 512 bit-times, node *E* has to send a jamming signal before *A* transmits more than 256 bits (i.e., 256 bit-times). This will then give *E* 256 bit-times for its jamming signal to reach *A*. If the LAN is up to spec, then the round-trip delay will not exceed 512 bit-times.

21. Oh! I get it. And because repeaters introduce delay, we have to limit how many are connected to the network so we can maintain that 512 bit-time upper limit.

That's right. Repeaters—like other electronic devices such as cables, NICs, bridges, and switches—add to the overall delay of a network. Thus, it is important that we understand what impact this delay will have on the overall performance of our LAN. Delay, which is also called *latency*, is measured by how long it takes the repeater to transmit a regenerated signal after receiving it (i.e., the amount of time the signal remains "inside" the box). Since the amount of delay introduced by repeaters varies from one vendor to another, the 5-4-3 repeater placement rule gives us a general rule of thumb to follow if you do not want to calculate the total delay yourself. The 5-4-3 rule requires: no more than 5 segments of up to 500 m each; no more than 4 repeaters; and no more than 3 segments can have end nodes connected to them. This rule is also known as the 4-repeater rule, or the 5-4-3-2-1 rule. In the latter, the "2" implies that two of the five segments are used as interrepeater links, and the "1" implies that a configuration using the maximum parameters permitted results into one collision domain.

22. I am curious as to how would you calculate the total delay.

Easy: (a) get the vendor latency specifications for all the repeaters, NICs and cable used on your network; (b) add the delays; (c) multiply the sum by two (you want round-trip values); and (d) see if the product is less than or equal to 51.2 μs. If it isn't, then your network is not Ethernet/802.3 compliant and could have problems. It is usually a lot easier to just follow the 5-4-3 repeater placement rule.

23. What kind of problems?

Some problems manifest themselves as bad or invalid frames. These include oversized frames, runt frames, jabbers, alignment or frame errors, and CRC errors; all were discussed in Chapter 5. Timing errors can also lead to retransmissions. For example, a sending node that does not receive an acknowledgment from the receiving node within a prescribed time period assumes the frame was lost and retransmits the frame. Continual retransmissions can ultimately lead to degradation of network performance. Probably the biggest problem with Ethernet networks is collisions. Box 8.1 contains a description of some common causes of collisions, and Box 8.2 addresses Ethernet performance issues.

24. Is there any way to get around the 5-4-3 repeater placement rule?

Yes. One way to increase the number of Ethernet ports and still comply with the 5-4-3 repeater placement rule is to use chassis-based repeater hubs (see Figure 6.8). These hubs have individual slots in which multiport Ethernet interface boards (also called *blades, cards,* or *modules*) are installed. The primary feature of a chassis hub is that each board is

BOX 8.1: Common Causes of Collisions

Common causes of collisions in Ethernet/802.3 include the following:

Propagation Delay. Different types of media have different propagation delays, as do repeaters and bridges. If nodes are far apart on long segments, there is a greater chance of collisions due to propagation delay, especially in high traffic environments. Add repeaters or bridges to the configuration and the probability increases. Thus, the length of certain types of Ethernet/802.3 media can have dramatic effect on whether collisions occur.

Nodes Not Following The Rules. The IEEE specifications establish specific rules for collision detection and retransmission. Unfortunately, not all vendors follow these rules. When a vendor violates the rules on collision detection, it can affect the performance of the network. The 802.3 standard specifies that after a collision, retransmission of a packet should occur after generation of a random amount of timer delay not to exceed 1024 slot-times, which occurs on the tenth to fifteenth consecutive collision. (One slot-time is defined as 51.2 μs for 10 Mbps cable plants.) This means that if a collision occurs, a jamming signal is sent, a random number is generated, and the controller waits that long before retransmission. The controller then seizes the cable (after sensing it) and sends the packet again (if the cable is idle). Although the wait interval is supposed to be between 1 and 1024, some vendors violate this rule and set a ceiling lower than 1024; doing so does not allow random numbers higher than a predetermined value, thus allowing a system to acquire the network quicker than those that generate a higher random number in accordance with the IEEE standard.

Noise. Noise is a pretty obvious source of collisions. Recall that noise is any type of undesirable signal. Noise can come from a variety of locations including external sources or harmonic distortion. Various everyday office equipment such as copiers, laser printers, ballast transformers on fluorescent tube lighting, and HVAC motors also causes noise problems. Noise is also more problematic with UTP cable.

Improper Segmentation of Cable. This reason is restricted to coaxial cables. ThickWire cable should be cut in accordance with the standard and at specific lengths (e.g., 23.4 m, 70.2 m, etc.); ThinWire networks should have at least 0.5 m distance between nodes. Improper segmentation can cause noise and harmonic distortion problems, thus increasing the likelihood of collisions.

Babbling Transceivers. When a transceiver fails it begins to spew all kinds of trash on a cable; collisions inevitably occur.

connected to the same backplane to which the repeater unit is connected. Since all boards share the same backplane with the repeater, all of the devices connected to the hub use only one repeater.

Another workaround is stackable repeater hubs (see Figure 6.9). These devices consist of individual hubs "stacked" one on top of another. Instead of a common chassis backplane,

BOX 8.2: Ethernet/802.3 Performance Issues

Many Ethernet/802.3 performance-related problems stem from poorly designed and installed Ethernets, systems that are configured incorrectly, and standards violations (e.g., incorrect cable length, too many devices connected to a segment, etc.). Following are some additional problems that can adversely affect an Ethernet/802.3 LAN's performance:

Frame Deferrals. When a host is ready to transmit a frame but the network is busy, a frame deferral occurs. Thus, the transmitting node must defer or wait until the network is idle. If the average load is high (over 30%), then deferrals might be normal. If, on the other hand, network load is light, there are few errors, and frame deferrals are present, then a high burst rate contention most likely exists on the network. As a general rule, frame deferrals should never exceed 10% of the transmitted frames of a given system.

Collisions. High traffic loads for short bursts of time on nodes that are close (electrically speaking for propagation delay) do not necessarily translate into collisions. However, high traffic loads from bursty nodes that are electrically distant do tend to cause serious collision problems on the network.

Session Disconnects. Nodes that cannot communicate effectively with each other eventually time-out. Such a disconnect might be due to network congestion (due to bursty traffic), or the inability of the network to send traffic back to the node in a prescribed time-frame.

Congestion. Hardware controllers for Ethernet/802.3 devices have a finite amount of CPU power and a finite amount of memory on the cards. When a burst of traffic arrives, the controller must collect all data frames—regardless if the node is the correct recipient—before the controller logic can determine if the frames are valid for the node. A high burst rate of traffic can cause all buffers on a node to fill quickly, and can cause the controller to lose data destined for that node while collecting data destined for other systems. When a node cannot receive data because its buffers are full, the frames are lost. Eventually, data retransmission occurs resulting in increases in both traffic load and bursting rates.

Retransmissions. As nodes lose data destined for them, the data must be retransmitted. This causes additional bursts of traffic and an artificially inflated traffic level on the network. The single most common problem that results in retransmissions is due to controller congestion—the Ethernet controller on the receiving host is not capable of reading the frames on the cable fast enough to capture all the frames offered by the sending node the first time.

stackable hubs use a "pseudo-backplane" based on a common connector interface. An external cable interconnects the individual hubs in a daisy-chain. Once interconnected, the entire chain of hubs becomes a single logical unit that counts as only one repeater hub on the network. Stackable hubs are less expensive than chassis-based devices, and permit additional hubs to be added to the stack without any need for worrying about repeater hop counts. There is an upper limit, though, to the number of hubs that can be stacked. Known as the *stacking height*, this number is between 6 and 12, depending on the manufacturer.

25. Is there a similar rule for bridges, switches, and routers?

No. Repeaters are layer-1 devices that regenerate all incoming signals, including collisions, and propagate these regenerated signals to all the segments connected to its ports. Repeaters extend the diameter of a network, but are considered to be part of the same collision domain of networks designed using only repeaters. Bridges, switches, and routers, however, are layer-2 or later-3 devices. They perform filtering and frame translations (e.g., from an 802.3 format to an 802.5 format). They do not propagate collision signals from one segment to another. Hence, these devices effectively partition a network into multiple collision domains. For example, in Figure 8.9, if the third repeater from the left were a bridge, then there would be two separate collision domains. The first contains the first two repeaters, and the second contains the last repeater. (The network diameter remains the same.) Thus, if you intend to extend your network using repeaters, be certain to limit the number of repeaters connected to the network so you can maintain the 512 bit-time upper limit.

26. It would seem that as more nodes are added to an Ethernet/802.3 LAN, the more likely collisions will increase, which in turn degrades overall network performance. How can you tell when an Ethernet/802.3 LAN is overloaded and what can you do about it?

You are correct in your observation. Given its design, Ethernet/802.3 LAN performance could indeed degrade considerably as more nodes are connected. As for assessing whether an Ethernet/802.3 LAN is overloaded, two strategies are available. The first one follows a basic management principle: delegate it to someone—in this case, the users. If user complaints become persistent, frequent, and loud enough, then the network probably is overloaded. The second strategy is to approach the task of analysis scientifically—get the proper tools and training, and then systematically measure the LAN's performance.

27. If I were to use the scientific approach, what do I need to consider?

First, you need to understand that it is very difficult to measure the true performance of any Ethernet/802.3 LAN because it does not use fixed frame sizes. For example, an Ethernet/802.3 LAN that transmits only maximum-sized frames (1518 bytes) has a theoretical maximum efficiency rate of more than 95 percent. This means 95 percent of the LAN's transmission time is being used to transmit real user data. At the other end of the scale, a LAN that transmits only minimum-sized frames (64 bytes), where user data is only one byte plus 45 bytes of padding, has a theoretical maximum efficiency rate of less than two percent. So, the efficiency of an Ethernet/802.3 LAN is a function of the frame sizes transmitted.

A second concept of performance is *utilization*, which is the amount of time the LAN spends successfully transmitting data. Many performance monitoring tools will provide a user with average and peak utilization times, which are reported as a percentage. Both have different meanings. For example, an average utilization of 20 percent means that over some period of time (e.g., a 10-hour period), on average, 20 percent of the LAN's capacity is used for successfully transmitting data. On the other hand, a 20 percent peak utilization means that at a specific moment in time, 20 percent of the LAN's capacity was utilized. Associated with the concept of utilization is *throughput*, which is a measure of the amount of data transmitted between two nodes in a given time period (see Chapter 4). Throughput and utilization are the same except they use different units of measure. For example, if the average utilization of a 10-Mbps Ethernet/802.3 LAN is 20 percent, then this implies that 20 percent of the possible 10-Mbps bandwidth, 2 Mbps, is being used on average to successfully transmit data.

28. What are some acceptable parameters?

Now you are asking us to address an often-debated topic—the subject of "acceptable" Ethernet/802.3 LAN performance parameters. Rather than commit to specific parameters, we will give you some guidelines so you can decide what is acceptable. First, every LAN is different. For example, a 50 percent average utilization rate might be acceptable in one context but unacceptable in another. Some network managers believe that when average utilization exceeds 30 percent of a 10-Mbps Ethernet/802.3 LAN (3 Mbps), access times to the channel become unacceptable and overall network performance degrades. Hence, they set their thresholds accordingly and take action when they are reached. Second, don't get alarmed at high peak utilization rates. It is not uncommon, particularly during large downloads from a server, for an Ethernet/802.3 LAN to experience a peak utilization of 95 percent or higher. If this rate is sustained for a prolonged period of time (e.g., 5 or 10 minutes), then there probably is a problem. Third, be sensitive to response times. Increased response times could imply that the network is becoming saturated, which implies a sustained utilization rate of more than 80 percent. Note that the way CSMA/CD is designed, nodes even on a saturated LAN will eventually be serviced. However, the response time might not be acceptable. The bottom line is this: Although a 10-Mbps Ethernet/802.3 LAN has a theoretical capacity of 10 Mbps, actual utilization will always be less than this theoretical value. Furthermore, it is up to you to decide what parameters are acceptable for your LAN.

29. What strategies can I use to increase the efficiency of my Ethernet/802.3 LAN?

One strategy is to partition it. *Partitioning*, which is also called *segmentation*, involves configuring a network so that it consists of several separate (but still interconnected) segments. Partitioning improves overall network performance, enhances security, and increases reliability.

30. How is this done?

One way to partition a network is to create separate segments with fewer users. This is illustrated in Figure 8.10. In part (a) a typical Ethernet LAN consisting of nine hosts is shown. In part (b) this network is partitioned into three separate segments, each consisting of three hosts and a bridge. Segments are interconnected by a common backbone and isolated from each other using the bridges. An alternative to the configuration of (b) is shown

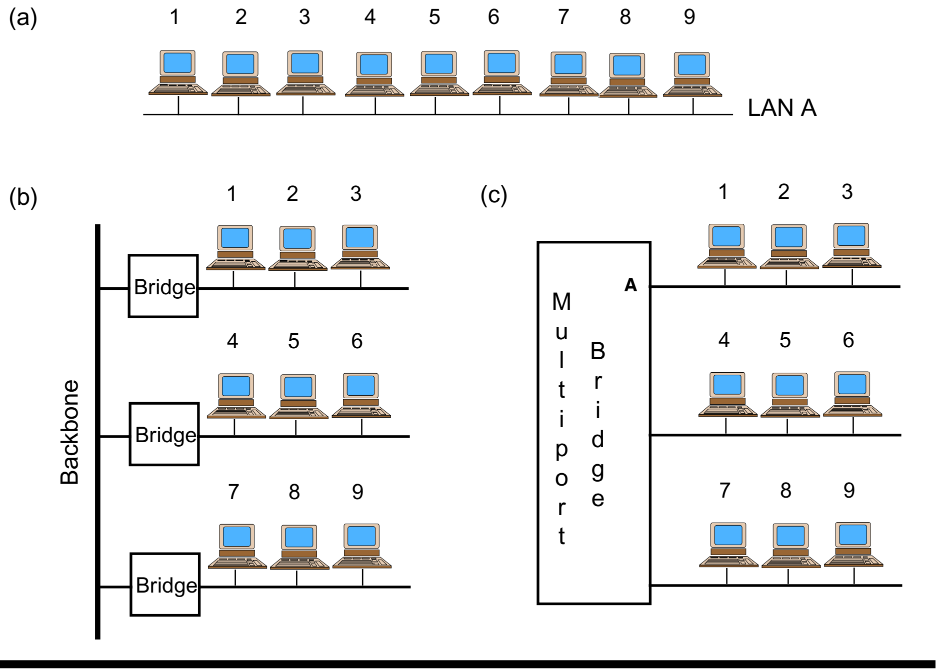


FIGURE 8.10 In (a), an unsegmented Ethernet LAN is shown. In (b), this LAN is partitioned into three separate segments using individual bridges connected to a backbone, and alternatively, in (c), a multiport bridge is used in which the backbone of (b) is "collapsed" into the multiport bridge. In a properly segmented network, at least 80% of the network traffic generated by the nodes on a LAN remains local to that LAN. Keeping remote access to a minimum (no more than 20%) minimizes backbone congestion and increases network performance.

in part (c). Once again the original Ethernet LAN of (a) is partitioned into three isolated segments, each consisting of three hosts. In this illustration, though, the segments are interconnected via a multiport bridge rather than a common backbone. Thus, the backbone of (b) has been "collapsed" into the multiport bridge. Comparing the configurations of (b) and (c) to that of (a) in Figure 8.10, it can be seen that segmentation helps reduce network traffic loads by reducing the number of nodes having to contend for the same shared medium. In part (a) nine hosts are in contention, but in (b) and (c) only three hosts each contend for the same medium. This strategy works well as long as it follows the 80/20 rule—80 percent of the traffic between nodes remains on the same physical cable segment; the remaining 20 percent of traffic traverses a layer-2 or layer-3 device. If not, then this configuration can actually degrade a network.

Another strategy is to partition a network by physically connecting all workstations and servers that need to communicate with each other to the same segment. One way in which this can be implemented is to place all servers at partition boundaries. This is illustrated in Figure 8.11. You can also partition a network in a similar manner using switches, firewalls, or routers.

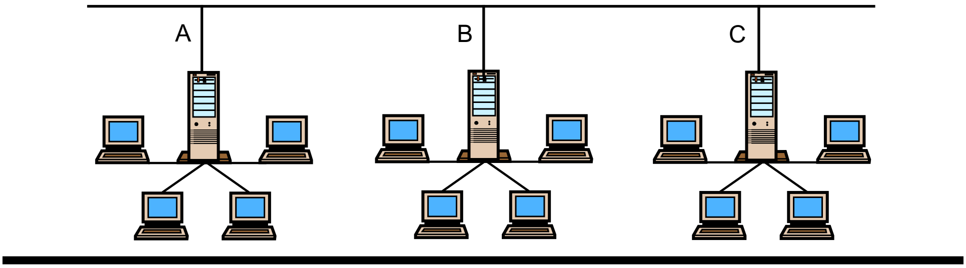


FIGURE 8.11 Another way to partition a network is to physically connect all nodes with their server and to place all servers at partition boundaries. In this illustration, there are three separate partitions—A, B, and C. Note that network traffic is reduced since the majority of communication occurs within a separate segment. At the same time, by having the servers on the partition boundaries, nodes from one segment are not locked out from communicating with a server located on another segment.

31. The concept is simple enough to understand, but how difficult is it to partition a network?

Partitioning requires proper network analysis prior to implementation. If not properly configured, you can actually increase traffic loads rather than reduce them. For example, suppose in Figure 8.11, a node on the C segment needs to communicate with the server on the A segment on a frequent basis. Now, instead of keeping this node's transmissions local to the C group, traffic must cross the backbone. This increases backbone traffic, which increases overall network congestion. Partitioning a network into separate, physical segments is easy to do and maintain when dealing with static workgroups. However, in dynamic workgroup settings where employees are assigned to work on different projects for different periods of time, partitioning is difficult to implement. For example, suppose 50 people located in three different buildings are assigned to a project that uses applications installed on a centrally located server. It is quite impractical to move these people and their workstations to the same physical location and segment that supports the server. Consequently, 100 percent of this workgroup's network traffic will traverse the backbone.

32. Can anything be done to rectify this situation?

Yes. You can establish *virtual local area networks*, or VLANs. VLANs are created using Ethernet switches, which are part of a switched Ethernet environment.

33. Hold it. Before you get into VLANs, brief me on Ethernet switches and the concept of switched Ethernet.

OK, but you should also review Chapter 6, which contains a general discussion about Ethernet switches and switch architecture. The concept of *switched Ethernet* is based on the following principle: If the traffic a node receives is restricted to only the traffic destined for it, then network loads are reduced because every host would not need to examine every frame placed on the network. Thus, switched Ethernet transforms traditional Ethernet/802.3 from a broadcast technology to a point-to-point technology—it isolates network traffic between sending and receiving nodes from all other connected nodes. This is done using Ethernet switches, which are improved bridges that were first introduced in 1990. Today, Ethernet switches are a key element in Ethernet/802.3 LANs, and the switch concept is now a part of ATM, token ring, FDDI, and 100-Mbps and 1000-Mbps Ethernet networks. Ethernet switches are available in various varieties including workgroup, private, and backbone.

34. Are these different from the store-and-forward and cut-through switches discussed in Chapter 6?

The terms *store-and-forward* and *cut-through* are used to describe a switch's architecture, that is, how it operates. The terms *workgroup, private,* and *backbone* are used to describe a switch's application. For example, *workgroup switches* (also called *segment switches*) might support 1024 MAC addresses per port. (Recall that 1024 is the maximum number of nodes permitted on an Ethernet/802.3 LAN.) These switches are really fast multiport bridges because each port supports a shared medium. A workgroup switch partitions a single, shared medium into multiple, shared media. For example, in Figure 8.12 (a) each node is contending for a piece of a 10-Mbps channel. In part (b), though, only 10 nodes per segment now contend for the shared medium. Thus, each node in part (b) effectively receives one-tenth of the channel instead of one-hundredth as in (a), which improves overall network performance tenfold.

Unlike workgroup switches, *private switches* support only one MAC address per port providing each node with its own dedicated 10-Mbps segment. This eliminates contention for the cable, thereby liberating the end nodes from performing collision detection. Private switches are appropriate for workstations running applications requiring high-bandwidth. An illustration of a private Ethernet switch is given in Figure 8.13. Switches supporting 100-Mbps and 1000-Mbps are available. These switches are particularly appropriate for client-server applications. For example, if, in Figure 8.13, node 8 were a server and nodes 1–7 were clients, all traffic would be to or from the server. Since each port supports a dedicated 10-Mbps link, a bottleneck could conceivably exist at the server port. This situation is resolved by using a 10/100 switch, which includes dedicated 10-Mbps and 100-Mbps ports. Clients are connected to the 10-Mbps ports, and servers are connected to the 100-Mbps ports. This is shown in Figure 8.14. Some switches accommodate both shared and dedicated segments. Thus, dedicated ports are assigned to users who require greater bandwidth (e.g., those who frequently transfer large graphical images to or from a server), and shared LAN segments are used for low bandwidth users (e.g., those whose applications include only e-mail). An illustration is provided in Figure 8.15.

A third and final application of Ethernet switches involves incorporating them into the network backbone. Within this context, switches are referred to as *backbone switches*, and the network topology is described as a *collapsed backbone*. Backbone switches can be employed either at the building level or for the entire enterprise. They usually are chassis-based devices and accommodate different media types. Backbone switches also are available with fault-tolerance (e.g., multiple power supplies) and redundancy features. Deploying a building backbone switch can contribute to a gradual reduction of overall backbone traffic, and collapsing the entire backbone into a single hub centralizes the backbone, which can enable better management control over the network elements. An example of a backbone switch is given in Figure 8.16.

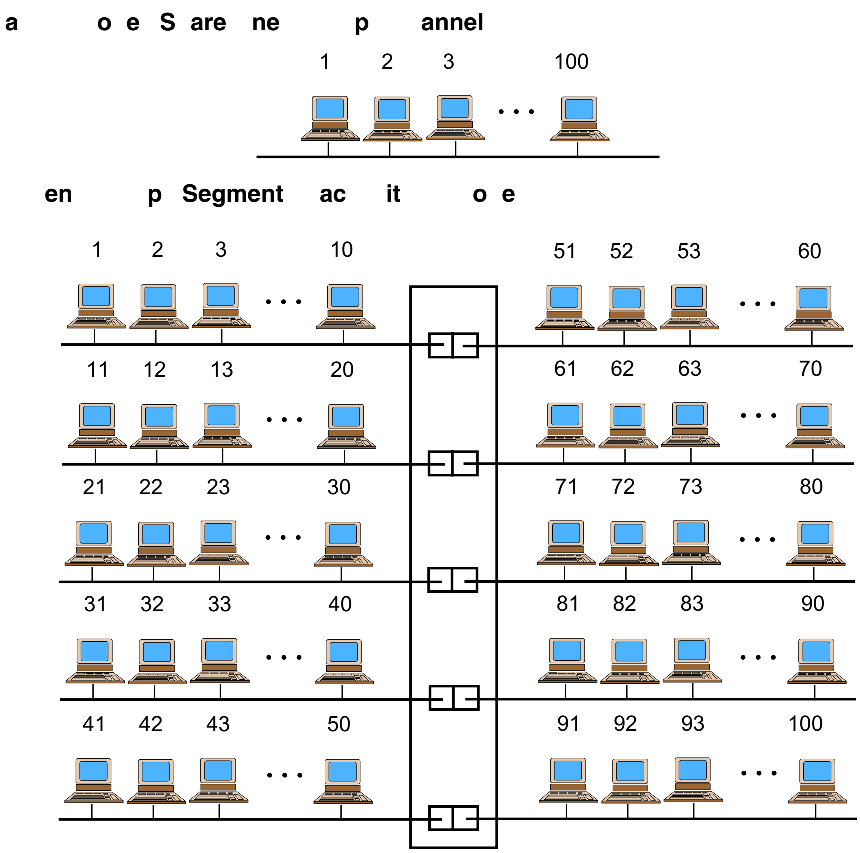


FIGURE 8.12 The 100-node unsegmented 10-Mbps Ethernet network in (a) implies each node occupies one-hundredth of the shared medium. In (b), a workgroup Ethernet switch is used to partition this network into multiple LAN segments with ten nodes each. Each node now receives one-tenth of a 10-Mbps channel, and overall network performance for all nodes increases by a factor of 10.

35. When reviewing a vendor's switch specifications, I see the term "wire speed" used. What does this mean?

Wire speed, when used in the context of a switch's specifications, means that the device's filtering and forwarding rate is equal to the maximum (i.e., fastest) rate possible. In Ethernet/802.3, wire speed is equal to 14,880 frames per second. This is frequently reported as 14,880 packets per second (pps). (See Box 8.3 for additional information.)

36. If a private Ethernet switch eliminates contention, is it possible for nodes to transmit and receive data simultaneously?

What you are referring to is *full-duplex Ethernet*, and the answer is yes. Full-duplex Ethernet requires a private Ethernet switch that supports full-duplex ports, and hosts require full-duplex NICs. Full-duplex Ethernet cards are best suited for servers rather than

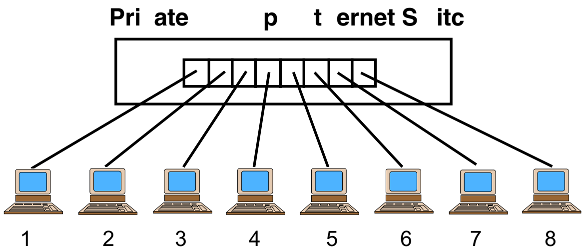


FIGURE 8.13 A private 10-Mbps Ethernet switch resembles a UTP repeater hub. Unlike a repeater hub, a private switch dedicates a full 10-Mbps channel to each port, which supports only one node (i.e., one MAC address). Since each node has its own dedicated segment, there is no need for a node to perform collision detection.

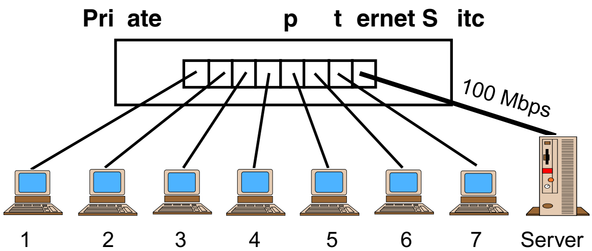
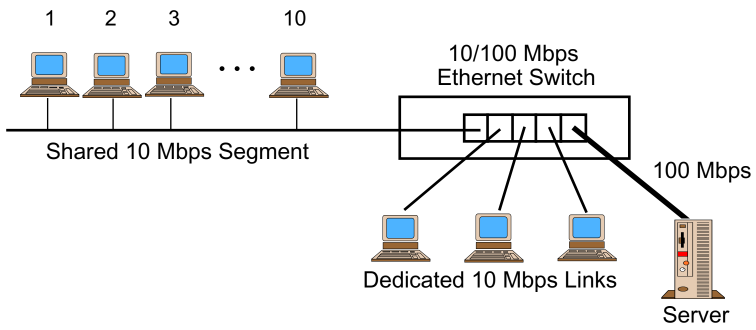


FIGURE 8.14 A 10/100-Mbps private Ethernet switch can be used in a client/server environment to assign dedicated 10-Mbps channels to each client and a full, dedicated 100-Mbps channel to the server. Since all traffic is to and from the server, the higher-bandwidth segment reduces the possibility of a bottleneck existing at the server port.

clients because servers have considerably more bidirectional traffic than clients. With collision detection disabled, full-duplex nodes can transmit and receive data at the same time. Thus, aggregate throughput per segment is doubled from 10 Mbps to 20 Mbps—10 Mbps on the transmit pair and 10 Mbps on the receive pair. The advantages of full-duplex Ethernet cannot be realized unless end nodes are running a multithreaded operating system such as UNIX or Windows NT. Nodes not running this type of OS will realize only a marginal performance increase from full-duplex hardware. Compared to other high-speed networking initiatives such as 100/1000 Mbps Ethernet or FDDI, full-duplex Ethernet is relatively inexpensive, easy to implement, and has no real price penalty. As with all networking initiatives, the issue of interoperability needs to be considered and addressed before a decision is made to upgrade a 10-Mbps Ethernet to full-duplex Ethernet.



IGURE 8.15 Some Ethernet switches are capable of supporting a mix of different segments. Here, en nodes are connected to a shared 10-Mbps segment, three nodes have their own dedicated 10-Mbps egments, and a file server is connected to a dedicated 100-Mbps segment.

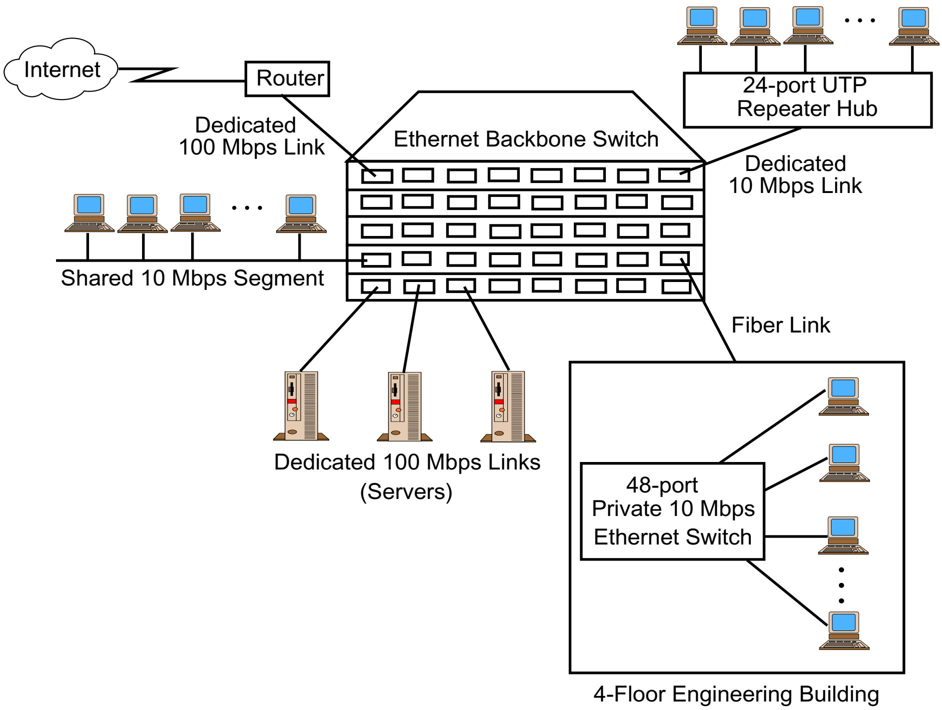
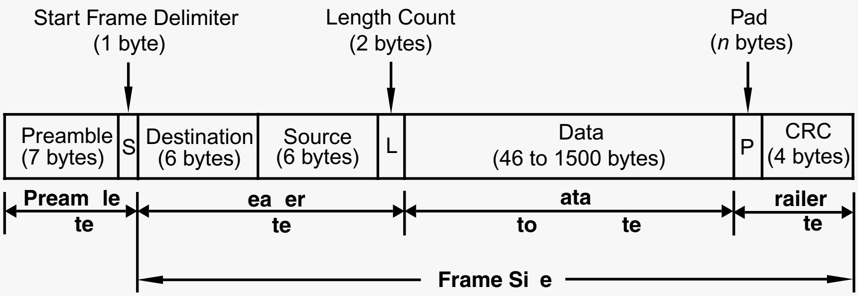


FIGURE 8.16 A backbone Ethernet switch enables an organization's entire network to be collapsed into the switch. These switches are chassis-based and have gigabit per second backplanes. They also support multiple media types, shared or dedicated segments, and both 10-Mbps and 100-Mbps segments. To compensate for a single source of failure, backbone switches have provisions for fault tolerance and redundancy.

BOX 8.3: Understanding Ethernet "Wire Speed" and Capacity



In a standard 10 Mbps IEEE 802.3 frame (pictured above), observe that the data field, also referred to as the frame's *payload,* is accompanied by the fields containing addressing, length, and error control information. These header and trailer fields require 18 bytes. Ignoring the 8-byte preamble, which is not considered part of the frame because it contains no interpretable data, the size of an IEEE 802.3 frame varies from 64 bytes to 1,518 bytes. Clearly, as the size of the data field increases, the ratio of fixed overhead to data increases. Thus, 64 bytes is not only the smallest, but also the most inefficient IEEE 802.3 frame size. A worst-case scenario for network devices such as bridges, switches, and routers is constantly having to process 64-byte frames. Thus, various performance tests of networking devices are usually conducted using 64-byte frames. Let's calculate the frame rate for these minimum-sized IEEE 802.3 frames.

- A 64-byte frame is equal to $8 \times 64 = 512$ bits
- A 10 Mbps transmission rate is equal to 10 bits per microsecond (μs)
- Thus, at 10 Mbps, a 64-byte frame is transferred in 512/10 = 51.2 μs

The 8-byte preamble requires another 6.4 μs, and between each two frames is an *interframe gap* requiring 9.6 μs. Summing all these times, we obtain a total of 67.2 μs per frame. If we divide 1 second by this sum, we get 0.0148809 μs. Thus, one frame occupies 0.0148809 μs. Since there are 1 million microseconds in 1 second, if we multiply 0.0148809 by 1,000,000, we get 14,880. Therefore, a 10-Mbps Ethernet network transmits a maximum of 14,880 64-byte frames in 1 second. This frame rate, 14,880 frames per second, is referred to as *wire speed* in the literature. Cut-through switches have wire speed filtering and forwarding rates. We further calculate that a maximum size Ethernet frame (i.e., 1518 bytes) plus all its associated overhead is transmitted in 1230.4 μs, which yields a maximum frame rate of 812. In other words, given a transmission rate of 10 Mbps, 812 maximum-length Ethernet frames can be transmitted in 1 second. Using these calculations, we can now estimate the amount of payload that can be transferred each second on a 10-Mbps Ethernet/802.3 LAN. We multiply the frame rate by the size of the data field. As noted above, the data field ranges from 46 bytes to 1500 bytes, which, at frame rates of 14,880 and 812 per second, respectively, yields 5.476 Mbps and 9.861 Mbps, respectively. Clearly, a network dealing with only the largest possible frames would come closest to delivering the promise of 10-Mbps Ethernet; however, in reality, we can expect a transmission rate closer to the middle of the range we calculated.

(continued)

BOX 8.3: Understanding Ethernet "Wire Speed" and Capacity (continued)

Since the frame format for Fast Ethernet is the same as 10-Mbps Ethernet, the time parameters given above for 10 Mbps hold except for the interframe gap, which is one-tenth as long or 0.96 μs. Doing similar calculations, the wire speed for Fast Ethernet is 148,809 frames per second, and the maximum frame rate is 8,120.

We cannot, however, calculate the time parameters for Gigabit Ethernet, which operates at 1000 Mbps, by simply multiplying our 10-Mbps Ethernet results by 100. This is because a Gigabit Ethernet frame has a *carrier extension* that increases the minimum frame size to 512 bytes. We, still, nevertheless, can calculate similar parameters for Gigabit Ethernet by following the same procedure we used for 10-Mbps Ethernet.

- A 512-byte frame is equal to $8 \times 512 = 4096$ bits
- A 1000-Mbps transmission rate is equal to 1000 bits per microsecond (μs)
- Thus, at 1000 Mbps, a 512-byte frame is transferred in $4096/1000 = 4.096$ μs

In Gigabit Ethernet, the interframe gap is one-hundredth of that for 10 Mbps, or 0.096 μs. Furthermore, the 8-byte preamble requires another 0.064 μs. Summing all these times, we obtain a total of 4.256 μs per frame. If we divide 1 second by this sum, we get 0.2349624 μs. Thus, one frame occupies 0.2349624 μs. Since there are 1 million microseconds in 1 second, if we multiply 0.2349624 by 1,000,000, we get 234,962. Therefore, a 1000-Mbps Ethernet network transmits a maximum of 234,962 512-byte frames in 1 second. Since Gigabit Ethernet's maximum frame size is still 1518 bytes, Gigabit Ethernet can transmit $812 \times 100 = 81{,}200$ maximum-sized frames per second.

A technology currently being reviewed by IEEE increases Ethernet's 1500-byte data field to 9000 bytes, which results in what is called an Ethernet *jumbo frame*. Jumbo frame technology enables data transfer rates to approach gigabit speeds on an Gigabit Ethernet LAN, which is currently limited in performance due to the relatively small data field. For example, a 900,000-byte message requires 600 frames that support a 1500-byte data field, but only100 jumbo frames. Thus, the amount of processing overhead for Gigabit Ethernet is six times more than the jumbo-frame Gigabit Ethernet's overhead for the same message.

37. Remind me again why standard Ethernet cannot operate in full-duplex mode.

Ethernet was initially designed as a broadcast network based on a physical bus topology. Its operation is limited to half-duplex because nodes have to use the receive pair to listen for collisions while transmitting.

38. OK. Let's get back to VLANs. What are they?

A VLAN is a virtual local area network. Unlike nodes connected to a physical LAN, nodes that comprise a VLAN are not physically connected to the same medium. They are connected in a virtual sense using specially designed software that groups several ports in a switch into a single workgroup. Nodes connected to these ports are considered to be part of a workgroup, and network traffic from any node/port is (usually) limited to only those nodes or ports assigned to the workgroup. Depending on the switch, traffic filtering is per-

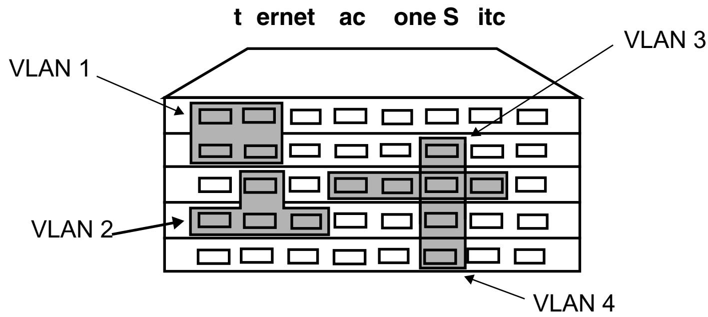


FIGURE 8.17 Some Ethernet switches support virtual LANs, which group together selected ports to form independent subnets. Traffic between ports of a VLAN is restricted to only those members of a "port group." Some switches also support routing protocols so that ports can belong to more than one VLAN, as is illustrated in VLANs 3 and 4.

formed using either MAC addresses (layer 2) or network addresses (layer 3). In the case of the latter, a switch must operate at layer 3 (the network layer), and the switch must support the routing protocol used in the network. In an IP-based network, if the switch supports the Internet Protocol, then workgroups are created by using a node's IP address. The advantage of using layer-3 protocols on which to base a VLAN is that an individual node can be assigned to more than one virtual subnetwork at the same time. A server, for example, can be shared on a virtual basis by more than one workgroup. VLANs based on MAC addresses enable any node to be assigned transparently to a workgroup, but virtual subnets cannot share nodes. An illustration of a VLAN is shown in Figure 8.17.

39. How do VLANs fit into the partition schema?

In a dynamic work environment, various workgroups are formed for short-lived projects. Personnel selection criteria for these projects are usually based on employee strengths and expertise, not employee location. Consequently, a workgroup might involve 25 people who are dispersed throughout an organization and who require access to equally dispersed servers. VLANs can resolve this problem because they permit users who are connected to different physical segments to be organized into logical workgroups, independent of users' physical location. As users relocate within an organization by changing job assignments, offices, or computers, their node addresses (MAC or network) are deleted from one workgroup table and added to another. VLAN also can reduce congestion, problems of isolation, and improve network security.

40. It sounds like Ethernet switches resolve all of conventional Ethernet's limitations.

It depends. An Ethernet switch certainly helps improve the overall performance of a heavily loaded LAN. However, if a network's architecture is poor to begin with, a switch will only mask the problems by increasing throughput without really identifying and resolving the true problems. Consequently, prior to installing a switch it is prudent to first study how a network is being used. This includes performing a network analysis to determine traffic loads and patterns, user applications, location of users and servers, and type of data transmissions. In some cases, giving attention to specific network administration tasks might prove sufficient. For example, Ethernet/802.3 performance and efficiency can be improved through collision management, standards adherence, proper system configurations, software upgrades, and segmentation using bridges. Nevertheless, there are certain situations that lend themselves to switching. These include dynamic workgroup settings, frequent data transmissions involving large data files such as graphical images, distributed network environments that employ several servers, and LANs that must support video or multimedia applications. If a decision to "switch" is made, remember that all switches are not alike. Switches have different architectures, are designed for different types of applications, and some switches also have routing capabilities (called layer-3 switches). Once again it is important to understand how a network is being used before modifying it.

41. What's next?

100-Mbps Ethernet, 1000-Mbps Ethernet, and 10,000-Mbps Ethernet.

42. Why are these needed when switched and full-duplex Ethernet are available?

Although switched and full-duplex Ethernet—and segmentation—improve overall network performance, these implementation strategies for enhancing 10-Mbps Ethernet performance do not actually increase data transmission rates, that is, they do not speed up the network. To address this issue of speeding up the network, IEEE increased Ethernet/802.3's 10-Mbps data transmission rate, first to 100 Mbps, then to 1000 Mbps, and in the very near future, to 10,000-Mbps Ethernet. This strategy of speeding up the network is different from the 10-Mbps Ethernet strategies discussed earlier and can be best understood by viewing a network as a crowded highway. A strategy that employs segmentation, switched Ethernet, or full-duplex Ethernet adds more lanes to the highway. A 100-Mbps, 1000-Mbps, or 10,000-Mbps technology, however, actually increases the speed limit of the highway by orders of magnitude, from 10 Mbps to 100 Mbps to 1000 Mbps to 10,000 Mbps.

43. Why do we need to increase the speed of the LAN?

There are several reasons. First, the introduction of more sophisticated operating systems and applications, faster processors, and greater disk and memory capacities, has led to a pronounced degradation in overall network and application performance. Second, many companies now use Internet technologies to build private, corporate intranets that rely on standard web browsers such as Netscape's *Communicator* and Microsoft's *Internet Explorer* to provide employees with access to critical corporate data. These web browsers are capable of supporting bandwidth-intensive rich media data types, including high-resolution graphics, 3D imaging, audio, video, and voice. Third, the introduction of multimedia client/server applications, the deployment of network computers and *Java*-based servers, and an increasing number of network users, are taxing first-generation Ethernet.

44. What's the history of 100-Mbps Ethernet?

The concept of 100 Mbps Ethernet was first introduced in 1992. By August 1993, the Fast Ethernet Alliance (FEA) was formed. Within two years, membership grew to more than 80 vendors, including 3Com, Cabletron, Intel, DEC, and Sun Microsystems. The primary purpose of FEA was to entice the IEEE to adopt Fast Ethernet technology as a standard. In June 1995, this new technology became the IEEE 802.3u standard and was given the specification 100BASE-T. With its goals accomplished, the FEA concluded its activities in September 1996.

45. Presumably, 100BASE-T was derived from 10BASE-T. How does Fast Ethernet compare to 10BASE-T and conventional Ethernet?

Your presumption is correct; 100BASE-T did indeed evolve from 10BASE-T, specifically, and conventional Ethernet, generally. At the physical layer, 100BASE-T employs a star topology and supports twisted-pair and fiber-optic cable. Unlike 10BASE-T, though, Fast Ethernet has three different media specifications: 100BASE-TX, 100BASE-T4, and 100BASE-FX (see Table 8.2). (The designation 100BASE-X is sometimes used to represent 100BASE-TX and 100BASE-FX.) 100BASE-TX and 100BASE-T4 use twisted-pair cable; 100BASE-FX uses fiber-optic cable. Note the absence of coaxial cable, which was the mainstay of the original Ethernet specification. All three components of the standard are designed to interoperate with one another. Fast Ethernet also uses the same connector types as 10BASE-T and maintains a 512 bit-time collision domain. There is one obvious difference, though—speed. Fast Ethernet is ten times faster than conventional Ethernet—100 Mbps versus 10 Mbps. This tenfold increase in speed has two direct implications. First, it reduces the amount of time it takes to transmit one bit by a factor of 10. In Fast Ethernet, 512 bit-times is equal to 5.12 μs instead of 51.2 μs. Second, the network diameter is reduced by a factor of 10. These diameters are summarized in Table 8.2.

At the data link layer, Fast Ethernet is unchanged from its 10-Mbps counterpart. The frame format, the minimum and maximum frame sizes (including the amount of user data a frame can transmit), and the MAC address format are all identical to conventional Ethernet. Perhaps most important, though, Fast Ethernet uses exactly the same media access method, namely, CSMA/CD, as conventional Ethernet. This provides an easy, simple, and seamless migration path for current users of 10-Mbps Ethernet/802.3, particularly shared or switched 10BASE-T users. (*Note:* Many network fundamentalists equate Ethernet with CSMA/CD. It is their belief that if the MAC sublayer protocol of a LAN technology is not CSMA/CD-based, then the word "Ethernet" should not be used to either reference or describe the technology.)

46. Could you expand on the information provided in Table 8.2.? What would be helpful is a comparison among the three specifications.

OK. We'll discuss them in their order of appearance in Table 8.2. Following is a brief description of each.

100BASE-TX 100BASE-TX transmits and receives data over two pairs of EIA/TIA 568-compliant Category 5 UTP cable, or two pairs of IBM Type 1 STP cable. It uses a full-duplex signaling system based on FDDI's twisted pair physical medium dependent (TP-PMD) sublayer, which is an ANSI standard that defines the manner in which data are encoded/decoded and transmitted. (*Note:* TP-PMD was once called copper distributed

TABLE 8.2 Summary of IEEE 802.3u (Fast Ethernet—100 Mbps) Specifications

Type	Description
100BASE-TX	**Medium:** 2-pair Category 5 UTP or IBM Type 1 STP; **Topology:** Star; **Maximum Segment Length:** 100 m; **Connectors:** Category 5 compliant 8-pin modular (RJ-45), patch panels, patch cables, punch-down blocks; **Media Access Control:** CSMA/CD; **Network Diameter:** 200 m when used with one Class I or one Class II repeater; 205 m when used with two Class II repeaters; 261 m when used with mix of UTP/fiber cable and one Class I repeater; 289 m when used with mix of UTP/fiber cable and one Class II repeater; 216 m when used with mix of UTP/fiber cable and two Class II repeaters; **Other:** Full-duplex operation.
100BASE-T4	**Medium:** 4-pair Category 3, 4, or 5 UTP or IBM Type 1 STP; **Topology:** Star; **Maximum Segment Length:** 100m; **Connectors:** 8-pin modular (RJ-45), patch panels, patch cables, punch-down blocks; **Media Access Control:** CSMA/CD; **Network Diameter:** 200 m when used with one Class I or one Class II repeater; 205 m when used with two Class II repeaters; 231 m when used with mix of UTP/fiber cable and one Class I repeater; 304 m when used with mix of UTP/fiber cable and one Class II repeater; 236 m when used with mix of UTP/Fiber cable and two Class II repeaters; **Other:** Half-duplex operation
100BASE-FX	**Medium:** 2-strand 62.5/125 multimode fiber-optic; **Topology:** Star; **Maximum Segment Length:** 412 m (half-duplex) or 2 km (full-duplex); **Connectors:** ST, SC, or FDDI's media interface connector; **Media Access Control:** CSMA/CD; **Network Diameter:** 272 m when used with Class I repeaters; 320 m when used with one Class II repeater; 228 m when used with two Class II repeaters; **Other:** Designed primarily to interconnect Fast Ethernet repeaters.

data interface, or CDDI, which applied to running FDDI over UTP. FDDI is discussed in Chapter 10.) Networks based on the 100BASE-TX standard must be Category 5-compliant throughout, including wire, connectors, patch panels, and punch-down blocks. Since many new network installations employ four-pair Category 5 UTP cable, a 100BASE-TX installation leaves managers with an "extra" two pairs of wires that can be used for voice communication or be reserved for future network enhancements. Presently, these "extra" two pairs cannot be used to support another high-speed LAN.

100BASE-T4 100BASE-T4 uses a half-duplex signaling system to transmit and receive data over four pairs of Category 3, 4, or 5 UTP cable, or four pairs of IBM Type 1 STP cable. One pair of wires is used exclusively for transmitting data, one pair is used exclusively for receiving data and collision detection, and the remaining two pairs are used either for transmitting or receiving. As a result, three pairs of wires can be used for data transmission or three pairs can be used for data reception. This scheme of using three wire pairs for transmitting or receiving of data reduces overall cable frequency because the signal can be divided among these wires. A direct effect of this design is that lower quality cable such as voice-grade Category 3 can be used to support a higher speed technology like Fast Ethernet. 100BASE-T4's advantage over 100BASE-TX is that the former can be used in Category 3 or 4 wiring installations. Thus, organizations can scale-up their 10-Mbps Ethernet networks to 100-Mbps without modifying their existing cable plants, or they can opt to install Category 3 wire, which is less expensive than the superior grade Category 5 wire, and still benefit from 100-Mbps Ethernet. Although 100BASE-T4 supports inferior cable such as Category 3, it does not support 25-pair Category 3 wire for horizontal runs, which is what is commonly installed for voice transmission. 100BASE-T4 is not without its drawbacks, however. Unlike 100BASE-TX, which uses only two pairs of wire, 100BASE-T4 must use all four wire pairs. It also does not support full-duplex operation.

100BASE-FX 100BASE-FX supports 100-Mbps Ethernet operation over two strands of 62.5/125 micron multimode fiber-optic cable (one strand for transmitting data and one pair for receiving data). It shares the same signaling system as that of 100BASE-TX except it uses FDDI's fiber physical media dependent sublayer. Unlike 100BASE-TX or 100BASE-T4, 100BASE-FX segments are known formally as *link segments*, which are designed to connect only two nodes in a point-to-point topology. Consequently, 100BASE-FX's primary application is at the backbone and is used to connect Fast Ethernet hubs.

47. Does Fast Ethernet use Manchester coding?

No. Unlike Manchester coding, which conventional Ethernet is based on, 100BASE-T relies on two different encoding techniques. 100BASE-TX and 100BASE-FX use an encoding scheme called 4B/5B, and 100BASE-T4 uses an encoding scheme called 8B/6T. Originally defined for FDDI (Chapter 10), 4B/5B takes data in four-bit codes and maps them to corresponding five-bit codes that are transmitted using NRZI. The 4B/5B-NRZI scheme makes it possible for Fast Ethernet, as well as other LAN technologies, to operate at 125 MHz and provides a data rate of 100 Mbps. The 8B/6T method maps eight-bit data blocks to a specific code group consisting of six symbols. These code groups are then transmitted across three output channels. The effective data rate on each channel is 33 Mbps, with a signaling rate of 25 Mbaud.

48. Wait a minute, here. You lost me. What's NRZI?

We thought we could slide this by you. Oh well! You caught us. Without getting too bogged down in the technical details, we'll give you an overview so you will have some knowledge of the encoding used in Fast Ethernet.

NRZI is part of the *nonreturn to zero* (NRZ) family of codes in which positive and negative voltages are used for encoding 0s and 1s. In one form of NRZ, called *nonreturn to zero level* (NRZL), a constant positive voltage is used to represent a 0-bit and a constant negative voltage is used to represent a 1-bit. NRZI is a variation of NRZL. Instead of using level voltages to encode the data, though, encoding is based on transitions from one voltage state to another (i.e., from a low-to-high state or from a high-to-low state). Specifically, in NRZI, data are coded 0 if no transitions occur, but are coded 1 at the beginning of a transition.

An application of NRZI can be found in several LAN systems, which use as their signaling method a group encoding strategy known as the 4B/5B (four bits to five bits) method. The 4B/5B encoding scheme takes data in four-bit codes and maps them to corresponding five-bit codes. For example, the four-bit data code for the letter F (1111) corresponds to the five-bit encoding 11101 (see Table 10.1). These five-bit codes are then transmitted using NRZI. By transmitting five-bit codes using NRZI, a logical 1-bit is transmitted at least once every five sequential data bits, resulting in a signal transition. The 4B/5B-NRZI scheme makes it possible for LANs such as Fast Ethernet and FDDI (Chapter 10) to operate at 125 MHz and provides a data rate of 100 megabits per second. Note that the use of one extra bit for every five bits translates to only 20% overhead for every clock encoding. In contrast, Manchester coding requires 50% bandwidth overhead for clock encoding because it guarantees at least one signal transition for every bit transmitted. As a point of interest, copper-based Gigabit Ethernet (discussed later) also uses an NRZI scheme on the copper version of the technology.

49. Does Fast Ethernet follow the 5-4-3 repeater placement rule?

No. The repeater rules for Fast Ethernet are different than for conventional Ethernet. First, IEEE 802.3u defines two classes of repeaters. *Class I* repeaters support both of Fast Ethernet's signaling schemes (100BASE-T4 and 100BASE-TX/FX), and *Class II* repeaters support only one signaling scheme (100BASE-T4 or 100BASE-TX/FX, but not both). Class I repeaters have a latency of no more than 0.7 μs; the latency of Class II repeaters is less than or equal to 0.46 μs. The two signaling types are interoperable at both node and hub levels. When maximum cable lengths are used, only one Class I repeater, or a maximum of two Class II repeaters—with a maximum interrepeater link of 5 m—can exist within any single collision domain. Furthermore, since Class II repeaters can only be used to connect segments that have the same signaling schemes, 100BASE-T4 segments cannot be connected to 100BASE-TX/FX segments using a Class II repeater. A direct consequence of these new classes of repeaters is that the 5-4-3 repeater placement rule for 10-Mbps Ethernet does not apply to Fast Ethernet. The network diameters for 100BASE-T also have different ranges depending on which repeater is used, the number of repeaters used, and the cable type (see Table 8.2).

50. How difficult is it to convert or migrate from 10-Mbps Ethernet to Fast Ethernet?

Assuming the physical layer satisfies Fast Ethernet specifications, all that is required to convert from 10-Mbps Ethernet/802.3 to Fast Ethernet is to swap out a node's network interface card. New 10BASE-T nodes also can be accommodated for 100BASE-T migration by installing 10/100 Mbps NICs, which employ an autosensing/negotiation feature that enables them to operate at data rates of either 10 Mbps or 100 Mbps. Many Ethernet switches also incorporate both 10 Mbps and 100 Mbps ports. This makes it possible for 10BASE-T and 100BASE-T segments to be connected to the same switch. From a technological perspective, the designers of 100BASE-T made it easy for network managers to deploy Fast Ethernet at their sites in a relatively seamless fashion.

51. How easy or difficult is the conversion to Fast Ethernet?

From a practical perspective, *any* network migration or upgrade endeavor is usually problematic. Given this basic tenet, converting to Fast Ethernet is not necessarily easily accomplished. Consider the cable plant. Although 100BASE-T supports UTP, it requires four pairs of Category 3, 4, or 5 UTP cable (100BASE-T4). Four pairs of Category 3 cable means that all eight wires of a standard Category 3 UTP bundle must be used to achieve a data rate of 100 Mbps. This is not feasible for sites using two pairs of Category 3 UTP for their 10BASE-T networks if the other two pairs of wire are being used for telephone connections (or for additional Ethernet connections). For 10BASE-T LANs using two pairs of Category 5 UTP, all is not well either. Cable installation requirements for 100BASE-TX are extremely stringent requiring all components (from connectors to patch panels to number of twists per inch) to be certified Category 5 compliant. Many so-called Category 5 10BASE-T LANs do not meet these specifications and hence will have to be modified. Finally, coaxial cable-based LANs (ThinWire or ThickWire) also will require major changes to their cable plants since 100BASE-T does not support coaxial cable. In fact, none of the newer higher-speed LAN technologies supports coaxial cable, which is viewed by some as a diminishing technology without a future.

In addition to the cable plant, incorporating Fast Ethernet into an existing LAN or as a new LAN installation also has an impact on several network design issues. There are shorter cable lengths, and Fast Ethernet only permits two types of repeaters. This translates to pronounced limitations on network diameters and collision domains. These restrictions can have a dramatic effect on how a network is designed. For example, more wiring closets might be necessary for a 100BASE-T than for a 10BASE-T installation, and additional hardware (e.g., bridges or switches) will be required to extend a 100BASE-T LAN.

Finally, attention should be given to both topology and network nodes. Fast Ethernet must be configured as a star, not as a bus. Furthermore, Fast Ethernet was designed with switches in mind, not bridges. Deployment of 100BASE-T also presupposes the use of nodes capable of supporting the increase in speed. Hence, LANs consisting of ISA bus-based nodes will not benefit from Fast Ethernet, and workstations using anything less than a 32-bit operating system will not realize an increase in throughput either.

Here's the bottom line: Networks are nontrivial and Fast Ethernet is no exception. Network managers considering implementing Fast Ethernet as part of a new LAN installation, or integrating it into an existing 10-Mbps Ethernet LAN, need to give serious attention to the various physical limitations, restrictions, and design issues related to 100BASE-T. It is also prudent for managers to understand the IEEE specifications and their ramifications before getting involved with Fast Ethernet. Box 8.4 contains specific strategies for migrating or upgrading to Fast Ethernet.

52. While we are on the subject of 100 Mbps Ethernet, what is 100VG-AnyLAN?

100VG-AnyLAN is an IEEE standard formally designated IEEE 802.12. It was designed as an upgrade path for 10-Mbps Ethernet/802.3 and 4/16-Mbps token ring. Although it is considered a competing technology to Fast Ethernet, 100VG-AnyLAN is not compatible with conventional Ethernet; its MAC sublayer is not CSMA/CD but instead a technology called *demand priority*, which is similar to that of token ring.

53. Besides the MAC sublayer, how does 100VG-AnyLAN compare to Fast Ethernet?

Instead of giving a lengthy dialogue about 100VG-AnyLAN, we provide a summary of the standard in Table 8.3 and then compare various specifications of the technology to Fast Ethernet (Table 8.4). Please review these tables on your own. We do not want to

BOX 8.4: Migrating or Upgrading to 100BASE-T

Following are several strategies to consider if you are planning to migrate or upgrade an existing 10-Mbps Ethernet/802.3 LAN to Fast Ethernet:

1. For organizations that are not yet ready to implement 100BASE-T, but are still adding new users to their 10BASE-T LAN, the simplest and most cost-effective 10BASE-T to 100BASE-T migration strategy is to install 10/100-Mbps network interface cards in new nodes. These cards can automatically sense the correct data transmission rate based on the hub port to which they are connected. This strategy will also preserve an organization's current investment in its 10BASE-T LAN.

2. For organizations that want a "blended" 10 Mbps/100 Mbps network, of paramount concern is how to interconnect the two networks. Several strategies are possible. One method is to use a bridge. This is probably the least expensive and easiest installation. A second method is to use a router. This can be cost-prohibitive, though, and usually increases the complexity of a network. A third strategy is to use a switch that can support both 10-Mbps and 100-Mbps connections. Some switches permit shared 10-Mbps segments to be mixed with dedicated 10-Mbps and dedicated-100 Mbps segments. Although more costly than bridges, switches do provide a nice migration strategy for a mixed environment.

In a mixed or blended environment, be careful with network diameters and collision domains. It is possible to maintain a 500-m network diameter in this type of environment. One configuration involves 100-m segments to a Fast Ethernet Class II repeater. This repeater is then interconnected to a 10/100-Mbps switch that contains 10BASE-T connections. The switch interconnects to an unpopulated 10/100-Mbps switch, which then connects to a second Class II repeater. This second Class II repeater can support end nodes such as servers.

3. For organizations planning to upgrade to 100BASE-T:

a. Confirm that you really need to upgrade. "Tuning" a network through network analysis, reengineering, segmentation, software upgrades, and collision management can provide tremendous improvements in network performance.

b. Confirm that your cable plant meets the proper specifications.

c. Confirm that all nodes have sufficient horsepower (e.g., PCI bus-based, 32-bit OS), and that all servers have sufficient buffering capacity.

d. Confirm that all users require 100 Mbps. If not, item (2) above might be more appropriate; consider installing full-duplex 10-Mbps switches.

e. Know, understand, and follow the various specifications related to 100BASE-T.

f. Plan on installing only one repeater hub per collision domain.

g. Use bridges or switches to connect to secondary wiring closets.

h. Use two-port switches to extend network diameter.

i. Invest in 100BASE-T compliant diagnostic equipment.

TABLE 8.3 Summary of IEEE 802.12 (100VG-AnyLAN) Specifications

Category	Description
Media	4-pair Category 3 UTP (100 m); 4-pair Category 4 UTP (100 m); 4-pair Category 5 UTP (200 m); 2-pair Category 5 UTP (under investigation); 2-pair Type 1 STP (200 m); 25-pair UTP cable using 50-pin telco connectors; 2 strands 62.5/125 multimode fiber (2000 m)
Media Components	8-pin modular (RJ-45) connectors; patch panels; patch cables; punch-down blocks
Network Cards	100VG-AnyLAN compliant NICs
Hubs	100VG-AnyLAN compliant repeater hubs—all hubs have an uplink port to connect to another VG hub; all ports can be used as downlink ports to connect to end nodes or another VG hub; all ports can be configured in normal mode (only receives data destined for it) or monitor mode (receives all data)
Maximum Nodes	1024 on a single-shared (unbridged) LAN; no more than 250 is recommended, however
Collision Domain	N/A; However, the term *priority domain*, is used to describe a 100VG-AnyLAN network that consists of a root hub and all of its connected nodes, including lower-level hubs and nodes
Network Diameter	8000 m
Topology	Hierarchical star with up to five levels of cascaded repeater hubs
MAC Sublayer	Demand priority—uses a priority-based round-robin arbitration scheme to determine network access; supports both Ethernet/802.3 and IEEE 802.5 frame formats
Transmission Mode	Half-duplex hubs cannot transmit and receive simultaneously because of crosstalk; this is due primarily to Category 3's lower electromagnetic characteristics
Future Enhancements	The following are currently under consideration as of this writing: (1) Data transmission rates of 1.063 Gbps and 1.25 Gbps for fiber-optic links and 500 Mbps for 4-pair Category 5 links; (2) Single mode fiber permitted; (3) Fiber-optic links based on Fibre Channel 8B10B link protocol; (4) Use of VG switches and full-duplex operation for dedicated links; (5) 8-km maximum network diameter with up to five levels of cascaded repeaters

spend a great deal of time discussing this technology because there are relatively few 100VG-AnyLAN installations today compared to Fast Ethernet installations. As noted earlier, 100VG-AnyLAN is not compatible with Ethernet/802.3 LANs because its MAC sublayer is not CSMA/CD. Although some people think 100VG-AnyLAN is a superior technology to Fast Ethernet, others do not consider it "Ethernet." If you're interested in learning more about 100VG-AnyLAN, see Costa (1994).

54. OK. That's fair enough. Can you at least explain demand priority for me?

Sure. 100VG-AnyLAN uses cascaded repeater hubs in a hierarchical star topology. The demand priority protocol specifies the manner hubs poll their ports to identify nodes

TABLE 8.4 A Comparison Between 100BASE-T and 100VG-AnyLAN

	100BASE-T (IEEE 802.3u)	100VG-AnyLAN (IEEE 802.12)
Media		
Category 3 UTP	4-pair (100 m)—100BASE-T4	4-pair (100 m)
Category 4 UTP	4-pair (100 m)—100BASE-T4	4-pair (100 m)
Category 5 UTP	2-pair (100 m)—100BASE-TX	2-pair (N/A)
	4-pair (100 m)—100BASE-T4	4-pair (200 m)
25-pair UTP	Not supported	Supported
IBM Type 1 STP	Yes (100 m)—100BASE-T4/TX	Yes (100 m)
Fiber-optic (62.5/125)	412 m half-duplex—100BASE-FX	Yes (2000 m)
	2 km full-duplex—100BASE-FX	
Topology		
Network diameter	Varies from 200 m to 320 m depending on cable type and repeaters used	8 km
Cascading repeaters	Two levels	Five levels
MAC Sublayer		
Media Access	CSMA/CD	Demand priority
IEEE 802.3 Frames	Yes	Yes
IEEE 802.5 Frames	No	Yes
Application Support		
Time-sensitive data	No	Yes
Performance		
100 m throughput	80%	95%
2500 m throughput	Not supported	80%

with data to transmit and the order of these transmissions. The protocol works in the following general manner (an example of this polling strategy is given in Figure 8.18):

> A 100VG-AnyLAN repeater hub polls each node connected to it for a transmission request. The hub performs this query by continuously scanning its ports sequentially, from lowest connected port to highest connected port. If a node needs to transmit data, a transmission request is conveyed to the hub at the time the node is polled. Only one data frame per node per polling cycle is transmitted, and data frames are identified by the hub as either normal- or high-priority. Frames designated high-priority (e.g., real-time video and audio) are processed (i.e., given access to the network) before normal-priority-designated frames (e.g., data files).

Each hub has at least one uplink port, which connects to a higher-level hub; every port can be used as a downlink port to connect to an end node or a lower-level hub. Hub ports can be configured to operate in either *normal* or *monitor mode*. Ports operating in normal mode receive only those data frames destined for it as determined by a frame's destination address. Monitor mode on the other hand is similar to Ethernet's *promiscuous mode*, which

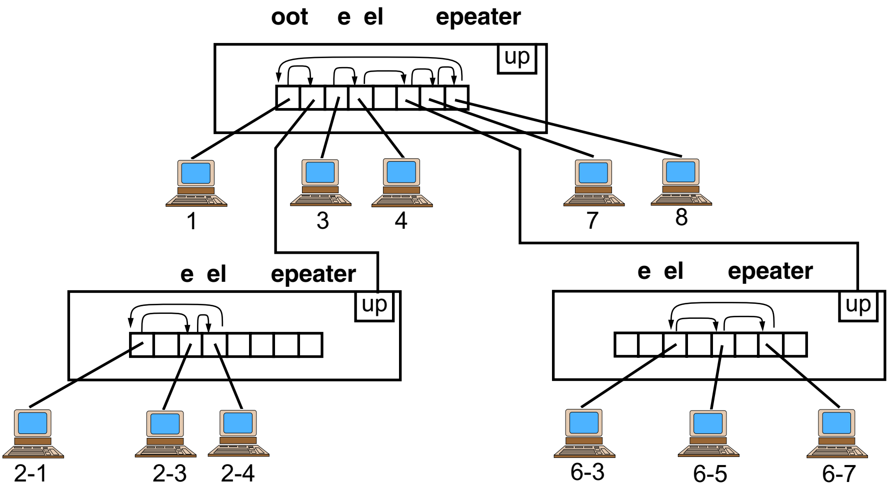


FIGURE 8.18 An example of a two-level 100VG-AnyLAN network. The hub's polling order in this illustration is 1, 2-1, 2-3, 2-4, 3, 4, 6-3, 6-5, 6-7, 7, 8. If during a polling cycle nodes 3, 2-4, 6-5, and 7 have high-priority requests pending, then the data transmission order is 2-4, 3, 6-5, 7, 1, 2-1, 2-3, 4, 6-3, 6-7, 8 for that cycle. If priority requests are the same (all normal or all high), then the data transmission order is the port order. However, if the priority requests are not the same, then the transmission order is determined by considering the priority of the node's pending request and the physical port order of the node. For example, if nodes 3 and 7 have normal-priority requests, but node 4 has a high-priority request, then the transmission order is 4, 3, 7. Source; Adapted from Schnaidt, 1994a.

indicates that every data frame received by the hub is directed to all ports configured to operate in monitor mode. (*Note:* An Ethernet NIC set in promiscuous mode collects all messages placed on the medium regardless of their destination address.) Using a round-robin scheme, repeaters continuously poll all their connected ports in sequence to determine which nodes have data transmission requests pending. A polling cycle begins when a hub polls the lowest numbered port connected to it and ends when the highest-numbered connected port is polled. A polling cycle is repeated continuously in port-order. Nodes are permitted to transmit only one frame of data per polling cycle. The process of polling and determining data transmission order is called *prioritized, round-robin arbitration* and is the heart of the demand priority protocol. Theoretically, a maximum of five levels of cascading are permitted in a 100VG-AnyLAN network. However, in practice, a maximum of three is the general rule.

The type of priority assigned to a data frame is not part of the demand priority protocol and hence does not occur at the MAC sublayer. Instead, data priority assignment is performed by the upper layer application software and passed to the MAC sublayer as part of a data frame. It is the demand priority protocol's design that enables it to identify the type of data transmission request and process high-priority data before normal-priority data. As a result, this media access method is highly suited for transmitting isochronous data, that is, time-sensitive data such as multimedia applications and real-time video and audio for video conferencing. To guard against high-priority data transmissions from monopolizing a network, VG hubs are designed with a *watchdog protocol* that monitors the "wait time" of pending normal-priority request. All normal-priority requests that have been waiting between 200 ms and 300 ms have their priority label changed automatically from normal to high. They are then placed on the hub's high-priority list and processed in port-order fashion. Thus, all normal-priority data are guaranteed access to the network.

55. Demand priority is interesting. It must be much more efficient than Ethernet.

You are correct in your observation. 100VG-AnyLAN's demand priority protocol is a very robust technology. It is deterministic, collision-free, supports a priority scheduling scheme, exhibits stable behavior during high load times, and uses bandwidth efficiently (demonstrated 95 percent throughput at 100 m). Demand priority also improves some of the shortcomings of token ring, including eliminating token rotation delays.

56. Why does 100VG-AnyLAN have an IEEE 802.12 designation instead of an IEEE 802.3 designation like Fast Ethernet? Aren't they both 100-Mbps Ethernets?

As noted before, many Ethernet purists do not consider 100VG-AnyLAN "Ethernet" because of its access method. They contend that Ethernet implies a contention-based MAC sublayer protocol and since demand priority eliminates collisions and the concept of collision domains, 100VG-AnyLAN is therefore not Ethernet. The absence of CSMA/CD led the IEEE to prohibit the 100VG-AnyLAN working group from using the Ethernet designation. IEEE also believes that Ethernet is not Ethernet without CSMA/CD.

57. I see. Should I consider 100VG-AnyLAN for my LAN?

That is up to you. If you compare the market share of the two technologies, you will find that the battle for 100 Mbps turf has already been won by Fast Ethernet. Regardless of IEEE's (and others') view about 100VG-AnyLAN not being "Ethernet," 100VG-AnyLAN supports both Ethernet/802.3 and token ring frame formats. It is also a compelling alternative to Fast Ethernet as an upgrade path for 10 Mbps Ethernet, and can serve as an upgrade path for 4/16 Mbps token ring LANs as well. Any decisions about LAN upgrades or new installations should be driven by an organization's networking needs and the best technology available that satisfies those needs.

58. Let's move on to Gigabit Ethernet. So far I've learned that the IEEE has gone from 10 Mbps Ethernet to two different 100 Mbps Ethernet specifications. Now there is 1000 Mbps Ethernet. That's 1 billion bits per second. When did this get started?

Just as the dust was beginning to settle on the Fast Ethernet standard, the IEEE, in March 1996, commissioned the Higher Speed Study Group (HSSG) to investigate increasing Fast Ethernet's data transmission rate tenfold to 1000 Mbps. In July 1996, the IEEE 802.3z Task Force was approved to define standards for the next evolution of Ethernet—*Gigabit Ethernet*. To support the IEEE's standards efforts in this endeavor, and to educate customers and the networking industry on this new technology, the Gigabit Ethernet Alliance (GEA) was formed in May 1996. Using the Fast Ethernet Alliance as its model, the Gigabit Ethernet Alliance sought to rally the same combination of suppliers and consumers of FEA to (a) support the extension of Fast Ethernet standards and (b) address customers' needs for interoperability among 10/100/1000 Mbps Ethernet products. GEA quickly grew in size to more than 100 vendor-members within six months of its creation.

59. Talk about working fast. Do we really need Gigabit Ethernet, though? When will we ever use that much capacity?

In the world of networking, fast is never fast enough. The deployment of bandwidth-hungry multimedia applications, the integration of faster computer systems, and the migration of Fast Ethernet from the backbone to the desktop are creating bottlenecks at the server level or at interswitch connections. Gigabit Ethernet alleviates this congestion by providing a faster backbone technology. Remember, when Ethernet was first developed, the majority of the applications and computer systems of that era could not saturate a 10 Mbps channel. Today, though, we have 64-bit servers, improved bus speeds, 100 Mbps desktop units, and applications such as real-time, two-way desktop video conferencing. So yes, Gigabit Ethernet really is necessary.

60. So the bigger and faster the pipe, the better the network. Right?

Not necessarily. Bigger is not always better. It is not prudent to increase bandwidth for the sake of increasing bandwidth. Bigger and faster backbones are no panacea for network congestion. Deploying Gigabit Ethernet undoubtedly will reduce overall network congestion, but without conducting an a priori network analysis to determine the source of the congestion, the deployment of Gigabit Ethernet will do nothing more than mask the problem. Some people call this strategy "bandwidth Band-Aid®." Before deploying any new high-speed backbone technology, including Gigabit Ethernet, network managers should examine all areas of potential bottleneck (e.g., desktop, servers, and backbone). This will help them assess exactly where increased bandwidth is to be deployed, when it needs to be deployed, and provide a sound rationale for deploying it.

61. OK. Tell me about Gigabit Ethernet.

Gigabit Ethernet, which is denoted in the trade publications as GbE or GigE, has two separate standards: IEEE 802.3z, which specifies Gigabit Ethernet over fiber, and IEEE 802.3ab, which specifies Gigabit Ethernet over copper. The IEEE 802.3z standard was approved in June 1998; the IEEE 802.3ab standard was ratified in October 1999.

62. Why are there separate fiber and twisted-pair specifications?

To provide a 1000 Mbps Ethernet standard expeditiously, the IEEE adopted the ANSI Fibre Channel signaling protocol and modified it for IEEE 802.3z. Because there was no similar existing protocol to run gigabit speeds over Category 5 UTP, IEEE created a separate task force (IEEE 802.3ab) to address this endeavor. By doing this, the IEEE 802.3z task force was able to maintain a dedicated focus to establishing a fiber-based gigabit Ethernet standard.

63. Wait a minute. I've got two questions here. First, I know there were Gigabit Ethernet products available in 1997, almost a year before the standard was approved. How was this possible? Second, what's a Fibre Channel?

You're right. There were indeed Gigabit Ethernet products available well before the standard was approved. In January 1997, the 802.3z task force closed the specification to new features, thus enabling leading network vendors to develop Gigabit Ethernet products. By agreeing on a stable first draft of the specification, vendors were able to get a jump on product development. They also were able to demonstrate their products' interoperability at the October 1997 Networld+Interop show. Remember, the Gigabit Ethernet Alliance, which was comprised of network vendors, was formed for the express purpose of designing a Gigabit Ethernet specification. So it was in their best interest to resolve any issues related to the specification as quickly as possible. In the end, Gigabit Ethernet was one of the fastest standards to be approved among the second generation, high-speed LAN standards. GEA was a quick study of the Fast Ethernet Alliance.

To answer the second part of your question, *Fibre Channel* (FC) is a family of ANSI (American National Standards Institute) standards that defines a specific communications interface for high-speed data transfers between different hardware systems. FC's applications include the medical profession, where large images (e.g., 100 MB+ X-rays) are transferred from a scanner to a computer to a screen, and the electronic publishing industry, where large files are transferred from an designer/creator's machine to a publisher's computer. It has also become the backbone of storage area networks (see Chapter 1). FC is organized into a five-level hierarchy (FC-0 through FC-4). IEEE 802.3z signaling is based on FC-0 and FC-1. FC-0 supports a variety of physical media and data rates; FC-1 defines the signaling encoding technique used for transmission and synchronization across a point-to-point link (8B/10B—8 bits of data are encoded into 10-bit characters and transmitted serially). IEEE 802.3z modified the link frequency from the 1.062 GHz ANSI standard to 1.25 GHz so that a full 1000-Mbps data rate is supported. That's all we are going to say about FC. For additional information, consult the Bibliography or the Fiber Channel Association's Web site (http://www.amdahl.com/ext/carp/fca/fca.htm).

64. What are the physical layer specifications for Gigabit Ethernet?

At the physical layer, IEEE 802.3z supports three specifications: 1000BASE-SX (short wavelength fiber), 1000BASE-LX (long wavelength fiber), and 1000BASE-CX (short-haul copper). The IEEE 802.3ab standard has one physical layer specification: 1000BASE-T, which defines running Gigabit Ethernet over Category 5 UTP cable at distances up to 100 m. This distance restriction is equivalent to that of Fast Ethernet. However, unlike Fast Ethernet, all four pairs of Category 5 UTP cable must be used. Table 8.5 contains a summary of the media and distance specifications for the fiber- and copper-based Gigabit Ethernet versions; Table 8.6 compares Gigabit Ethernet to conventional and Fast Ethernet; and Figure 8.19 shows Ethernet's family tree.

65. What is Gigabit Ethernet's data link layer like?

At the data link layer, Gigabit Ethernet supports the conventional 802.3 frame format. Thus, it has a 64-byte minimum frame size and maintains the same 96-bit interframe gap

TABLE 8.5 Conventional Ethernet versus Fast Ethernet versus Gigabit Ethernet

	Conventional Ethernet	Fast Ethernet	Gigabit Ethernet
Data Rate	10 Mbps	100 Mbps	1000 Mbps
Max Segment Lengths:			
Category 5 UTP	100 m	100 m	100 m
IBM Type 1 (STP)	500 m	100 m	25 m
Multimode Fiber	2 km	412 m (half-duplex) 2 km (full duplex)	260–550 m
Single Mode Fiber	25 km	20 km	3 km

Source: 3Com; Adapted from Tolley, 1997a

TABLE 8.6 Media and Distance Comparisons of Gigabit Ethernet

	Media	Max Distance
1000BASE-SX	62.5-µm Multimode Fiber 50-µm Multimode Fiber	220–275 m[a] 500–550 m[b]
1000BASE-LX	62.5-µm Multimode Fiber 50-µm Multimode Fiber 9-µm Single-Mode Fiber	550 m 550 m 5000 m (5 km)
1000BASE-CX	Coaxial	25 m
1000BASE-T	Category 5 UTP	100 m

[a] 200 m for TIA 568 fiber optic wiring standard; 275 m for ISO/IEC 11801 building wiring standard
[b] 550 m based on ANSI Fibre Channel specifications
Source: Adapted from Conover, 1998; Henderson, 1998; and Tolley, 1997a.

(see Box 8.3). It also supports full-duplex and switched connections. The fiber-based version of Gigabit Ethernet (IEEE 802.3z) also supports the same CSMA/CD access method in full-duplex mode, but it uses a slightly modified version of CSMA/CD in half-duplex mode—the minimum CSMA/CD carrier and slot-times are 512 bytes instead of 64 bytes. Thus, the minimum-sized frame of Gigabit Ethernet is 512 bytes, not 64 bytes. This modification was necessary to maintain a 200 m collision diameter in half-duplex. If this was not done, then the maximum diameter would have been one-tenth the size of a Fast Ethernet LAN (25 m) because when you increase the bit rate, the collision domain and overall network diameter decrease. A maximum network diameter of 25 m is not very practical.

66. Does Gigabit Ethernet really work over Category 5 UTP?

Yes it does, though the issue of how well Gigabit Ethernet will run over Category 5 UTP has been a nagging one. Suffice it to say that defining the physical layer for 802.3ab was not easy for the task force. Although it is possible to transmit data over Category 5 UTP at gigabit speed, success depends on a clear signal path. This means that if anything obstructs the signal as it travels from source to destination, then reflections can occur and

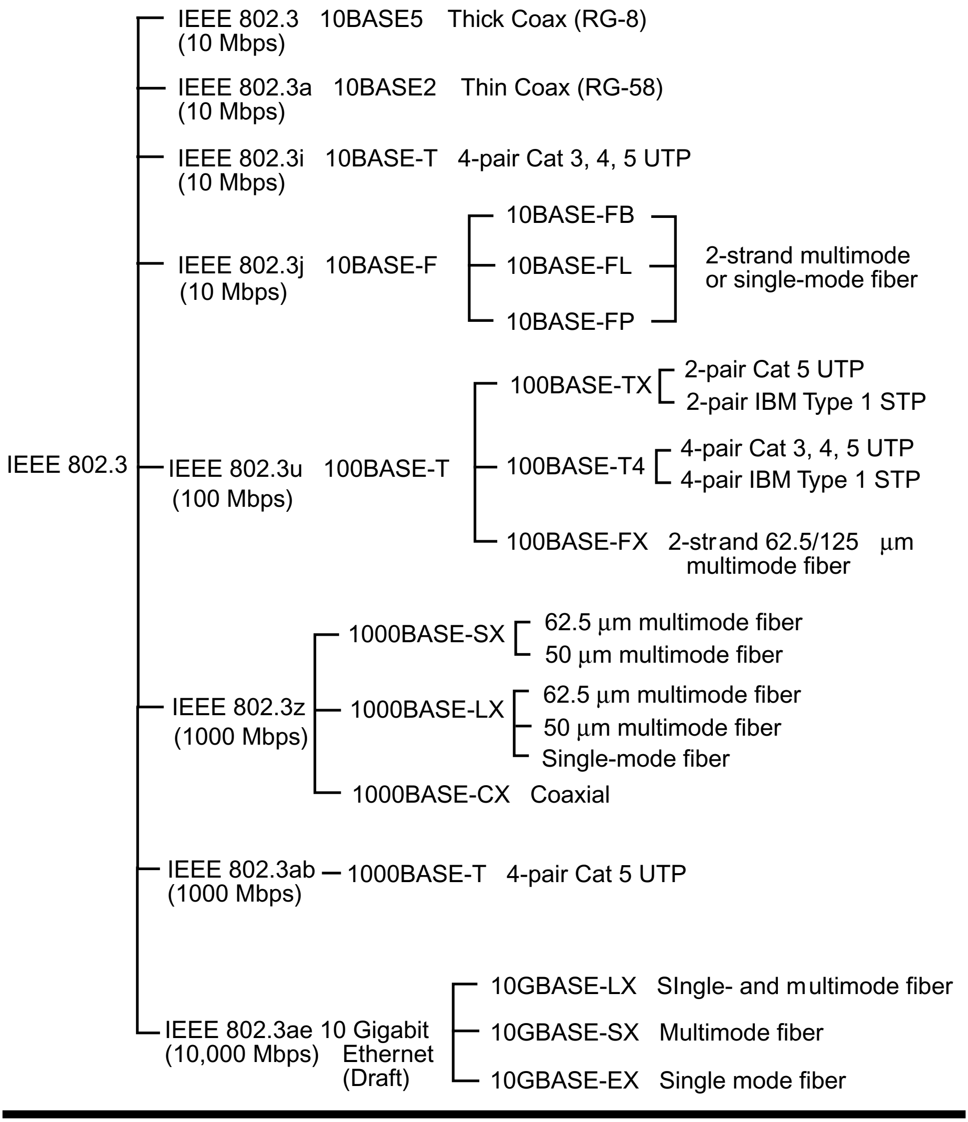


FIGURE 8.19 Various IEEE 802.3 specifications for the different variants of baseband Ethernet and their respective media.

reduce the reliability of network performance. "What objects can cause reflections?" you might ask. Oh, little things like eight-pin modular (RJ-45) connectors or punch-down blocks. It is important to note that this is not unique to Gigabit Ethernet. Reflections are always present in copper cable when used for radio frequency transmission in both 10- and 100-Mbps Ethernet links (see Chapter 4). The difference is that at these relatively lower speeds, reflections and related interference are not as serious. Furthermore, unlike

100BASE-TX (Fast Ethernet over Category 5 UTP), which uses only two pairs of wire (one pair to transmit and one pair to receive), 1000BASE-T requires four pairs and transmits signals bidirectionally on all four of them, that is, 250 Mbps per pair. Since Gigabit Ethernet uses all four pairs of copper simultaneously, the potential for electrical noise increases. This includes something called *far-end crosstalk* (FEXT), which is similar to NEXT (see Chapter 4) except FEXT involves the occurrence of crosstalk at the end of the cable opposite from the transmitter. Another type of electrical noise that needs to be considered when running Gigabit Ethernet over copper is *echo*, which describes what happens when a signal is reflected back to the transmission source. Also called *return loss*, echo is usually a function of impedance mismatches caused by bad connectors or patch cables. (You might want to review these concepts from Chapter 4.)

As you can see, implementing Gigabit Ethernet over copper involves considering parameters other than just cable length, attenuation, and NEXT. Attention must also begin to ensuring that your cable plant is 1000BASE-T-compliant. To this end, the Telecommunications Industries Association (TIA) modified its current cable testing standards document, TSB-67, to include the factors that have to be considered when all four pairs of copper cable are being used at the same time. This new document, TSB-95, defines the necessary testing parameters for certifying the copper cable plant for Gigabit Ethernet. For an update on these standards, see TIA's Web site at http://tiaonline.org.

67. What about some of the newer copper cable such as Category 6 or 7 UTP? Aren't these cable types available and shouldn't they be able to support 1000BASE-T?

New high-speed copper cable designed specifically for gigabit speed is indeed available from various vendors. Called "gigabit copper," these new offerings include Category 6 UTP, Category 6 STP, and Category 7 UTP. (see Table 4.1). Although gigabit copper has higher frequencies, this does not necessarily translate to higher speeds. Contrary to what some network managers might believe, there is not a 1:1 ratio between megahertz and megabits. From a design perspective, though, gigabit copper supports gigabit encoding schemes. One drawback to gigabit copper is a lack of standards. This means that parameter specs provided by vendors are subject to different interpretations. Case in point: One cable manufacturer uses the number 350 in the name of one of its products, giving the impression that the cable is rated at 350 MHz. Upon further investigation, the cable's spec sheet reveals that the cable cannot support data transfers beyond 200 MHz. It is so named, though, because it provides "stable electrical performance" up to 350 MHz. Another potential problem with gigabit copper is that standards committees like EIA/TIA are vendor-based consortia and therefore politically charged. Trying to assess what a final gigabit copper standard will look like is difficult enough without being influenced by politics. The bottom line is wait—wait until gigabit copper standards are ratified; wait until IEEE 802.3ab is ratified, deployed and tested; wait for others to find out what works and what doesn't, that is, adopt a "state of the practice" and not a "state of the art" mindset.

It is also important to realize that even if gigabit copper cabling were standardized, the primary focus of IEEE when it designed a gigabit Ethernet standard for copper cable was the installed base of Category 5 UTP. In other words, IEEE wrote 1000BASE-T so that users currently running Ethernet/802.3 over Category 5 UTP could migrate to Gigabit Ethernet without replacing their cable plant. According to the IEEE BASE-T task force, a

100BASE-TX connection will support 1000BASE-T. On the other hand, if you are planning a new cable installation, the Gigabit Ethernet Alliance recommends that you install Category 5e. If you want to grab some gusto, you also might want to consider installing the new gigabit copper cables (either Category 6 or Category 7 UTP) because 1000BASE-T should be able to operate over these cable types as well.

68. Does this mean, then, that 1000BASE-T also uses the 4B/5B encoding scheme used by Fast Ethernet?

Not exactly, but close. As noted earlier, Fast Ethernet (i.e., 100BASE-TX) uses two wire pairs—one to transmit and one to receive—and a signaling rate of 125 MHz. Gigabit Ethernet over copper (i.e., 1000BASE-T) uses the same signaling rate as Fast Ethernet but as noted earlier requires four wire pairs and transmits signals bidirectionally on all four of them, 250 Mbps per pair. Unlike 100BASE-TX, though, 1000BASE-T uses a five-level coding scheme. When you combine the coding scheme with the four wire-pair transmission, 1 byte of data is transmitted in parallel at each signal. The arithmetic works out as follows:

$$
\begin{aligned}
& 4 \text{ pairs} \times 125 \text{ MHz} \times \frac{2 \text{ bits}}{\text{symbol}} \\
&= 4 \times \frac{125 \text{ Msymbols}}{\text{second}} \times \frac{2 \text{ bits}}{\text{symbol}} \\
&= 4 \times 125 \times 2 \ \frac{\text{Mbits}}{\text{second}} \\
&= 500 \times 2 \text{ Mbps} \\
&= 1000 \text{ Mbps}
\end{aligned}
$$

It is also interesting to note that as part of AT&T's Hero project in 1984, there was a 1-Gbps test of Category 3 UTP, as well as a test of 40-Mbps Fiber Network protocol over coaxial cable. It was determined that STP could handle, without any modification to the waveform, up to about 1 GHz. UTP, on the other hand, with modifications to the waveform, would permit transmission rates up to about 1 GHz. With phase encoding, compression, and other types of bit-level encoding, it was recently shown that Category 5 UTP can support transmission rates of 2.4 Gbps over a 20-m link and that 1 Tbps (terabits per second, which is 1 million million, or 1 million Mbps) is on the horizon.

69. In what ways can Gigabit Ethernet be deployed?

Gigabit Ethernet can be deployed in many different ways. In one application, Gigabit Ethernet can be used to upgrade switch-to-server or switch-to-switch connections. For example, server farms or Fast Ethernet switches can be connected to a Gigabit Ethernet switch to boost 100 Mbps links to 1000 Mbps links. This will provide users with higher-speed access to application servers (thus reducing the potential for bottlenecks) and increase the number of Fast Ethernet segments that can be supported by the network. In a second application, a Gigabit Ethernet switch can be used as a collapsed backbone (see Chapter 6). In this scenario, the switch becomes an aggregation device that provides 1 Gbps links to multiple switches, repeater hubs, and routers. A fourth application is to use Gigabit Ethernet to replace shared FDDI rings (Chapter 10), which interconnect buildings within a campus environment. Gigabit Ethernet can also be deployed directly to the desktop making bandwidth-intensive applications such as desktop video conferencing accessible. Figure 8.20 illustrates some of these applications.

70. What's the bottom line on Gigabit Ethernet?

On the positive side, Gigabit Ethernet:

- provides a tenfold increase in raw performance over Fast Ethernet;
- provides familiar technology so that existing investments in hardware, software, and personnel are maintained (and protected);
- represents a relatively small learning curve;
- offers tremendous scalability;
- is a natural extension to 10/100-Mbps Ethernet networks;
- supports all existing networking protocols;
- is complementary to ATM;
- is intended to run over Category 5 UTP;
- is an IEEE standard; and
- might support acceptable levels of CoS and QoS for real-time video and voice transmissions.

On the negative side, Gigabit Ethernet:

- requires new switches and NICs;
- might require workstation upgrades to take advantage of the increase in speed;
- is relatively expensive since it is still a new technology (although prices are falling and it is relatively less expensive than competing technologies); and
- has no inherent CoS and QoS support.

71. Will Ethernet max out at 1000 Mbps?

Are you kidding? As of this writing, a 10-Gbps Ethernet technology, IEEE 802.3ae, is being prepared as a supplement to the existing IEEE 802.3 standard. Following the successful blueprint established by its predecessors, FEA and GEA, the 10 Gigabit Ethernet Alliance (10 GEA) was formed in February 2000 to (a) promote the advancement of a 10 Gigabit Ethernet standard, (b) provide technical assistance to IEEE so it can quickly adopt 10 Gigabit Ethernet technology as a standard, and (c) provide resources to address interoperability issues among multivendor 10 Gbps Ethernet products. The IEEE 802.3ae 10 Gigabit Ethernet Task Force completed the first draft of the standard in September 2000, prestandard products began appearing in 2001, and the draft standard is on track for ratification during the first half of 2002. (The actual target date is March, 2002.) In all likelihood, 802.3ae will be an IEEE standard shortly after this book is published. When examined from the context of Fast Ethernet's and Gigabit Ethernet's development, to quote Yogi Berra: "It's *deja vu* all over again." It doesn't stop with 10-Gbps Ethernet (reported in the trade magazines as 10 GbE). There are also projections that by 2005 we will have a 40-Gbps Ethernet standard, and a 100-Gbps Ethernet version might be available in 2009.

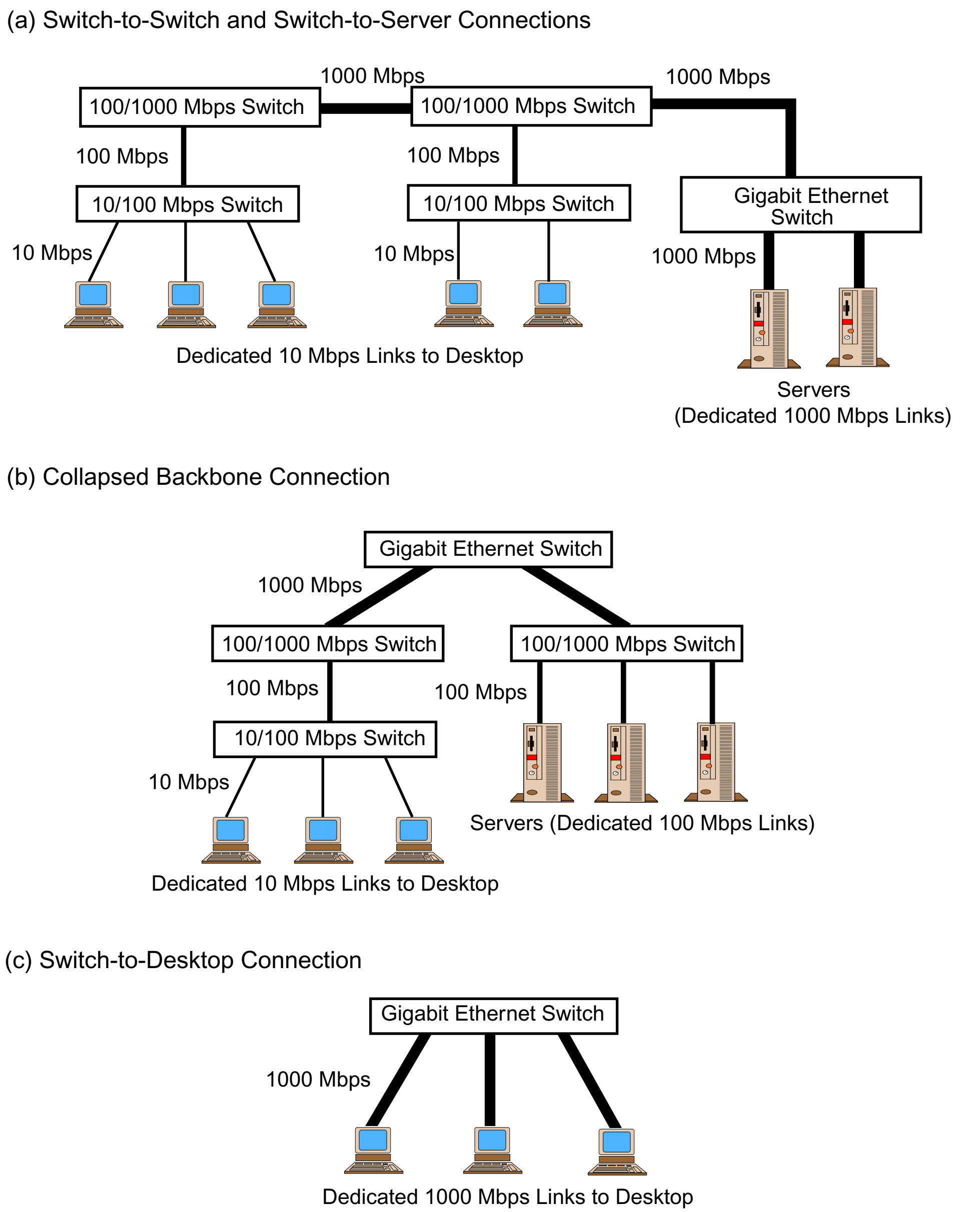


FIGURE 8.20 Sample Gigabit Ethernet deployments. In (a), Gigabit Ethernet is used to upgrade switch-to-switch or switch-to-server connections. Here, Fast Ethernet switches with 1000 Mbps modules are interconnected via a 1000 Mbps link; servers also have dedicated 1000 Mbps links. In (b), a Gigabit Ethernet switch is used as an aggregate device in a collapsed backbone topology. In (c), a Gigabit Ethernet switch provides dedicated 1000 Mbps links directly to the desktop. Source: Adapted from Gigabit Ethernet Alliance, 1996.

72. Yikes! What's the purpose of a 10-Gbps Ethernet technology?

The primary purpose for designing a 10-Gigabit Ethernet technology is to extend IEEE 802.3 from 1000 Mbps (i.e., 1 Gbps) to 10,000 Mbps (i.e., 10 Gbps). There is another purpose, though. The designers also endeavor to move Ethernet from the LAN to MAN/WAN environments. The deployment of 1 GbE began Ethernet's migration to MAN environments. A 10 GbE technology will provide Ethernet with a more pronounced presence in the MAN and extend its presence to the WAN. If you think about this for a moment, a single Ethernet throughout—from the desktop to the WAN—makes sense because it will keep the overall network simple. With 10-Gigabit Ethernet, we can design LANs, MANs, and WANs using Ethernet as the end-to-end layer 2 transport. The benefits are many. For example, data transported across an Ethernet-based LAN/MAN/WAN would not require any protocol translations. The inherent transmission latencies that are attendant to a multiprotocol world, as well as the management problems that often accompany working in a multiprotocol environment, would be eliminated. Equally important is that an end-to-end Ethernet protocol will ensure the highest level of compatibility with currently installed 802.3 LANs.

73. This makes sense. I'm curious, though. What changes did IEEE have to make to the physical or data link layers to make Ethernet work at 10 Gbps?

IEEE went to great lengths to define 10 GbE so that data transmission rates can occur at 10 Gbps with minimal modifications to current 802.3 physical and data link layer protocols. They did have to make a couple of sacrifices, though. For example, the technology supports only one medium: fiber; no other media are supported. The standard also operates only in full-duplex mode; a half-duplex 10-Gbps CSMA/CD is not supported. Thus, IEEE 802.3ae does not need the CDMA/CD protocol and hence provides a collisionless environment. This means that unlike its predecessors, 10 GbE cable length restrictions are not a function of the network's collision domain. Link distances are instead a function of optic constraints of the fiber cable. Specifically, target cable lengths are 65 m and 300 m over multimode fiber and 2 km, 10 km, and 40 km over single-mode fiber (Table 8.7). (See Chapter 4 for the differences between multimode and single-mode fiber.) The new 10 GbE standard also supports two data rates: 10 Gbps and OC-192c (see Table 7.3 for OC data rates).

74. Why two data rates? Does this imply two physical layer specifications?

The reason for two data rates is to support LAN and WAN connections. The first rate is compliant with the LAN; the second rate supports data transmissions across the WAN. And yes, the two data rates imply two physical layers. Both layers are similar except the WAN physical layer includes an optional sublayer called the WAN interface sublayer (WIS) that enables 10 GbE to be compatible with service providers' existing SONET backbones. (See Chapter 7 for information about SONET.) WIS enables 10 GbE switches/routers to be connected to SONET/SDH equipment and Ethernet frames to be transported across a SONET backbone at a data rate compatible with SONET/SDH transmissions. Although there are two physical layers, both LAN and WAN specifications can use the same optical transceivers. The difference between the two specifications is distinguished via their encoding methods (Table 8.8). Finally, Ethernet/802.3's attachment unit interface

TABLE 8.7 10 Gigabit Ethernet Target Cable Lengths

Optical Transceiver Wavelength	Fiber Type	Target Link Distances
850 nm serial	Multimode	65 m
1310 nm WWDM[a]	Multimode	300 m
1310 nm serial	Single-mode	2 km and 10 km
1310 nm WWDN	Single-mode	10 km
1550 nm serial	Single-mode	40 km or more

[a] Wide Wave Division Multiplexing (see Chapter 4)
Source: 10 Gigabit Ethernet Alliance, 2001

TABLE 8.8 10 Gigabit Ethernet Signal Encoding

Optical Transceiver Type[a]	Encoding Scheme	Wavelength—Data Transmission Method
10GBASE-LX		
• 10GBASE-LX4	8B/10B[b]	1310 nm—WWDM[c]
• 10GBASE-LR	64B/66B	1310 nm—Serial
• 10GBASE-LW[d]	64B/66B	1310 nm—Serial
10GBASE-SX		
• 10GBASE-SR	64B/66B	850 nm—Serial
• 10GBASE-SW[d]	64B/66B	850 nm—Serial
10GBASE-EX		
• 10GBASE-ER	64B/66B	1550 nm—Serial
• 10GBASE-EW[d]	64B/66B	1550 nm—Serial

[a] Names tentative as of this writing
[b] Same as 1 GbE (i.e., 1000BASE-X)
[c] Wide Wave Division Multiplexing (see Chapter 4)
[d] Also supports a WAN interface sublayer (WIS) extension for connecting to SONET devices

Source: 10 Gigabit Ethernet Alliance, 2001

(AUI) was modified to support 10 GbE. This new interface is denoted XAUI (pronounced "zowie"); the X, taken from the Roman numeral system, represents 10 Gbps.

75. In what way will 10 GbE be deployed?

The initial application of 10 GbE is expected to be in a service provider's WAN backbone and points of presence (POPs). It will also compete directly with SONET (see Chapter 7), which is currently emplaced within a carrier's WAN backbone and can achieve equivalent data rates. Ultimately, though, 10 GbE will permeate throughout LANs, MANs, and WANs. Examples of these applications follow.

10 GbE LAN-Based Deployment/Applications The deployment of 10 GbE in LANs will most likely follow similar upgrade/migration paths as those of 1 GbE (see Figure 8.20). Thus, 10 Gigabit Ethernet can be used to link multiple 1 GbE switches or provide very high-speed dedicated links to server farms or SANs; 10 GbE switches can be used to aggregate all of a LAN's connections and thus serving as a collapsed backbone; and 10 GbE technology can be used to deliver dedicated 10-Gbps links directly to the desktop for bandwidth-intensive applications such as computation-intensive processing, remote visualization projects, and full motion video.

10 GbE MAN-Based Deployment/Applications The deployment of 10 GbE in MANs will be used to upgrade current 1 GbE MAN backbones or to establish new 10 GbE backbones. These "metro Ethernets" will deliver network services throughout a metropolitan area (up to 100 km) using point-to-point connections via Ethernet switches or routers. Because Ethernet technology is being used, 10 GbE MANs will enable network service providers to provision 10/100/1000 Mbps Ethernet links similar to the manner in which links are provisioned in LANs. Thus, it is conceivable that customers can have their bandwidth needs met throughout a metropolitan area on an "on-demand" basis.

Newly established 10 GbE MANs are expected to be deployed using *dark fiber*, which refers to single-mode fiber links that are installed but unused. This fiber is referred to as "dark" because in its unused state it is not "lit." That is, the fiber is neither terminated nor supporting any traffic and hence no light is passing through it. As a 10 GbE MAN evolves, Ethernet will also invade the local loop and resolve the "last mile" issue of providing high-speed access between a provider's point of presence (POP) and a customer site such as a home (see Chapter 15). Thus, data transmission rates for home network users, for example, will range from dialup speeds of 56K to Ethernet rates of 1 Gbps.

10 GbE WAN-Based Deployment/Applications The deployment of 10 GbE in WANs is expected to ape that of MANs: New 10 GbE WAN backbones will employ dark fiber and bypass SONET networks completely; existing SONET backbones will connect to 10 GbE switches or routers via 10 GbE's SONET-compatible physical layer interface. In this latter scenario, Ethernet frames will be transported across SONET backbones as encapsulated SONET frames. Ultimately, though, native 10 GbE will run directly over the fiber, and an "Ethernet everywhere" solution will emerge providing true end-to-end Ethernet networks.

76. What will happen to current MAN/WAN technologies such as ATM and SONET?

Well, at the moment, the future doesn't look too bright for either of them. For example, the bandwidth of the fastest public SONET/ATM MAN is OC-3 (155 Mbps) or OC-12 (622 Mbps). These data rates are much less than what 10 GbE has to offer. With an Ethernet-based LAN/MAN/WAN, there is also no protocol conversion or added complexity, which are both "features" of a SONET/ATM solution. Another nail in SONET's coffin is its overall structure. Recall from Chapter 7 that SONET comprises 64-kbps voice channels (DS-0) and was developed for high-volume voice traffic. Although it can support any data type, SONET was optimized for voice transmissions and hence is inefficient for data transmissions because channel capacity is wasted when data packet sizes are not commensurate with the channel.

Whether these two technologies will become obsolete in the face of 10 GbE is still unclear. What is clear, though, is that newly formed Ethernet MANs/WANs will bypass preexisting SONET and ATM networks. As Ethernet establishes itself in the WAN, running ATM over SONET will give way to running Ethernet over SONET; that is, Ethernet frames will be encapsulated within SONET wrappers and transported across the network as SONET frames. This will be done to appease current network providers/managers who are comfortable with using SONET and don't want to change. Ultimately, though, these same providers/managers will begin running native 10 GbE throughout their networks. Nevertheless, you still cannot rule out SONET and ATM completely because just like science fiction stories where anything can happen (Did you really think Spock died in *Star Trek 2*?), anything can happen in the technology world.

77. Speaking of ATM, how does Gigabit Ethernet (either 1 GbE or 10 GbE) compare to ATM?

First, all Ethernet variations are connectionless technologies that transmit variable-length frames; ATM, on the other hand, is a connection-oriented technology that transmits fixed-sized cells (see Chapter 14). Second, now that we have 10 GbE, both Ethernet and ATM technologies can be deployment in LAN/MAN/WAN environments. Unlike Ethernet, though, ATM was designed specifically to transmit any data type, including voice and video traffic. ATM also supports different class of service (CoS) for data prioritization, and can guarantee a quality of service (QoS) needed for real-time voice and video traffic.

Prior to 10 GbE, many people viewed 1 GbE and ATM as noncompeting technologies. For example, if network traffic was primarily data-based and cost was a consideration, 1 GbE was the logical choice. If explicit QoS was paramount for voice and video traffic support, or if a seamless LAN-to-WAN or WAN-to-LAN connection was desired, then ATM was the more appropriate technology. A third illustration portrayed the two as complementary technologies: 1 GbE was deployed within buildings to provide a gigabit-speed building backbone, and ATM was deployed as the enterprise-wide backbone. Today, however, the emergence of 10 GbE has changed many network managers' opinions and ATM and Ethernet are viewed as competing technologies with ATM on its way out.

78. One last question. I've heard of another technology that bears the Ethernet name—IsoEthernet. What is this?

IsoEthernet is short for Isochronous Ethernet, and is an IEEE standard—IEEE 802.9a. The term isochronous means time-sensitive. Hence, in the context of networking, IsoEthernet is designed to support time-sensitive applications such as video conferencing and telephony. It is also inextricably linked to ISDN technology (see Chapter 12)—it runs both Ethernet and ISDN B channels over the same network. The Ethernet channel is used for normal data networking needs; the ISDN B channels are used for time-sensitive applications. Thus, IsoEthernet is really two networks in one. It contains a 10 Mbps Ethernet channel for 10BASE-T traffic, and a separate 6.144 Mbps channel for isochronous traffic. IsoEthernet requires IsoEthernet network adapters at all end nodes requiring isochronous capability, and IsoEthernet hubs at wiring closets.

IsoEthernet has been overshadowed by Gigabit Ethernet and local ATM technologies, and therefore has found little vendor support. Nevertheless, it is anticipated that the 802.9a standard will be modified to include support for 100BASE-T, switched 10BASE-T, and 16 Mbps ATM.

END-OF-CHAPTER COMMENTARY

In this chapter we presented a formal discussion of Ethernet and the various IEEE "Ethernet" protocols. Although it was originally designed as a shared media, contention-based LAN protocol, Ethernet technology has evolved into a robust, high-speed, collisionless MAN/WAN technology boasting speeds (currently) up to 10 Gbps. Ethernet networks can also be designed as shared 10 Mbps half-duplex links to dedicated, full-duplex, switched gigabit links. Several topics discussed in this chapter warrant further review to facilitate a better understanding of Ethernet. These include: the concepts of network topology and architecture (Chapter 2); physical layer concepts, especially multiplexing methods and fiber-optic cable (Chapter 4); data link layer concepts, including IEEE's MAC and LLC sublayers, framing, flow control, and QoS; Ethernet switches (Chapter 6); SONET (Chapter 7); and ATM (Chapter 14). In the next chapter, we present many of these same concepts but apply them to token ring networks.

Chapter 9

Token Ring

In this chapter we present an overview of token ring networks, IEEE 802.5. Unlike Ethernet/802.3 LANs in which nodes contend for media access, token ring LANs use a token-passing scheme; that is, media access in token ring LANs is controlled by the possession of a token. We discussed general issues relating to token passing, including a comparison of random access and token passing protocols, in Chapter 5, which you might want to review before proceeding. In addition to token ring, the IEEE standards include a second token passing protocol called token bus. Although token bus use is uncommon, we present a brief comparison of the two token passing schemes for completeness. An outline of the major topics we present in this chapter follows:

- Definition and Operation (Questions 1–2)
- Frame Formats (Question 3)
- Priority Scheduling (Question 4)
- Monitor Stations (Questions 5–6)
- Physical Layer Issues (Questions 7–8)
- Token Ring vs. Token Bus (Question 9)
- Advantages and Disadvantages of Token Ring Networks (Question 10)
- Switched, Dedicated, and Full-Duplex Token Ring (Questions 11–14)
- High-Speed Token Ring (Question 15)
- Token Ring's Future (Question 16)

1. What is a token ring network?

A *token ring* network is a local area network technology based on a token-passing protocol for media access control. (See Chapter 5 for more information about token passing protocol concepts.) Data frames on a token ring network are transmitted from node to node, in either a clockwise or counterclockwise direction, over a point-to-point link. A token ring LAN is implemented either as a logical ring using a physical ring topology (Figure 9.1), or as a logical ring structure arranged in a physical star configuration (Figure 9.2). It is also

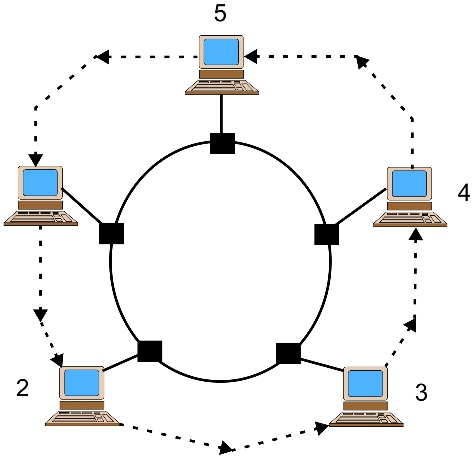


FIGURE 9.1 A token ring network consists of a logical ring implemented in a physical ring topology. A token, which is a special frame, and data are transmitted in a point-to-point manner from one lobe (called a lobe) to the next. The direction of circulation is fixed and either clockwise or counterclockwise (but not both). For example, on a counterclockwise rotating ring, if lobe 3 has a "free" token and wants to send data to lobe 2, data frames must circulate the ring in the order 3-4-5-1-2. On a clockwise rotating ring, though, the transmission order is 3-2.

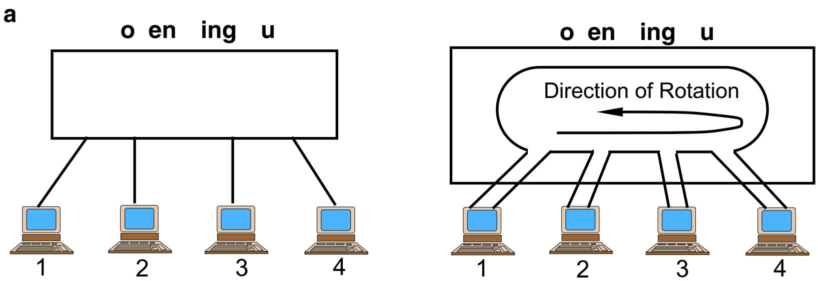


FIGURE 9.2 A typical token ring network consists of lobes connected to a hub in a physical star configuration (a). Internally, lobes are actually interconnected via a logical ring (b).

possible to extend the configuration in Figure 9.2 to include a ring consisting of several interconnected hubs. This is shown in Figure 9.3, which describes the arrangement in the special language of token rings. From Figure 9.3, note the following: Token ring hubs are called *multistation access units* (MAUs); nodes are called *lobes*; the distance between MAUs is called the *main ring length*; and the distance between an MAU and its lobes is called the *lobe length*.

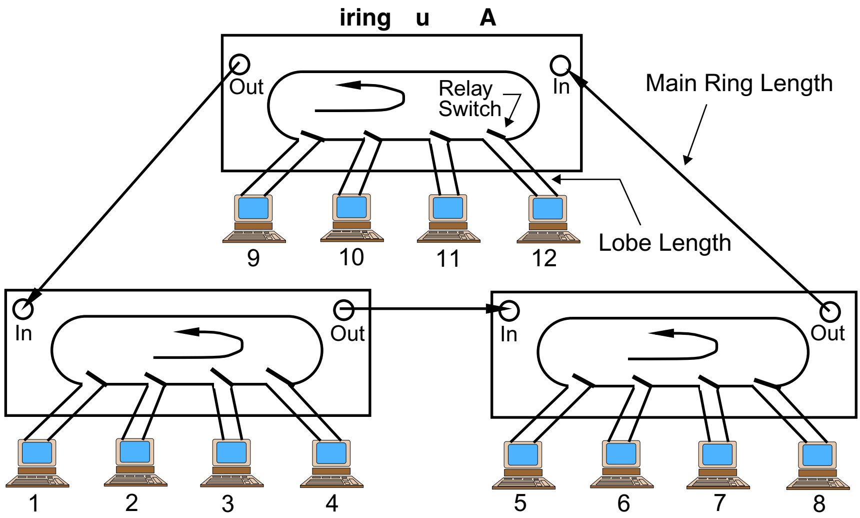


FIGURE 9.3 An example of a typical token ring network configuration. Token ring hubs are called multistation access units (MAUs), and nodes are called lobes. The main ring length is the distance between MAUs, and the lobe length is the distance between an MAU and its lobes. Physically, lobes are connected to an MAU in a star configuration. Within an MAU, however, a logical ring topology exists. Lobes are connected to the ring using an IBM Data Connector, which enables lobes to be removed without disrupting the ring. MAUs also can be interconnected using special "ring in/ring out" ports, which preserve the ring structure. Note the presence of relay switches within each hub. Relay switches (also called bypass switches) are used to maintain the integrity of the ring in the event of lobe failure. For example, if lobe 12 stops working or if there is a break in the cable connecting lobe 12 to the ring, the ring is broken. In such instances, the relay switch closes, thus preserving the ring.

In a typical token ring connection, lobes are physically connected to an MAU in a star configuration, but there is a logical ring topology within the MAU. Lobes are connected to the ring using an IBM Data Connector, which enables lobes to be removed without disrupting the ring. MAUs also can be interconnected using special "ring in/ring out" ports, which preserve the ring structure. Special relay switches (also called bypass switches) are used to maintain the integrity of the ring in the event of lobe failure. For example, in Figure 9.3, if lobe 12 stops working or if there is a break in the cable connecting lobe 12 to the ring, the ring is broken. In such instances, the relay switch closes, thus preserving the ring.

Token ring networks are defined by IEEE 802.5, which is based in part on IBM's set of token ring specifications. IBM is the primary vendor associated with token ring LANs. Although IBM's token ring specifications differ from the official IEEE specs, most people usually speak about the IBM specs when discussing token ring LANs.

2. How does a token ring network work?

Access to the network is controlled by a special "token" frame, which circulates around the ring when all lobes are idle. A token frame comprises a one-byte start frame delimiter,

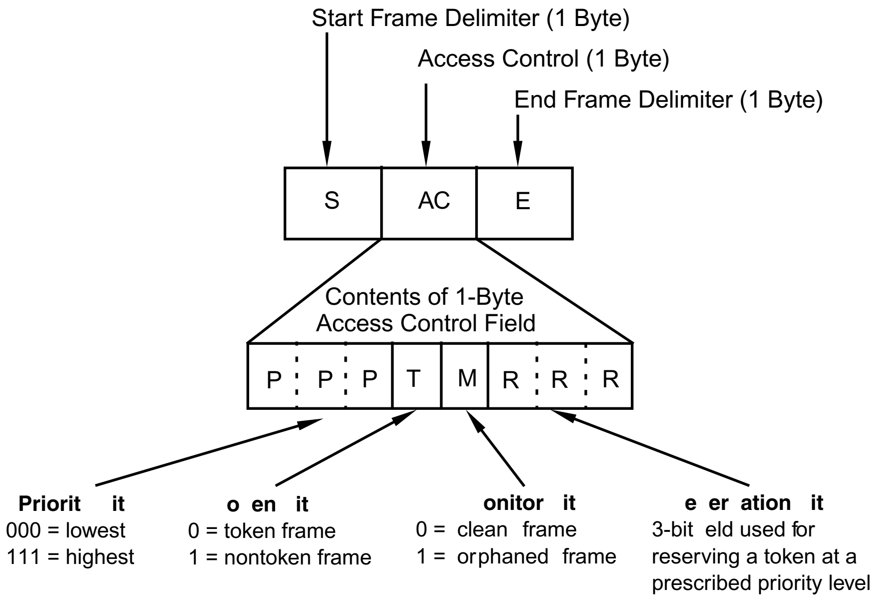


FIGURE 9.4 Format and contents of an IEEE 802.5 token frame.

a one-byte access control, and a one-byte end frame delimiter. This is shown in Figure 9.4.) If the token bit of the access control field is set to 0, the token is considered "free" or idle. Only one free token is permitted on the ring, and the lobe that has the free token controls the ring and is permitted to transmit data. Thus, only one lobe at a time can transmit data on a token ring network.

A lobe that possesses the free token and has data to transmit changes the access control field's token bit to 1 and then augments the token frame by including a frame control field, destination and source addresses, user data, a CRC checksum, and a frame status field. In other words, the token frame is transformed into a data frame (Figure 9.5), which is then transmitted around the ring from lobe to lobe.

When a lobe receives a data frame, it will identify the frame as data and not a token because the token bit of the access control field is set to 1. The lobe will also check the frame's destination address. If the lobe is not the intended recipient, it then places the frame back on the ring. When the intended destination node receives the frame, it copies the frame into memory. The lobe then sets the A and C bits of the frame status field to 1 and places the entire frame back on the ring where it will continue to be transmitted from lobe to lobe. When the sending lobe receives the frame after the frame's complete pass around the ring, it examines the frame status field. If the A and C bits are set to 1, then data transmission was successful. If not, then the frame is retransmitted. In the case of a successful transmission, the lobe removes the data from the frame and changes the token bit to 0. Thus, the data frame is transformed back to a token frame. This free token is then placed on the ring and sent to the next lobe in line. The lobe that has possession of the

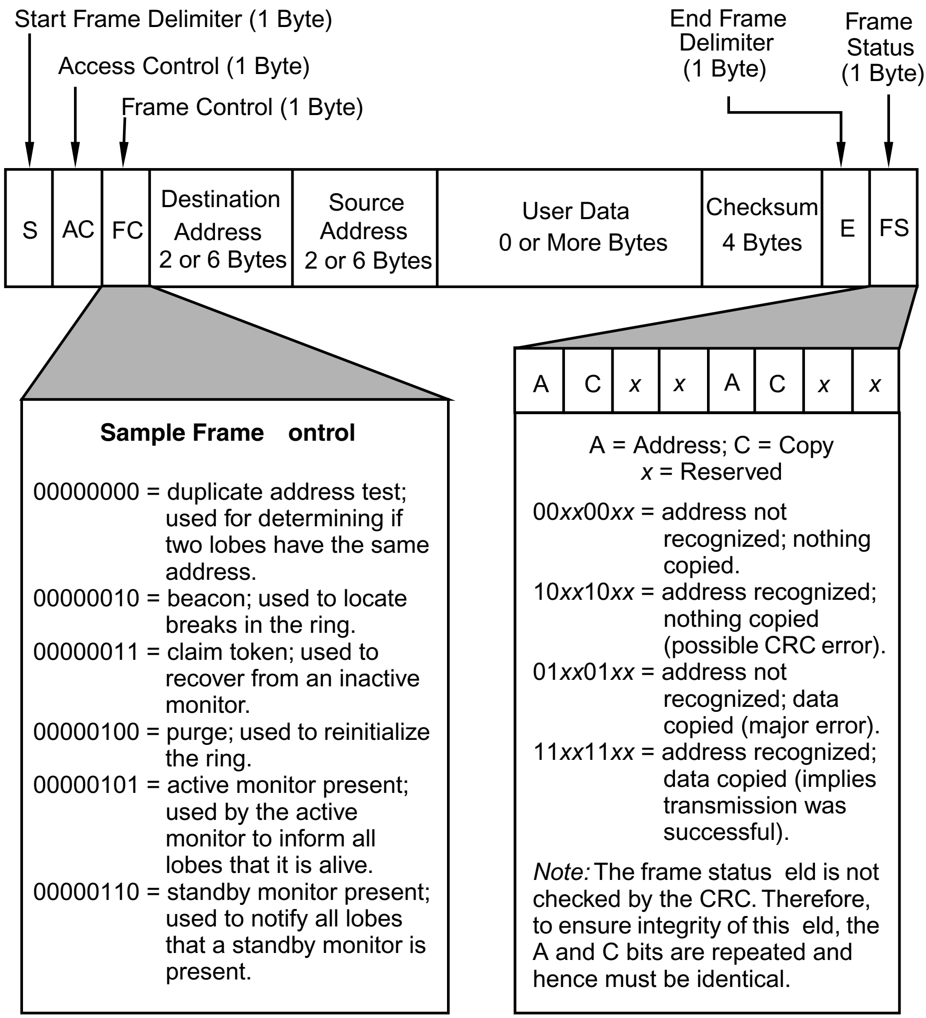


FIGURE 9.5 Format and contents of an IEEE 802.5 frame. Note that the S, AC, and E fields comprise the token frame. Thus, an IEEE 802.5 data frame is really a modified token frame that includes the frame control, destination and source addresses, user data, checksum, and frame status fields.

token is permitted to continue transmitting data until it has no more data to send or its time limit expires, whichever comes first. In IEEE 802.5, the length of time a lobe can possess a token—called the token-holding time—is 10 ms.

To illustrate this operation, consider the sample LAN in Figure 9.1. Let's assume that lobe 2 has data to transmit to lobe 5, and lobe 4 has data to transmit to lobe 1. Let's further assume that lobe 2 has the free token. Lobe 2 modifies the free token to "busy," inserts its data, and sends the frame to lobe 3. Lobe 3 examines the frame, determines it is not the intended recipient, and returns the frame to the ring where it is transmitted to lobe 4. Lobe 4 grabs the frame off the ring and sees that it is data and not a token. Hence, it cannot transmit its data. Lobe 4 also determines that it is not the intended recipient so it returns the frame to the ring where it is transmitted to lobe 5. Lobe 5 receives the frame and ascertains that its destination address matches that of the frame's. The lobe then makes a copy of the frame and stores it in memory. If the frame is valid and the data successfully copied, lobe 5 then sets the A and C bits to 1 and returns the frame to the ring. (It also processes the data it receives.) Lobe 1 receives the frame, examines it, and returns it to the ring. The frame, having completed a full pass around the ring, is now received by lobe 2. Seeing that the A and C bits were properly set, lobe 2 strips the data from the frame. If its token-holding time has elapsed, it then resets the token bit to 0 and places the token frame on the ring. Otherwise, it transmits another frame of data. When lobe 2 returns the free token to the ring, it is transmitted to lobe 3. Since lobe 3 does not have any data to transmit, it returns the token to the ring for transmission to lobe 4. Since lobe 4 has data to transmit, it transforms the token into a data frame and the data transmission process begins again with lobe 1 being the intended recipient.

3. Could you explain the contents of the access control field?

Sure. The access control field consists of eight bits: three priority bits (P), a token bit (T), a monitor bit (M), and three reservation bits (R). *Priority bits* can be set from 0 (lowest) to 7 (highest). Thus, a token ring network has eight possible priorities (000 to 111) relative to data transmission. The use of these bits is discussed later in the chapter. As indicated earlier, the *token bit* identifies the frame as a token (T = 0) or data (T = 1). If the token bit is set to 0, the frame is considered a "free" or idle token. Only one free token is permitted on the ring, and the lobe that has the free token controls the ring and is permitted to transmit data. Thus, unlike Ethernet/802.3 networks, only one lobe at a time can transmit data on a token ring network. The *monitor bit* is used by one lobe, called the *monitor station* or active monitor, to oversee the status of the token. When a lobe transmits data, or when the token is idle, M is set to 0. When the monitor station receives a data frame, it sets M to 1. If the monitor station now receives a data frame that has M set to 1, it knows that the transmitting lobe did not strip the data off the frame after the frame completed a full pass around the ring. The monitor station then removes this "orphaned" frame from the ring and issues a new token. The monitor station is discussed later in the chapter. The *reservation bits* are used for reserving a token at a particular level of priority. These, too, are discussed later in the chapter.

4. How do the priority and reservation bits in the token frame function?

As indicated earlier, a token can have several different priority levels, ranging from 0 (lowest) to 7 (highest). When a lobe receives a free token, it must first compare the priority value contained within the token to the priority of the data it has to transmit. If a lobe's data has a priority level equal to or greater than that of the token's, then the lobe may transmit its data. If the token's priority is higher than the lobe's data, then the lobe cannot transmit its data and must pass the token to its neighbor lobe. During a particular transmission, a lobe may reserve the token at a higher level than the token's current level. By doing so, it preempts the lobe that is currently transmitting data. To illustrate this, consider the token ring LAN in Figure 9.2(b). Let us make the following eight assumptions: (a) the ring rotates counterclockwise; (b) lobe 2 has data to transmit to lobe 1; (c) the priority level of lobe 2's data is 0 (e.g., an e-mail message); (d) lobe 2 has just received a free token with a priority level of 0; (e) lobe 3 has data to transmit to lobe 4; (f) lobe 3's data priority is 1; (g) lobe 4 needs to transmit a video frame to lobe 3; and (h) the priority level of lobe 4's data is 5. Given these assumptions, the operation of the ring is as follows:

1. Lobe 2 transforms the token frame into a data frame.
2. Lobe 2 transmits the data frame to lobe 3.
3. Lobe 3 examines the frame, but takes no action because it is not the recipient. It also cannot transmit its data because the token is busy. Lobe 3 passes the frame to lobe 4.
4. Lobe 4 receives the frame. Because it has data with a priority of 5, lobe 4 makes a reservation at priority 5 by setting the three reservation bits of the token's access control field from binary 000 to 101 (i.e., 5). Lobe 4 passes the frame to lobe 1.
5. Lobe 1 receives the frame. Since it is the recipient, it saves the source address (lobe 2), computes the 32-bit checksum, changes the frame's frame status field to reflect that it received the data, and then transmits the frame to lobe 2.
6. Lobe 2, seeing that the transmission was successful, strips the data from the frame. Normally, lobe 2 would continue transmitting data until it has no more data to transmit or until its time limit expired. It would then issue a free token with a priority level of 0 to lobe 3. However, because the frame that was returned has a reservation priority of 5, lobe 2 cannot transmit any more data frames because their priority level is 0, which is less than the reserved priority of 5. As a result, lobe 2 must issue a new free token with a priority level of 5 and transmit it to lobe 3.
7. Lobe 3 receives the free token but is not permitted to transmit its data because the data's priority level is less than 5. It passes the token to lobe 4.
8. Lobe 4 receives the token, changes it to "busy," and transmits its data to lobe 1.
9. After the frame is returned to lobe 4, if there are no additional frames to transmit, or if the time limit expired, lobe 4 transmits a free token to lobe 1; the priority level remains at 5, though.
10. Lobe 1 passes the token to lobe 2.
11. Lobe 2 receives the token and notes that the token is "free" and the priority level is the same one it used when it last issued a new token. Lobe 2 reissues a new token at its previous priority level of 0. (The lobe that upgraded the priority level of the token is also responsible for reestablishing the previous level after all higher-priority data frames are transmitted.)

IEEE 802.5's priority scheduling is excellent for transmitting time-sensitive data such as real-time video or voice. Any lobe that has data frames with a higher priority than that of the frame currently being transmitted can reserve the next token at this higher level when the current token and frame are passed to it. As demonstrated in the illustration, when the next token is issued it will be at this higher level. Furthermore, no other lobe is permitted to transmit data unless its data frames have a priority level equal to or greater than the newly issued one. As beneficial as this scheme is, though, token ring priority makes it possible for lobes with high priority data to prevent lobes with low priority data from ever accessing the medium.

5. What do you mean by an "active monitor"?

A token ring network employs a monitor station to oversee the ring and ensure that it is functioning properly. This monitoring lobe is called the *active monitor*; which is usually a high-priority lobe. All other lobes, known as *standby monitors*, monitor the active monitor. If the active monitor becomes disabled, a contention protocol is invoked among the standby monitors to elect a new active monitor.

6. What does an active monitor do?

When a token ring network is first started, the active monitor generates the first token and begins the process that enables each lobe to learn the address of its neighbor that is next in line (called the "downstream" lobe). During the operation of the ring, the active monitor performs several tasks, including monitoring the ring for valid frame transmissions, maintaining the ring's master clock, and ensuring there are proper delays in the ring. The active monitor also is sensitive to two possible error conditions. The first is a lost token. If no token is detected after a predetermined amount of time expires, the active monitor assumes the token is lost and issues a new one. The second possible error is a persistently busy token. To check for this condition, a special bit within the token is set. If this bit is still set when the token is returned to the active monitor, it assumes the source station did not remove the data from the network. It then changes the token to "free" and passes it to the next lobe. One task an active monitor cannot do is detect breaks in the ring, which can occur if a lobe fails or the cable connecting the lobe to the ring is broken. To recover from either of these cases, special relay switches (also called bypass switches) are used. In case of a host failure, a bypass switch can be closed either manually or automatically, effectively removing the dead lobe from the ring. This is illustrated in Figure 9.3.

7. So far we have discussed the data link layer of token ring networks. Tell me about the physical layer.

At the physical layer, IEEE 802.5 supports STP, UTP, coaxial, and fiber-optic cable. STP cable has a 150-ohm impedance; UTP cable has a 100-ohm impedance. The topology is usually star-based using token ring hubs (called wiring concentrators or multistation access units—MAUs), with hubs being interconnected to form a main ring path (see Figure 9.3.) Data rates include 4 and 16 Mbps, although some variations can include 20 and 40 Mbps. In a 4-Mbps token ring, lobes can transmit only one frame at a time during a single transmission. Lobes connected to token rings with higher data transmission rates, however, can transmit multiple frames during a single transmission.

Maximum cable lengths for lobe connections (called lobe lengths) are 100 m if IBM Type 1 or 2 cable is used, 66 m for Types 6 and 9, and 45 m for UTP. (See Chapter 4 for an explanation of IBM Type cables.) Maximum cable lengths for hub interconnections depend on several factors, including the number of repeaters used, the number of hubs, and so forth. Some general guidelines are as follows: 200 m if using Type 1 or 2 cable; 120 m if using Type 3 cable; 45 m if using Type 6 cable; and 1 km if using fiber-optic cable. Type 1 and Type 3 networks can operate at 4 or 16 Mbps. Also, with STP cable, 260 devices can be connected to a single token ring network; with UTP cable (Category 3, 4, or 5), only 72 devices can be connected to the ring.

8. Does token ring also use Manchester encoding as in Ethernet/802.3?

No. It does use a form of Manchester encoding called *differential Manchester encoding*. Manchester and differential Manchester encoding are similar in that each bit-period is partitioned into two intervals and a transition between "high" and "low" occurs during each bit-period. The difference between the two techniques is the interpretation of this transition. In Manchester encoding (see Figure 8.1), a 1 bit represents a low-to-high mid-bit transition, and a 0 bit represents a high-to-low mid-bit transition. In differential Manchester encoding, the interpretation of these low-to-high and high-to-low mid-bit transitions is not as simple—they are a function of the previous bit-period. A low-to-high transition could be a 0 or a 1, depending on the value of the previous bit-period. More specifically, the presence of a transition at the beginning of a bit period is coded 0, and the absence of a transition at the beginning of a bit period is coded 1. Note also that for a token ring network to achieve its maximum bandwidth, its clock speed must be twice the transmission rate. Thus, a 16 Mbps token ring must have a clock speed of 32 MHz.

9. You mentioned token bus in the introduction as another token-passing LAN. How does it compare to token ring?

Token bus is defined in IEEE 802.4. A token bus network is characterized as a logical ring on a physical bus. Physically, the network resembles a bus topology, but logically, the network is arranged as a ring with respect to passing the token from lobe to lobe. An illustration of a token bus network is shown in Figure 9.6. Although token passing is point-to-point (i.e., from lobe-to-lobe), data transmission is based on broadcasting. For example, in Figure 9.6, if lobe 44 has the token and wants to send data to lobe 70, it simply places data frames on the bus. The broadcast nature of the bus topology causes all lobes to hear the transmission, but only lobe 70 reads and processes the data. After lobe 40 has completed its transmission it then passes the token to lobe 32. Thus, from the perspective of actual data transmission, IEEE 802.4 is similar to that of IEEE 802.3—it is based on broadcasting. However, medium access control is similar to IEEE 802.5.

At the physical layer, IEEE 802.4 uses 75-ohm coaxial cable or fiber-optic cable. If coax is used, data rates include 1, 5, and 10 Mbps. If fiber-optic cable is used, data rates are 5, 10, and 20 Mbps. IEEE 802.4 also supports priority scheduling and can be configured to operate in one of four priority modes, 0 (lowest), 2, 4, and 6 (highest). As is the case with IEEE 802.5, this priority scheme makes it possible to allocate network bandwidth to high priority (level 6) data such as video, digitized voice, and multimedia applications. Lower priority data get transmitted if sufficient bandwidth is available.

IEEE 802.4 is not a popular LAN protocol; it is primarily used for control of industrial and factory automation processes. It also is the basis of the General Motors Manufacturing Automation Protocol (MAP), which supports real-time applications.

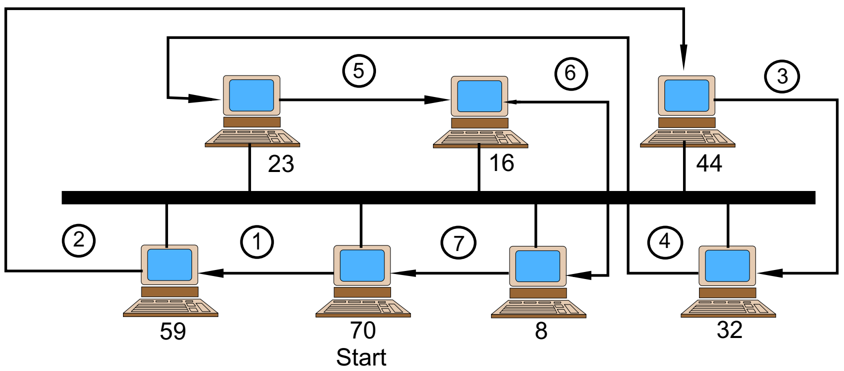


FIGURE 9.6 A token bus network physically resembles a bus topology; logically, however, it is a ring. A token is transmitted from lobe to lobe using network addresses and occurs in descending order. The lobe that possesses the token is permitted to transmit data. The lobe order in the figure is 70-59-44-32-23-16-8.

10. What are some of the advantages and disadvantages of token ring networks?

Some of token ring's advantages include its ability to run on many different media types, its efficient use of bandwidth (e.g., if packet size is 1,000 bits, efficiency is 99 percent), its stable behavior during high load times, its deterministic nature (it has a known upper bound on channel access), and its priority scheme that enables lobes with high-priority data to reserve the network for data transmission. Disadvantages include the need for special recovery procedures when the network fails, difficulty in configuring new hosts to an established LAN, and in the case of priority scheduling, the susceptibility of low priority lobes to increased delays in accessing the network.

11. I realize that token ring is not as popular as Ethernet. However, is there a "second generation" token ring as there is with Ethernet?

Yes. Token ring has benefited from some of the technological advances made for Ethernet/802.3. These include switched token ring, dedicated token ring, full-duplex token ring, and 100-Mbps token ring. A gigabit token ring specification is also in development.

12. Let's start with switched token ring. I can understand how switches benefit traditional Ethernet/802.3 LANs. But how do token ring LANs benefit from switching technology? They don't have collisions.

You are right. Ethernet/802.3's MAC layer protocol (CSMA/CD) is contention-based and highly susceptible to performance problems during periods of high activity. Token ring's token-passing protocol, however, is inherently deterministic and quite stable during peak load times. Token ring networks do not experience the same type of performance problems as those found in Ethernet/802.3 networks. A token ring network's performance, for example, does not begin to degrade until its utilization rate approaches 90 percent, and in many cases can achieve a bandwidth utilization rate as high as 95 percent. So, yes, Ethernet/802.3 networks are a better candidate for switching technology than token ring networks because the former can benefit more from the type of performance boosts available from switches. With the introduction of client/server applications, imaging, multimedia, and the consolidation of servers, however, token ring networks—including 16-Mbps token ring, which has higher bandwidth than 10-Mbps Ethernet—are experiencing congestion and performance problems similar to those of Ethernet/802.3 networks. To help alleviate these problems, several vendors have ported the switched technology originally developed for Ethernet/802.3 to token ring.

13. Are the switches like those found on Ethernet networks?

Yes. Token ring switches are similar to Ethernet switches in that they are capable of supporting workgroups, desktop, or backbone connections, and they use either store-and-forward or cut-through technology to forward frames from one network to another. In a workgroup environment, individual token ring networks are interconnected via the switch. In this capacity, the token ring switch acts like a multiport source routing bridge that is connected to multiple ring numbers. (See Chapter 6 for information about the concept of source routing and source routing bridges.) With private connections, individual stations have a full, dedicated 16 Mbps link to the switch. Lobes do not share a 16 Mbps channel with other lobes. Finally, as a backbone switch, an organization's entire network backbone is incorporated into the switch, which resembles the collapsed backbone concept discussed earlier for Ethernet/802.3 networks.

In large token ring networks, workgroups usually feed into the backbone. This configuration promotes congestion, particularly if the workgroups are 16-Mbps networks and the backbone also is operating at 16 Mbps. Although source routing bridges or routers are used to segment the network into smaller components, this introduces additional delay into the network. Consequently, one of the primary applications of token ring switches is at the backbone where they replace source routing bridges. This application enables organizations to consolidate local servers into centrally located "super" servers since the backbone is now the switch itself. An illustration is shown in Figure 9.7.

Some token ring switches also provide support for *virtual rings* and source route transparent bridging. With virtual rings, several individual token ring networks are collectively viewed as a single (virtual) ring. Although these individual networks are indeed separate and connected to different switch ports, source routing between lobes connected to two separate networks is now transparent because the networks appear to be on the same (virtual) ring. Virtual ring support also eliminates the need to configure new ring numbers when connecting new token ring networks to a switch because the new rings appear to the network as part of the original ring. An illustration of the virtual ring concept is given in Figure 9.8.

14. I see. And with private switches in the configuration, token ring LANs can have lobes connected to dedicated segments and operate in full-duplex mode just like Ethernet. Is this correct?

Yes. As is the case with switched Ethernet, a token ring switching environment advances the concept of dedicated token ring, which enables lobes to have access to a full 16-Mbps

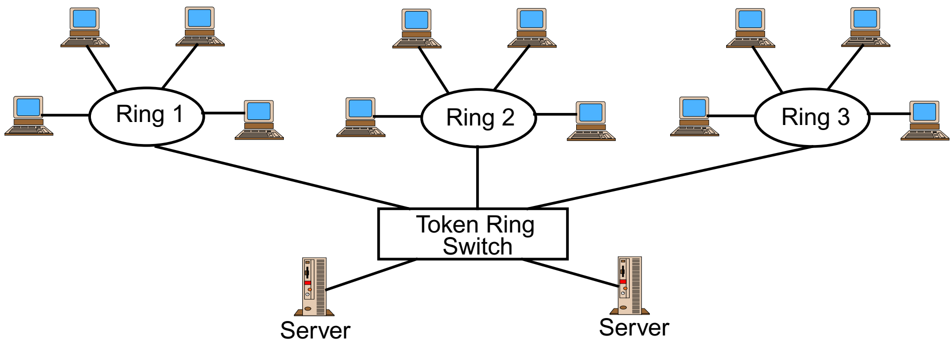


FIGURE 9.7 Incorporated within the backbone, a token ring switch acts as a multiport source routing bridge and enables large networks to be partitioned into smaller segments. The net effect is a collapsed backbone, which reduces both overall network traffic and traffic at the server ports. (Compare this configuration to the one shown in Figure 6.16.)

channel instead of sharing the segment with other lobes. Dedicated token ring does not require any new hardware or software either—all you do is connect a lobe (via its NIC) to a port on the switch. In a dedicated switched environment, only two lobes are involved in the transmission and receipt of data. Consequently, dedicated lobes can transmit data whenever they have data to send. A new method for dedicated token ring, called Transmit Immediate (TXI), also is defined by IEEE 802.5. In TXI, dedicated lobes are not permitted to begin transmitting data until they possess a token. Assigning lobes, especially those that transmit and receive a high volume of traffic, to dedicated links can improve overall network performance since this further segments a network.

Although dedicated switched ports enable lobes to have their own private network segments, they are still operating in half-duplex. By installing additional software drivers or upgrading the firmware of NICs, lobes can support full-duplex token ring, which provides 16 Mbps of bandwidth in both directions. This enables stations to transmit and receive data at the same time. Stations such as "super" servers, which are the source of high volume traffic, are excellent candidates for full-duplex token ring. The combination of full-duplex token ring with a dedicated switched environment can dramatically boost the performance of a token ring network (Figure 9.9.)

15. What's the status of 100 Mbps and gigabit token rings?

The IEEE 802.5 working group has created three separate initiatives for high-speed token ring (HSTR): IEEE 802.5t, which is HSTR over Category 5 UTP cable; IEEE 802.5u, which is HSTR over fiber-optic cable; and IEEE 802.5v, which is gigabit token ring. The first two specifications were completed in 1990. The third specification was ratified in 2000. HSTR is a switched-only 100-Mbps technology. Deployment is restricted to the backbone, interswitch links, and server connections. There are no shared links as with classic token ring. HSTR uses the same MAC sublayer as 4/16-Mbps token ring, and incorporates autonegotiation into its NICs for 4/16/100 Mbps autosensing.

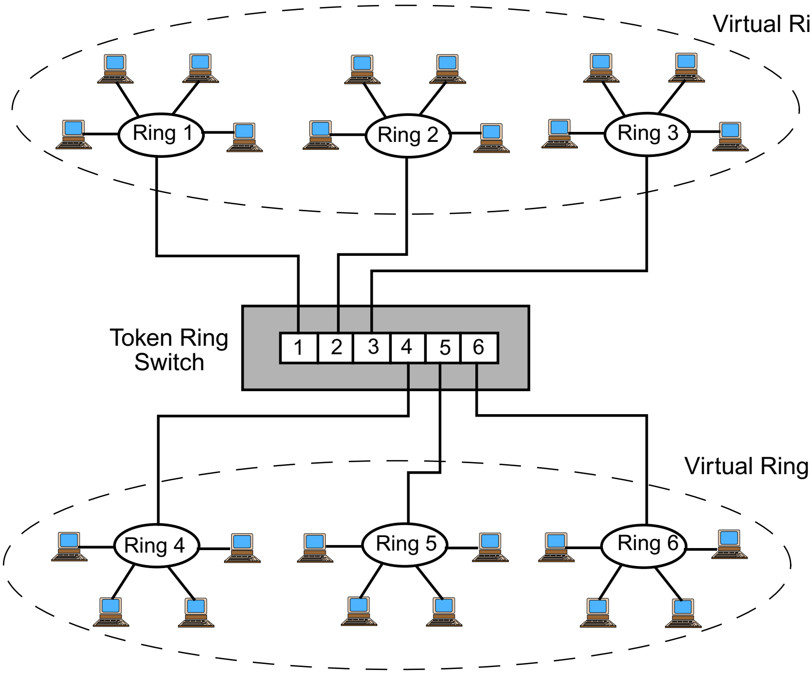


FIGURE 9.8 Token ring switches that support virtual rings enable multiple independent rings to be viewed as a single ring. For example, rings 1, 2, and 3 are viewed by the switch as a single ring, namely, virtual ring 1. Support for virtual rings eliminates the need to configure new ring numbers when a new ring is added to the switch. Additionally, since the networks comprising a virtual ring are considered a single ring, transparent bridging is in effect. Thus, transparent bridging exists for the rings connected to switch ports 1, 2, and 3 and for the rings connected to switch ports 4, 5, and 6. Source routing bridging, however, is necessary for exchanging data between virtual rings 1 and 2.

16. Why all this fuss about a technology whose market share pales in comparison with that of Ethernet? Does token ring really have a future?

These are tough questions. Compared with Ethernet, the installed base of token ring networks and the market for token ring products are indeed relatively insignificant. In fact, in a latest survey, 98% of all LAN switches purchased in 2001 were Ethernet switches. Nevertheless, the vendors that comprise the High-Speed Token Ring Alliance (HSTRA) feel an obligation to protect the existing market and established users by advancing token ring technology to be on par with that of Ethernet. At the very least, HSTR gives users and managers of token ring technology hope that token ring is not being ignored. Unfortunately, all is not well with the political and marketing sides of HSTR. Cisco, Cabletron, and Texas Instruments—all charter members of HSTRA—withdrew from the organization. Furthermore, Cisco reported that it will not develop any products that are IEEE 802.5

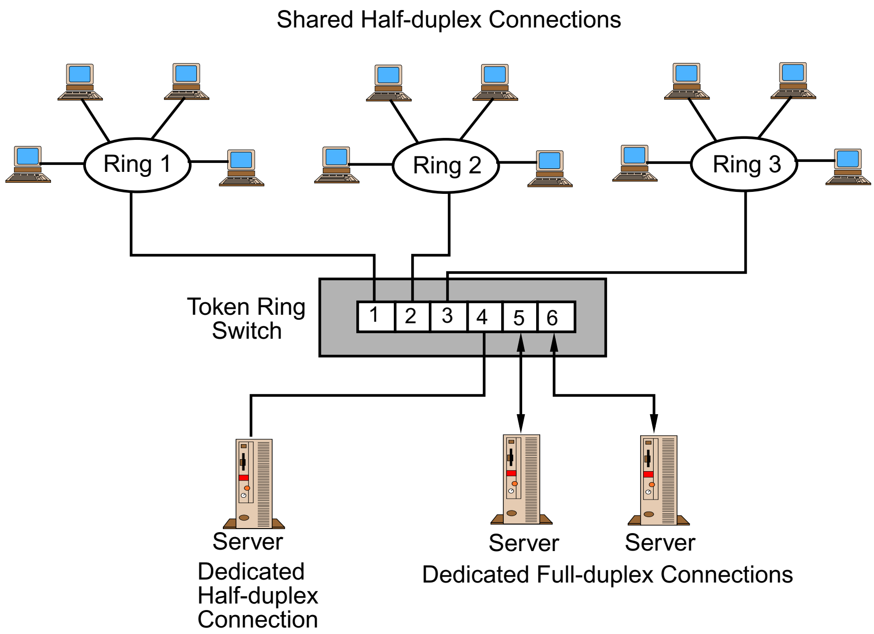


FIGURE 9.9 Dedicated token ring connections provide lobes with private network segments instead of shared segments. Incorporating support for full-duplex token ring into a dedicated lobe's NIC, bandwidth to the lobe is doubled to 32 Mbps since the lobe can simultaneously transmit and receive data. A token ring switch, in combination with dedicated and full-duplex support, dramatically increases overall network performance

compliant, but instead will develop and market its own proprietary HSTR products. Many token ring managers also have indicated via surveys that they intend to migrate to Fast Ethernet, ATM, or Gigabit Ethernet and hence will not purchase HSTR products. So, although HSTR appears to be a promising IEEE technology standard that will serve as a smooth migration from 4/16-Mbps token ring, vendor support is problematic at best. Additional information about HSTR can be found at the High-Speed Token Ring Alliance's home page at http://www.hstra.com.

END-OF-CHAPTER COMMENTARY

This chapter presented an overview of token ring networks. Two related chapters are Chapter 10, FDDI, which uses a token ring technique similar to the IEEE 802.5 specification, and Chapter 14, ATM, which is viewed by various token ring managers as the logical migration path for classic token ring LANs. In addition to these chapters, you also might want to review Chapter 5, which contains a general discussion and comparison between random accessing protocols and token-passing protocols.

Chapter 10

Fiber Distributed Data Interface (FDDI)

In this chapter we present information about a network technology called fiber distributed data interface (FDDI). FDDI employs a ring topology with fiber-optic cabling as its physical layer medium. We also discuss a "sister" standard called copper distributed data interface (CDDI), which uses copper instead of fiber. An outline of the terms and concepts we define and discuss follows:

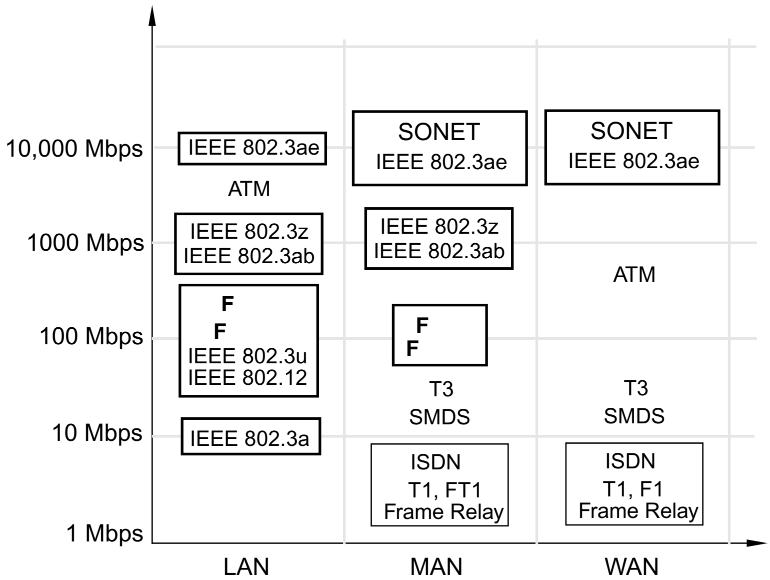
- General Information (Questions 1–6)
- Physical Layer Issues (Questions 7–10)
- Data Link Layer Issues (Questions 11–13)
- Operation and Configuration Issues (Questions 14–22)
- FDDI and Ethernet/802.3 (Questions 23–24)
- FDDI and ATM (Question 25)
- CDDI (Questions 26 through 28)
- Future of FDDI (Question 29)

1. What is FDDI?

FDDI stands for *fiber distributed data interface*. FDDI networks are described by ANSI standard X3T9.5 created in 1986 for interconnecting computer systems and network devices typically via a fiber ring topology at 100 Mbps. Figure 10.1 shows where FDDI networks fit in the hierarchy of network architectures. (For information about the other network architectures listed in this figure, see the appropriately named chapters.)

2. In what capacity are FDDI networks used?

FDDI's bandwidth gives it considerable flexibility in how an organization allocates its resources. Its applications include directly connecting workstations and servers in workgroups, and serving as a high-speed backbone to connect other networks in a building, in a campus environment, or in a city. An example of the first application is the interconnection of high-speed servers to other high-speed servers. For instance, a very large video server system cannot be effectively connected to a broadcast video server at 10-Mbps Ethernet/802.3



¥T1, FT1 (Fractional T1), T3: (Chapter 7)
¥ SONET: Synchronous Optical Network (Chapter 7)
¥ IEEE 802.3a:10-Mbps Ethernet (Chapter 8)
¥ IEEE 802.3u:100-Mbps Ethernet Fast Ethernet (Chapter 8)
¥ IEEE 802.12:100-VG-AnyLAN Alternative 100 Mbps Ethernet (Chapter 8)
¥ IEEE 802.3z:1000-Mbps Ethernet Fiber-based Gigabit Ethernet (Chapter 8)
¥ IEEE 802.3ab:1000-Mbps Ethernet Copper-based Gigabit Ethernet (Chapter 8)
¥ IEEE 802.3ae:10,000 Mbps Ethernet 10 Gigabit Ethernet (Chapter 8)
¥ ISDN:Integrated Services Digital Network (Chapter 11)
¥ Frame Relay : (Chapter 12)
¥ ATM: Asynchronous Optical Network (Chapter 14)

FIGURE 10.1 Comparing FDDI to other network technologies relative to bandwidth.

or token ring speeds, but at FDDI's data rate of 100 Mbps, the connection is adequate for server transmission. As a backbone network, FDDI interconnects network devices such as routers, bridges, switches, and concentrators to create a large network environment consisting of smaller networks. FDDI networks are not used for WANs where network radii typically exceed 100 km. FDDI was very popular in networks that required 100-Mbps capability prior to 1996.

3. You say FDDI *was* popular prior to 1996. What happened?

When first introduced, FDDI's data transmission rate of 100 Mbps was ten times faster than 10-Mbps Ethernet/802.3 systems. Since 1996, however, 100-Mbps Ethernet/802.3 (Fast Ethernet), 1000-Mbps Ethernet/802.3 (Gigabit Ethernet), and 10,000-Mbps Ethernet (10 Gigabit Ethernet) technologies have displaced new FDDI installations.

4. So why should I bother studying about FDDI networks?

Simple. You are going to encounter FDDI networks in just about any larger company. Additionally, many telecommunications companies operate metropolitan area networks (MANs) that consist largely of FDDI or FDDI emulations over faster networks such as the synchronous optical network, SONET (see Chapter 7).

5. What makes FDDI special compared to other 100-Mbps networks?

First, FDDI can be configured as two independent, *counterrotating* ring networks, called a Class A configuration (Figure 10.2). This greatly increases network reliability. If the physical topology of the network is designed such that both fiber paths for both networks are physically diverse (geekspeak for putting the two fiber paths for the two networks in completely different physical locations so that one backhoe does not kill both networks at the same time while trenching up the lawn), then it is very difficult to destroy the network with a single or even multiple fiber cuts to the network cable plant. Second, FDDI is much less susceptible to a network disruption; it has the ability to "self-heal" if the ring topology is cut in a single spot. This is called *autowrapping*. The break in the active ring is corrected by establishing a loopback connection to the inactive ring. This creates a single virtual ring and allows the FDDI network to continue to function at full speed (Figure 10.3). Third, FDDI transmits information in frames up to 4500 octets (bytes), which increases network efficiency and lowers protocol overhead. Finally, FDDI encodes data quite a bit differently from other types of networks to increase transmission efficiency.

6. What does "self-healing" (autowrapping) really mean?

Self-healing resembles the expression "Physician, heal thyself." The network hardware is capable of detecting a fiber path failure between connection points on the rings. Since there are two fibers (one transmitting clockwise, the other counterclockwise) in the configuration, the stations that detect the failure join the two rings together and effectively "wrap" together to make a single fiber network twice as long as the original two-fiber network (hence the term autowrapping). If the network fiber path is destroyed in two different spots, the result is two healed network rings. So, self-healing is very beneficial for a single location failure because the network continues to function as illustrated in Figure 10.3. In the case of two failures, the network components that will maintain the connectivity are determined by which nodes need to talk to whom and how the network was designed. Self-healing is not for all situations in which the network could be disrupted.

7. Obviously, FDDI's physical layer is based on fiber-optic cable. What fiber types are allowed and what are the rules?

FDDI uses 62.5/125-µm multimode fiber, 85/125-µm multimode fiber, or 8.7/125-µm single-mode fiber. Other specifications include 50/125 µm and 100/140 µm are also specified. Smaller fibers allow higher speeds, but also cause higher connector loss. Further, the fiber must be specified for light transmission at the 1300-nm wavelength. Since most fiber plants transmit at 850 nm, 1300 nm, or 1550 nm, finding compliant fiber is usually not a

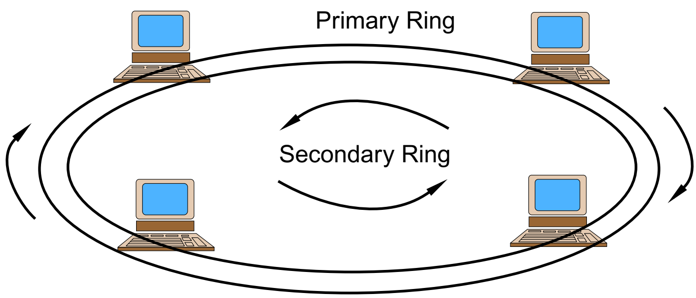


FIGURE 10.2 Example of FDDI's counterrotating ring architecture. The primary ring is active in normal operation; the secondary ring provides redundancy. All devices on the ring are dual-attachment stations (Class A nodes) or dual-attachment hubs.

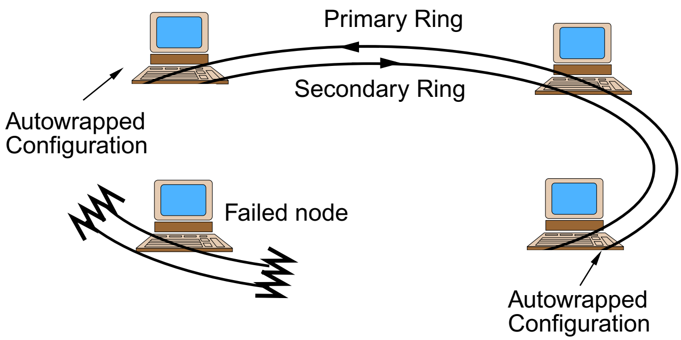


FIGURE 10.3 Example of FDDI's "self-healing" capability. In the event of a fiber cut or an inoperative node, an FDDI network automatically "heals" itself by wrapping the ring at point of failure. This is done by interconnecting the primary and secondary rings into a single functional ring. Source: Adapted from Daniels, 1990.

problem for a network manager. Usually, in networks less than 1 km in length, 850 nm is adequate; however, as the need for performance of the medium increases, the 850-nm light source becomes inadequate. Light transmission above 1550 nm usually requires a sophisticated and expensive light source, such as a laser system. Fiber runs cannot be longer than 2 km between connections for multimode fiber (up to 60 km with single-mode fiber) and there is a total allowable distance of 100 km per FDDI ring (two rings are allowed). Each ring consists of two fibers. Thus, two rings obviously use four fibers. (See Chapter 4 for additional information about fiber-optic cable.)

8. How are FDDI data encoded? I heard that FDDI uses a technique different from what is used for Ethernet/802.3 or IEEE 802.5 token ring networks.

FDDI's physical layer does not use Manchester encoding, which is used in Ethernet/802.3 and 802.5 LANs. (See Chapter 8 for additional information about Manchester coding.) In Manchester encoding, each bit requires at least two signal transitions or baud. This means that a 16-Mbps token ring network requires a signaling rate of 32 MHz. Ethernet/802.3 running at 10 Mbps requires 20 MHz. If we were to use Manchester encoding on an FDDI network, more than 200 MHz would be required to provide the FDDI rated speed of 100 Mbps. Instead, FDDI uses a "group" encoding scheme known as the 4B/5B method, which stands for four bits in five baud, or four-bit to five-bit.

9. Could you please expand on this?

Certainly, but we will have to get a little technical.

10. Go for it. If it's over my head I can always ignore it or seek additional information for further clarification.

OK. The 4B/5B encoding method takes data in four-bit codes and maps them to corresponding five-bit codes (Table 10.1). These five-bit codes are then transmitted using a technique called *NRZI*, which stands for *nonreturn to zero, invert on ones*. By transmitting five-bit codes using NRZI, a logic 1-bit is transmitted at least once every five sequential data bits resulting in a signal transition. The 4B/5B-NRZI scheme makes it possible for FDDI to operate at a rate of 125 MHz and provides a data rate of 100 Mbps. The use of one extra bit for every five bits translates to only 20 percent overhead for every clock encoding. In contrast, Manchester coding requires 50 percent bandwidth overhead for clock encoding because it guarantees at least one signal transition for every bit transmitted. The 4B/5B-NRZI scheme allows FDDI networks to provide high-speed capability over less optimal media and data symmetry that allows for simpler implementation of analog capture circuitry for receiving nodes.

TABLE 10.1 FDDI Symbols and Codes (4B/5B Encoding)

FDDI Code	Bit Encoding	FDDI Code	Bit Encoding
0	1 1 1 1 0	C	1 1 0 1 0
1	0 1 0 0 1	D	1 1 0 1 1
2	1 0 1 0 0	E	1 1 1 0 0
3	1 0 1 0 1	F	1 1 1 0 1
4	0 1 0 1 0	S (Set)	1 1 0 0 1
5	0 1 0 1 1	R (Reset)	0 0 1 1 1
6	0 1 1 1 0	Q (Quiet)	0 0 0 0 0
7	0 1 1 1 1	I (Idle)	1 1 1 1 1
8	1 0 0 1 0	H (Halt)	0 0 1 0 0
9	1 0 0 1 1	T (Terminate)	0 1 1 0 1
A	1 0 1 1 0	J (Start 1)	1 1 0 0 0
B	1 0 1 1 1	K (Start 2)	1 0 0 0 1

11. I think that's enough for the physical layer. What about FDDI's data link layer? Is it equivalent to the data link layer of a token ring network since FDDI is a ring topology?

Not quite. FDDI uses a token passing scheme as its MAC sublayer protocol similar to that of IEEE 802.5 token ring type networks. However, FDDI does not operate like a "classic" token ring network.

12. In which way or ways is FDDI different from token ring?

We described a few differences between token ring and FDDI earlier in. We'll summarize them here and include some additional ones. (You might want to review Chapter 9's discussion on token ring networks at this time.)

- FDDI networks can have two counter-rotating fiber-optic rings. This allows configurations of redundant topologies for highly reliable networks. IEEE 802.5 networks operate on a single ring topology (see Figure 10.2).
- On IEEE 802.5 token ring type networks, it is possible for stations to implement a priority scheme whereby token ring nodes can "reserve" a token for access to the medium. This scheme does not exist on FDDI networks because it would not work properly in the FDDI environment. FDDI nodes usually send a token at the end of a data transfer, which means that reservation techniques do not work. This is referred to as "new token after send" which is different than IEEE 802.5's "new token after receive."
- FDDI networks have an explicit maximum data size of 4500 octets per frame. There is no such explicit data frame size for IEEE 802.5 networks. Specification of an explicit frame size precludes a node from "hogging" the cable.
- FDDI networks have the capacity to support a distributed recovery capability in case of ring failure. This means that if the ring is cut, nodes on the FDDI network automatically isolate the fault and actively reconfigure the network to provide maximum availability.

Other differences between FDDI and IEEE 802.5 token ring networks include:

- FDDI does not use bit definitions for various fields. All FDDI fields are defined by at least four bits and may be defined by a byte of information so that the various fields can easily be modified or replaced by the nodes on the network as the frames and token travel through the ring(s).
- An optional technique in token ring networks, but implemented as a feature in FDDI, is the concept of *early token release* (ETR). ETR places a token on the network *before* the generated frame has had the opportunity to circulate throughout the entire network.
- FDDI tokens are not modified to a start of frame (SOF) as on other token ring networks. Tokens are absorbed and regenerated after a message has been sent. Tokens in an FDDI network also react differently from those of other token ring networks in that there are more accommodations for statistical network interconnections than on classic token ring architectures.

- On token ring networks, one clock on the network is responsible for providing clocking signals for all nodes on the cable. The main clock node also provides an "elastic" buffer capability that slides to compensate for speed differentials that appear on the network; this is referred to as *jitter*. In a 100-Mbps FDDI environment, this type of clocking mechanism is impractical and difficult to maintain. At 4 Mbps, the bit-time is 250 ns as compared to 10 ns per bit-time at 100 Mbps. Consequently, FDDI nodes provide their own clock (hence a "distributed" clocking scheme) and correct for timing jitter via each node's own internal elastic buffer.

13. What about the format of an FDDI frame? How different is it from the format of a token ring frame?

FDDI networks employ two types of "frames." The first is the token, which is a special frame that enables a node to access the ring; the second is the frame itself. FDDI token and frame formats are quite different from a token ring frame format (Chapter 9). Instead of bit definitions, FDDI networks use "symbols," which are defined by at least four bits (Table 10.1). Following is a summary of the token and frame formats:

Token An FDDI token consists of a preamble (PA) of 16 or more I symbols, a starting delimiter (SD) of a JK symbol pair, a frame control (FC) consisting of two symbols, and an ending delimiter (ED) field of two T symbols. The FDDI token format is shown in Figure 10.4. There also are two classes of tokens. A *restricted token* enables two specified nodes to use all of the unused or unreserved bandwidth of the network for the duration of their data transmission. An *nonrestricted token* is used for normal operation. Only one token is permitted on the ring. Thus, only one node is permitted to transmit data.

Frame An FDDI frame consists of a preamble (PA) of 16 or more I symbols, a starting delimiter (SD) field consisting of a JK symbol pair, a frame control (FC) field of two symbols, destination and source addresses (DA and SA) each consisting of four or 12 symbols, an info field of zero or more symbol pairs for user data, a frame check sequence (FCS) of eight symbols, an ending delimiter (ED) of one T symbol, and a frame status of three or more R or S symbols. The format of an FDDI frame is shown in Figure 10.5.

Note that the first two fields (PA and SD) are collectively called the *start of frame sequence* (SFS); the next five fields (FC, DA, SA, Info, and FCS) are collectively known as the *frame check sequence* (FCS) *coverage*; and the last two fields are collectively referred to as the *end of frame sequence* (EFS). Also note that addresses can be either 16 or 48 bits in length. Finally, the FCS uses a 32-bit CRC checksum for error control.

14. How does an FDDI network operate?

To access the ring, a node must first gain possession of the token. Once it has the token, the node is permitted to transmit multiple frames until a timer expires. When its time is up, the node retransmits the token on the ring. Only one token is permitted, thus only one node can access the ring at any time. As frames circulate around the ring, if the bits do not match a specific pattern then the receiving station is aware that the token is used and that this frame should be examined to see if it is to continue being passed along

FIGURE 10.4 FDDI token format. The preamble consists of 16 or more I symbols; the starting delimiter consists of one JK symbol pair; the frame control consists of two symbols; and the ending delimiter consists of two T symbols.

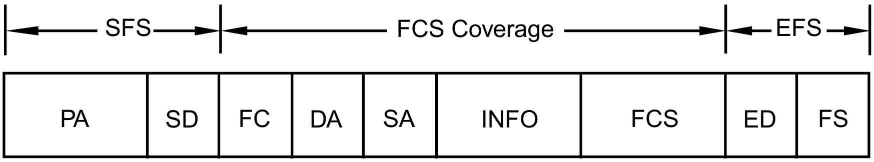


FIGURE 10.5 FDDI frame format. The preamble (PA), starting delimiter (SD), frame control (FC), and ending delimiter (ED) are the same as for a token (Figure 11.4). The destination address (DA) and source address (SA) are four or eight symbols. The info field is user data and consists of zero or more symbols. The frame check sequence (FCS) consists of eight symbols and uses a 32-bit CRC checksum. The frame status (FS) consists of three or more R or S symbols. The status can be E for error detected in frame, A for destination address recognized, or C for frame copied. The entire frame can be grouped into three primary fields: start of frame sequence (SFS), frame check sequence (FCS) coverage, and end of frame sequence (EFS).

the ring. If the node examining the frame is not the intended recipient, then the frame is regenerated to the next station in the path. If the current station is the intended recipient, then the station "snapshots" the data, passes it to the host system, and marks the bit pattern at the beginning of the frame to signify that the data were received and read into the receiving node. The frame is then passed along to the next node, continuing until it eventually reaches the sending node, which "strips" the frame by not retransmitting it. This scheme is really critical in view of the potential size of FDDI networks—1000 or more systems. It would take a very long time for frames in a classic token ring network to traverse such a large network (most token ring networks have less than 250 stations).

15. One *thousand* or more systems? That sounds like a large network!

It can be. It's not uncommon to find FDDI networks with over 400 connections in a single campus environment, although most FDDI networks have fewer than 100 connections to ensure good performance (Figure 10.6). Having such a large number of nodes on a single network implies that if it is not carefully designed to avert network outages, a very large number of people who rely on the network will find themselves unable to accomplish anything.

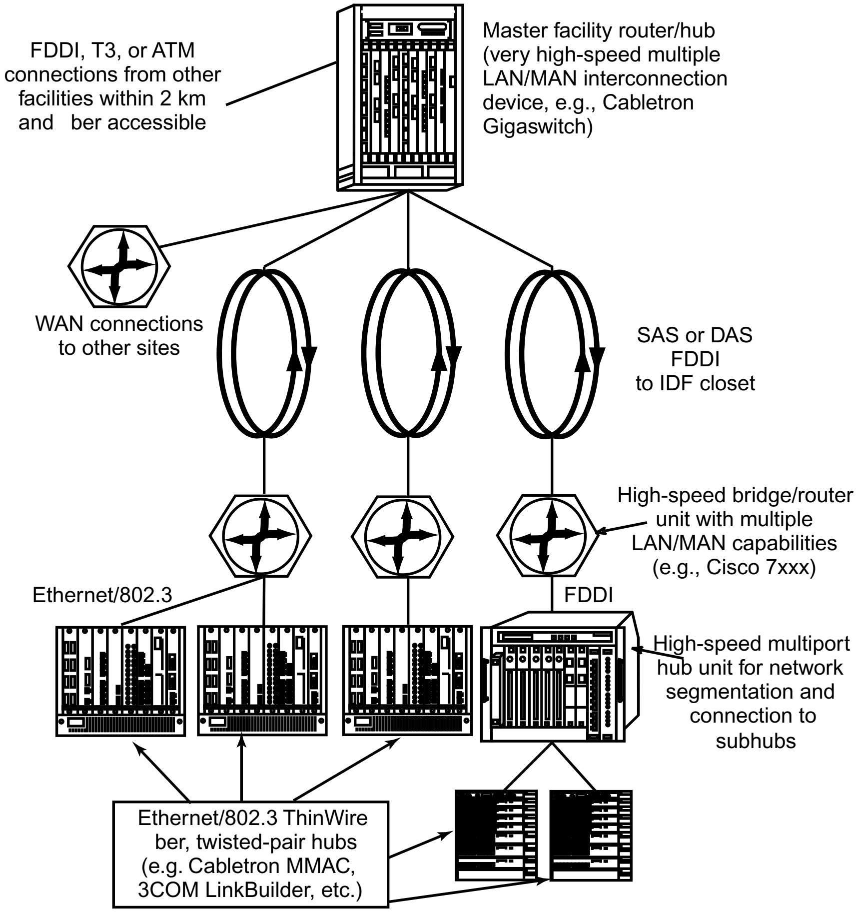


FIGURE 10.6 An example of a typical high-speed network backbone that uses FDDI to interconnect several different types of LANs and WANs.

16. What are the rules in configuring an FDDI network?

Pretty simple, really. A few specific fiber types are supported and specific lengths are required between optical interconnections, depending upon the vendor hardware selected. The two individual fiber rings' total length must not exceed 200 km when autowrapped.

17. How many systems are allowed on a single FDDI network?

The ANSI standard specifies that a maximum of 500 nodes is permitted on an FDDI network with a maximum ring circumference of 100 km and a maximum distance of 2 km between hardware devices. Some vendors allow a greater number of nodes depending upon their hardware and configuration rules. Some allow fewer. FDDI networks of over 1000 nodes are not uncommon.

18. Earlier you mentioned something called a Class A configuration. Could you please elaborate?

You betcha. Within FDDI, there are two types of network connections (nodes): Class A and Class B. Class A nodes are connected to two full, dual-fiber rings and have the ability to reconfigure the network to form a valid network from components of the two rings in case of a failure. Class B nodes only connect to the primary pair of fibers and can be isolated from the network in the case of some types of failure. Class A nodes are called *dual-attachment stations (DAS)*; Class B nodes are called *single-attachment stations (SAS)*. Some Class B nodes are equipped with bypass connections, which allow light source continuation even if the node connection fails. These bypass connections serve to maximize uptime of the network. An FDDI network designed with Class A nodes is called a Class A configuration.

19. How would an FDDI network be configured as the main or backbone network?

In most FDDI wiring configurations, the network designer configures a main backbone to be set up as a series of Class A nodes for reliability purposes. The length of the backbone is a function of how widely distributed a network is in a building, in a campus, or in a metropolitan area. In a building environment, the backbone may be located in a single room with all segments connecting to it. The backbone may be located vertically throughout telephone closets in the building. There may be other variations as well. In all these cases, the FDDI backbone network is contained within the building. In a campus environment, the backbone could also be in a single room in a single building, but most likely runs from building to building in the campus. In this configuration, each building has a main network control closet, so there is only one fiber coupler (tap) on the network backbone at each building location. In any case, the backbone network may vary in size and number of connection points, depending heavily upon where the network is located and how it is used. An FDDI network also may appear as a subsidiary backbone within a building that is connected to the main FDDI backbone. In this example, there are two FDDI network levels: the main backbone and a separate FDDI network per building. This may be expanded further to an FDDI network on a floor connected to the building hub, which is, in turn, connected to the main backbone in a campus building environment. In this configuration, we run the fiber to the actual machine(s) on the network; other networks may or may not be bridged onto the FDDI.

20. How is a system connected to the FDDI fiber?

For a node to be connected to a fiber-optic network, the network must have a tap attached to it. These couplers require that the cable be cut and terminated properly with approved connectors. These are typically the SC or media interface connector (MIC) variety of fiber connectors. (MICs are designed to be used by untrained personnel.) Splices in fiber may be done via a "score and break" technique where the fiber is scored, broken with special tools, and the ends of the two fibers connected with epoxy. A second method is to use a fusion splicer, which may employ an electrical arc to connect the ends of two fibers together. Splices are reasonably straightforward. Termination of cable, however, requires that the ends be cut and polished before they are inserted into terminating hoods. Polishing is typically done in a coarse way with a hand tool and completed with a polishing machine. This allows a smooth, flawless end that will properly pass the light from the source to the destination transceiver or repeater. This polishing effort may take up to 40 minutes per fiber strand (imagine this on a 48-fiber run!), which accounts for the high cost of fiber installation. Improper polishing and connection of the termination points on a fiber will at most render the fiber useless or will, at the very least, result in poor network performance. After it is terminated, the fiber must be tested to ensure proper installation. This normally involves using an optical time domain reflectometer (OTDR), an expensive testing device. In summary, properly installing fiber for FDDI is painful, time-consuming, and expensive.

21. So now I know how FDDI networks generally work. How would I use one in a "real" environment?

Glad you asked. Typically, FDDI networks implement backbone configurations for buildings and campus environments. In this design, a master FDDI "switch" connects separate FDDI networks on each floor in the building. This configuration is called a *collapsed backbone design*, where the master backbone is the switch and all major connections have been connected to the network switch (thus "collapsed" into a single box). This enables expansion and traffic isolation between the networks and reduces network failure potential. If the switch is "intelligent," it further isolates traffic and may include redundant facilities to keep the switch operational in adverse conditions such as motherboard failure or power supply failure.

The master "hub" switch of the building would be a very high speed, multinetwork concentration facility. Some hubs can support up to 36 FDDI networks (in the extended configuration) and can support T3 and asynchronous transfer mode (ATM) technologies in the same box. (See Chapter 7 for information about T3 circuits and Chapter 14 for information about ATM.) This tiered interconnection scheme using a very high-speed concentrating device is essential in developing high-speed, high-reliability networks.

22. What is an FDDI "switch?"

An FDDI "switch" interconnects many FDDIs through a common "box." This "box" or switch manages and sorts traffic between other switches so that it is sent only to where it needs to go. On an FDDI, all connected systems "see" all frames on the network. In a switched environment, the only traffic viewed by a specific FDDI is that which originates on it or is destined to it. Thus, all traffic on all other FDDI segments is isolated to each individual segment by the switch. This reduces overall traffic between the segments, reduces unnecessary traffic between networks, and improves performance overall network performance. (See Chapter 6 for additional information about switches.)

23. I have an Ethernet/802.3 LAN. How would I interconnect it to FDDI?

Interconnecting FDDI and Ethernet /802.3 is easier than ever today, especially given the communications hub technologies available from vendors. What is normally required is an Ethernet/802.3 to FDDI bridge unit. Such bridge units are available in many varieties, including a card in a hub or a dedicated box that does nothing but interconnect the two networks. Most manufacturers of hub units produce Ethernet/802.3 and FDDI cards that allow interconnecting the two types of networks.

One thing to be careful about is the issue of frame size conversion. Frames headed from Ethernet/802.3 networks to the FDDI network are not a problem because the frame formats have nearly identical addressing and data containers. However, machines on the FDDI network that use the maximum frame size of 4500 bytes will cause the Ethernet/802.3 network to be very unhappy. Therefore, the bridge must segment the network frame when moving it from the FDDI network to the Ethernet/802.3 network. This segmenting can cause all sorts of problems. If this situation exists, we recommend that you restrict the maximum frame size generated by a FDDI system to 1500 octets of data, which will reduce the magnitude of the problem. While this decreases the efficiency of the FDDI network, the hassle of large-frame-to-small-frame conversion would otherwise offset any potential performance benefits.

Another option is to use a router to interconnect the two network types. The benefit here is that the router can also handle frame conversions for specific protocols and automatically segment the traffic for routable protocols such as TCP/IP, AppleTalk, Novell's IPX, DECnet, SNA and others. The downside is that routers are usually substantially slower than switching bridges and can be pricey.

24. Can you provide a practical FDDI network application?

Sure. A really good example of an Ethernet/802.3-FDDI interconnected network is the Boeing 777 jetliner, which has an FDDI network that serves as its backbone with Ethernet/802.3 connections at the passenger seats. The Boeing 777 is one of the most technologically advanced commercial airplanes in the world, so it is fitting that it is delivered with advanced networks installed. Its designers implemented an MLAN (mobile local area network) consisting of two complete FDDI rings, one in the cabin and one in the cockpit. The FDDI network in the cabin is connected to Ethernet/802.3 LANs in the three compartments (economy, business class, and first class) by "brouters" (see Chapter 7). The purpose of these LANs is to transmit information to the multimedia units at each of the seats. The FDDI ring in the cockpit conveys data from two redundant servers to multimedia instrumentation units and navigation displays while it collects and stores maintenance information. The Litton FiberCom Avionics Bridge/Router (brouter) is the main internetworking device on the Boeing 777 aircraft. The brouter is a networking bridge which supports FDDI and Ethernet/802.3 network interfaces. The brouter has the built-in capacity to internetwork between three dual FDDI networks, a 10BASE2 Ethernet, and two 10BASE-T Ethernet hubs with eight ports each. The brouter is designed to be the boundary device for the two main aircraft FDDI networks on the 777 aircraft: the Optical LAN (OLAN, formerly called PlaneNet) and the Cabin LAN (CABLAN). Its design also provides a Gatelink connection via the third FDDI interface. The brouter allows the aircraft maintainer easy access to the aircraft's on-board maintenance system via multiple Ethernet ports. These Ethernet ports are tapped by portable maintenance access terminals via connectors located at various points around the aircraft (wheel well ports, tail port, electronic equipment bay port, etc.). The brouter can support both software and hardware upgrades for additional capabilities (e.g., OSI, TCP/IP, IPX protocols). It is data loadable onboard the aircraft via the FDDI interface to support future brouter software upgrades. If you would like to play with network configuration and deployment on a Boeing 777 jetliner, you can purchase one from Boeing for about $150 million. At that price, the networks are included.

25. Can FDDI interface with an ATM network?

Of course. In fact, high-speed technologies such as Cabletron's Gigaswitch and the Cisco Catalyst series of switches support not only FDDI networks but also ATM networks in the same box at the same time. The only "gotcha" with these technologies is that any time large data frames, such as those of FDDI, must traverse an ATM network, the frames must be pared to the payload size of ATM cells, which is 44 octets. (The overall cell is 53 octets, with the remainder keeping track of such things as where the cell is going.) Taking a full 4500-octet FDDI network connection frame and chopping it up into 53 octet cells is time-consuming and demanding of the networking hardware. Therefore, even if the ATM network is faster than the FDDI (such as an OC-12-622 Mbps ATM connection), the effective throughput may actually be lower due to the additional processing required to "cellify" the FDDI frames to ATM and then deliver the cells to their destination. If the connection is an ATM network interconnecting two or more FDDI networks, then the workload is even greater on the link path between the two locations.

26. You mentioned CDDI in the introduction to this chapter. I'm guessing it's the same as FDDI with copper instead of fiber at the physical layer. Am I right?

Right you are. However, the use of copper wire in CDDI is restricted to connections between concentrators on the ring and single-attachment devices, not for the ring itself. (*Note:* FDDI concentrators are DAS devices that can be used to connect multiple SAS devices.) Copper is, at present, somewhat cheaper for use in short runs (up to 100 m).

27. What type of cabling does CDDI use?

CDDI supports both unshielded twisted-pair (UTP) and shielded twisted-pair (STP). As for which type of UTP, the familiar and popular EIA/TIA-568 Category 5 cabling, already installed in many buildings' wiring closets, may be used for CDDI. Similarly, IBM Type 1 STP is acceptable for CDDI use; occasionally you may encounter the abbreviation SDDI, which is sometimes used in reference to CDDI networks employing shielded cabling. At one time there was also a proprietary CDDI product that supported ThinWire coaxial.

28. Are CDDI and SDDI real standards, like FDDI?

They started out as de facto standards created by a group of five vendors. In 1995, ANSI issued a standard for copper-based FDDI, X3.263-1995, entitled *Fibre Distributed Data Interface (FDDI)—Token Ring Twisted Pair Physical Layer Medium Dependent (TP-PMD)*.

29. What about the future of FDDI? Is there a second-generation FDDI?

There was at one time, but the introduction of other, faster network technologies such as Gigabit Ethernet and ATM have caused a halt to any development efforts on FDDI follow-ons. For example, a technology called FDDI-II was intended to handle not only traditional FDDI network traffic, but also synchronous, circuit-switched PCM data for voice or ISDN systems. Development has all but ceased and no commercial systems have been deployed in the FDDI-II environment, especially given the growth of other, more popular technologies such as ATM and SONET. As we mentioned earlier in this chapter, FDDI is now rarely used as a direct-system connection network platform, being primarily relegated to backbone deployments. Other network types, such as Ethernet/802.3, have impinged upon the applicability of FDDI, displacing it because of less expensive implementation, interconnection, and upgrade paths. Therefore, any implementation of an FDDI network needs an upgrade strategy in order to provide for the day when it will be too slow for its applications. Current network designers looking toward FDDI for expansion may find cheaper methods to provide 100-Mbps connectivity with greater longevity and expansion capabilities than FDDI.

Summarizing, although FDDI is still a viable and ubiquitous network technology, its days are numbered as less expensive Ethernet/802.3 100-Mbps, 1000-Mbps, and 10,000-Mbps systems supplant what were traditionally FDDI network environments. ATM also has encroached on FDDI deployments with its scalability and support of isochronous transmission (used for video). FDDI still has its fans, however, and will not be leaving any time soon—especially given its ability to provide very high bandwidth and redundancy failover capabilities, while remaining stable for all types of networking applications.

END-OF-CHAPTER COMMENTARY

FDDI was one of the first “second-generation” LAN technologies designed specifically to meet the needs of users with high bandwidth requirements. Other second-generation LAN technologies discussed in this book are the various evolutions of Ethernet, including full-duplex Ethernet, switched Ethernet, Fast Ethernet, and Gigabit Ethernet (see Chapter 8). We also look at a revolutionary “Ethernet” technology called 100VG-AnyLAN. In Chapter 9, we examine some evolutionary changes to traditional 4/16-Mbps token ring networks, including dedicated token ring, switched token ring, full-duplex token ring, and 100-Mbps token ring. Finally, in Chapter 14, asynchronous transfer mode (ATM) networks are discussed from both LAN and WAN perspectives.

Chapter 11

Integrated Services Digital Network (ISDN)

The integrated services digital network (ISDN) represents the overhaul and redesign of our conventional telephone network from an analog system to an end-to-end digital network. This new, completely digital-based network is capable of transmitting voice and data communications over a single telephone line using inexpensive and conventional twisted-pair cable (i.e., standard copper telephone wire). A brief overview of various fundamental ISDN concepts was provided in Chapter 7 as part of our discussion of WAN technologies and services. In this chapter, we extend this discussion and provide more detailed information about ISDN. An outline of the major topics we present follows:

- History of ISDN (Questions 1–3)
- General Overview and Components (Questions 4–6)
- Channel Types (Question 7)
- BRIs, PRIs, and SPIDs (Questions 8–9)
- Line Sets and Feature Sets (Questions 10–12)
- ISDN Protocols (Questions 16–18)
- AO/DI and B-ISDN (Questions 19–20)
- Alternative Implementation Strategies (Question 21)

1. What is ISDN?

The integrated services digital network is a carrier service that is offered by telephone companies (telcos) and designed to transmit voice and nonvoice (e.g., computer data, fax, video) communications on the same network. The advantage ISDN offers over other services is that separate connections are not needed for these different transmissions. Thus, instead of having a telephone line for voice communications, a second telephone line for fax or computer dialup connections, and a coaxial cable link for video communications, a single ISDN connection will support all of these transmissions. That is, ISDN *integrates* all of these services into a single system. ISDN service (and hence, an ISDN connection) is completely digital from end-to-end. This represents both a departure from and an improvement in today's conventional telecommunication services, which use a hybrid of

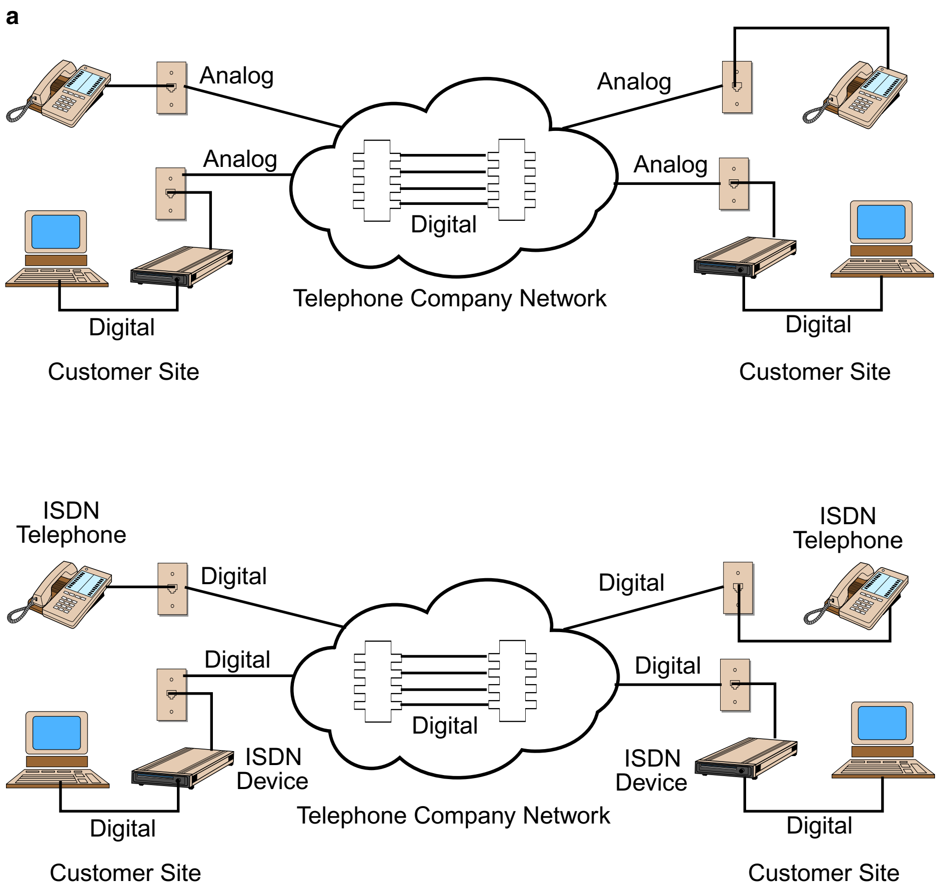


FIGURE 11.1 Today's conventional telephone network (a) is a hybrid of analog and digital technologies. Note that a computer connection requires four conversions: digital to analog from the PC, which is digital, to the customer's phone system, which is analog; analog to digital from the customer's site to the telephone company's site, which is all digital; digital to analog from the telephone company's site to the customer's site; and analog to digital from the customer's phone system to the PC. An ISDN connection (b), however, which uses special ISDN-compatible devices, is digital from end to end. In the case of the computer connections, no digital-to-analog or analog-to-digital conversions are necessary, which result in faster connections.

analog and digital technologies (Figure 11.1). Furthermore, ISDN's technology permits standard twisted-pair wiring to carry circuit- or packet-switched digital data. ISDN also provides a cost-effective strategy for internetworking. Instead of paying for dedicated leased lines, remote sites (i.e., user workstations or LANs) can interconnect with other sites via dialup links.

2. How did ISDN get started?

The advent of digital technology in the telephone industry in the 1970s marks the roots of ISDN. During this time, telcos modernized their central offices (COs) with digital switching equipment, replacing the analog-based public switched telephone network (PSTN) that was originally designed for voice transmission with one predicated on digital signaling and circuitry. What emerged was an *integrated digital network* (IDN), which engendered a vision of a network capable of transmitting any information source—voice, video, graphics, images, and text—directly to customers regardless of their location. In today's jargon, this vision was to treat all of these information sources as data, market them as services needed by businesses and consumers (e.g., videoconferencing), and then deliver these services directly to the desktop or home. There was one problem, though: Delivery of these services required end-to-end digital connectivity. Although IDN was digitally-based and capable of voice, data, and video transmissions, it was not end-to-end. Delivery of these services was restricted because the *local loop* (the circuit between a user and the telco's nearest point of presence) was still analog. In the 1980s, standards for digital end-to-end connectivity and transmission of both voice and nonvoice services to the user were approved. This new network was named the *integrated services digital network*, or ISDN.

The development, approval, and acceptance of ISDN standards spurred widespread deployment of ISDN services throughout Europe where the telcos were, at the time, mostly government-owned. (More on this later.) In the United States, though, ISDN essentially became a product looking for a market and laid dormant for nearly 10 years. It wasn't until the Internet achieved critical mass in the mid-1990s before ISDN came to life in the United States. Given this 10-year period, from the approval of ISDN standards to its widespread availability and ultimate implementation in the United States, alternative interpretations of the ISDN acronym emerged. These included, among others, I Still Don't Need it, Innovative Services users Don't Need, I Still Don't kNow, and It's Still Doing Nothing. Today, ISDN has emerged as a viable, cost-effective solution for remote and WAN applications, and its acronym for some now represents Innovations Subscribers Do Need.

3. Who is responsible for ISDN standards?

ISDN standards development is conducted under the auspices of the International Telecommunications Union (ITU), which is the former Consultative Committee for International Telephony and Telegraphy (CCITT). A subgroup of ITU, Telecommunications Standardization Section (ITU-TSS), is responsible for communications, interfaces, and other standards related to telecommunications. ITU also works in cooperation with other accredited standards committees such as the American National Standards Institute (ANSI).

The initial set of ISDN recommendations, formally called the I-series Recommendations, was published by CCITT in 1984. These recommendations were published in CCITT's *Red Books*, which comprises all CCITT standards. Additional work on the I-series Recommendations continued after 1984, and in 1988 an updated and more complete set of ISDN standards were incorporated into CCITT's *Blue Books*. Although subsequent updates have been made, the 1988 publication still adequately describes the basic principals and reference model of ISDN. A summary of the I-series Recommendations is given in Table 11.1, and a copy of the ISDN reference model is shown in Figure 11.2.

TABLE 11.1 Summary of ISDN I-Series Recommendations

Series Number	Description
I.100	Describes the general concepts of ISDN. It addresses fundamental ISDN principles, objectives, and vocabulary.
I.200	Specifies the various services ISDN provides.
I.300	Discusses ISDN network requirements—including its architecture and addressing scheme—and specifies the manner in which an ISDN network is to provide the services described in I.200.
I.400	Addresses issues related to the user interface from the perspective of the first three layers of the OSI model. Examples include data transmission rates, hardware configurations, and data link and network layer protocols.
I.500	Discusses interconnectivity issues (e.g., interconnecting ISDN networks with non-ISDN networks).
I.600	Devoted to ISDN maintenance issues.

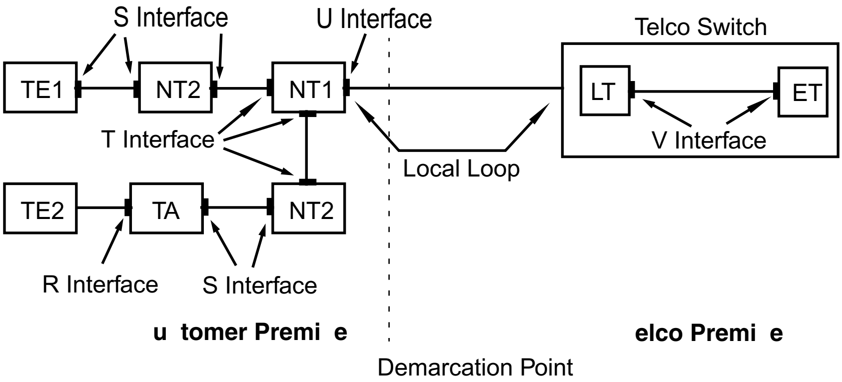
Speaking of standards, in the United States the *regional Bell operating companies* (RBOCs)—which are now called *incumbent local exchange carriers* (ILECs) or "baby Bells"—established a national ISDN initiative to foster end user equipment interoperability. A presumption of this initiative is that any ISDN end equipment that is National ISDN-compliant should be able to connect to any telco switch. The initiative's first protocol, National ISDN-1 (NI-1), was ratified in November, 1992 with a better than 80 percent compatibility rate. (Some RBOCs opted not to upgrade to NI-1.) The goal of subsequent protocols (e.g., NI-2) is 100 percent compatibility.

4. Figure 11.2 is a bit confusing. Can you please explain it to me?

We can understand your confusion. Figure 11.2 shows the basic components of an ISDN connection from the perspective of a reference or block diagram. These components, which do not necessarily correspond to actual pieces of equipment, include two types of network termination modules (NT1 and NT2), two types of terminal equipment (TE1 and TE2), and terminal adapters (TA).

The NT1 module provides the interface between the customer's premises equipment and the telco's equipment. It consists of a two-wire U interface at one end and up to seven T interfaces at the other end. Its purpose is to convert the two-wire ISDN link provided by the telco into four-wire ports that connect ISDN devices. From an OSI model perspective, NT1 is a layer 1 device since it deals with the physical and electrical termination of the circuit at the customer's site. An NT1 also performs synchronous time division multiplexing (TDM—see Chapter 4) to combine more than one channel. (More on this later.)

The second network termination unit, NT2, is a secondary termination module that converts the T interface from the NT1 module into S interfaces that connect terminal equipment or adapters (TE or TA). An example of an NT2 unit is a *private branch exchange (*PBX), which is a telephone switching system that provides telecommunication services throughout an organization's private network. If an organization's PBX is a digital switch,



Network Termination 1 provides connectivity between a customer's site and the telco's site. It converts the two-wire U Interface into a four-wire S/T Interface. NT1 modules can support up to eight connections (TEs, TAs, or NT2s).

Network Termination 2 converts the T interface into the S interface. NT2 provides data link and network layer functions. Connectivity to ISDN devices (TE1 or TE2) is via an S interface, and connectivity to an NT1 unit is via the T interface. An example of an NT2 device is a PBX system.

Terminal Equipment 1 an ISDN-compatible device. TE1s have built-in ISDN network interfaces and can connect directly to NT1 units via an S/T interface. Examples of TE1 devices include ISDN telephones and ISDN fax machines.

Terminal Equipment 2 a non-ISDN compatible device. TE2s can be connected to an ISDN network via a terminal adapter (TA) through an R interface. TAs are then connected to NT2 units via an S/T interface. Examples of TE2 devices include analog telephone or fax machines and computers without an ISDN connection.

A Terminal Adapter provides ISDN connectivity to non-ISDN devices (TE2s).

Line Termination represents the local loop connection; that is, it is where a circuit from a customer's NT1 module terminates at the telco's switch.

Exchange Termination connects a telco's ISDN switch to other ISDN switches within the telco's network.

FIGURE 11.2 An ISDN reference diagram showing the relationship between ISDN equipment and interfaces. The local loop (also called the subscriber loop) is the access line between the telco's central office (CO) and the customer's site. This link is terminated at the customer's site via an NT1 device. In the United States, NT1 devices are purchased by the customer. This makes the demarcation point (the point that separates customer premises equipment from the telco equipment) the U interface. In Europe, the telcos own NT1 devices and install them on the customers' premises. In this setting, NT1 devices are considered telco equipment, and the demarcation point is now the T interface. Source: Adapted from Frank, 1995, and Leeds, 1996.

which implies that the telephones connected to it are digital, then ISDN connectivity can be provided throughout the enterprise by connecting the PBX to an NT1 module. Another example of an NT2 unit is a LAN device such as a router. An NT2 performs functions that operate up to the third layer of the OSI model.

The terminal equipment (TE) represent specific communication devices that connect to the network. Two TEs are referenced, TE1 and TE2, to distinguish between compatible and incompatible ISDN equipment. TE1 examples include digital telephones and computers with built-in ISDN ports (e.g., a Sun SparcStation 10 was one of the first computers to have built-in ISDN capability). TE2 examples include analog telephones (i.e., those with RJ-11 jacks) and computers without built-in ISDN ports (e.g., RS-232C or equivalent serial ports). Most ISDN compatible devices have built-in NT2 modules and connect directly to an NT1. (*Note:* From a technical perspective, all ISDN devices must go through an NT2 unit.) Some ISDN TE1 devices also have built-in NT1 and NT2 modules (sometimes referred to as NT12), with U-interfaces that enable them to connect directly to the local loop. Such devices eliminate the need to purchase a separate NT1 unit. They also eliminate the capability of connecting more than one device to the network.

Finally, a terminal adapter, (TA), is a device that connects incompatible ISDN devices to an ISDN network. If a TA is used for an ISDN dialup connection, then it can be thought of as a modem (see Chapter 15); if a TA is used to connect a device to a LAN, then it can be thought of as a network interface card (NIC) (see Chapter 6). It should be noted that although a TA is frequently referred to as an ISDN modem or digital modem in the context of an ISDN dialup connection, this reference is incorrect. By definition, a modem performs analog-to-digital and digital-to-analog conversions. Since ISDN is completely digital, no such conversions are necessary, which makes the expressions *ISDN modem* or *digital modem* incongruous. Nevertheless, both expressions are frequently used because the general public can better relate to them than the term *terminal adapter*.

To help clear up any residual confusion, perhaps a more conceptual illustration of these units will help. Figure 11.3 provides examples of two typical home-based ISDN connections. In this figure, various home-based telecommunication devices are connected to an ISDN network. The devices at home 1 are ISDN terminal equipment (TE1). They have built-in NT2 modules and ISDN ports with the proper S/T-interfaces that enable them to connect directly into an NT1 module. The devices at home 2 in Figure 11.3 are not ISDN-capable. Thus, they require terminal adapters in order to connect to an ISDN network. These devices connect to their respective TAs via an R interface. TAs can be external units (as shown in Figure 11.3), or internal adapter cards that plug into a device's motherboard.

5. Thanks. That helped. I still have one more question about this, though. What's with all of those interfaces?

ISDN defines three primary interfaces to foster global interoperability of ISDN equipment—R, S, and T (see Figure 11.2). The U and V interfaces are U.S.-specific. The R interface provides a mechanism for non-ISDN devices to connect to an ISDN network. The ISDN device that supports the R interface is the ISDN terminal adapter, which is similar to and functions like a network interface card (see Chapter 6) or an analog modem (see Chapter 15). For example, to connect a computer to the Internet via a conventional (i.e., analog) dialup line requires either an internal or external modem. An internal modem is connected directly to the computer's logic board and provides an RJ-11 port for the telephone line, and an external modem provides two interfaces—a serial port (e.g., RS232-C or equivalent) and an RJ-11 port. To connect this same computer to an ISDN network via a

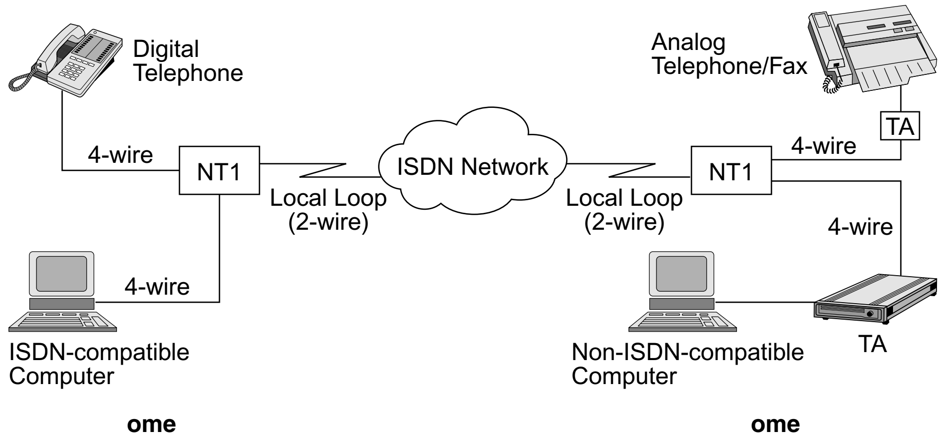


FIGURE 11.3 A home-based ISDN connection consists of a four-wire circuit that connects the home to the telco's ISDN network. This circuit physically and electrically terminates at each house's demarcation point, which is an NT1 module. All of the communication devices at home 1 have built-in NT2 modules with ISDN connector ports that provide an S/T interface to connect directly to the NT1 module. The devices at home 2, however, are not ISDN compatible. They do not comply with ISDN interface requirements and hence require terminal adapters (TA) to connect to the NT1 module. The devices at home 1 are TE1 devices; those at home 2 are TE2 devices. (The TA connected to the computer at home 2 is commonly, although incorrectly, called an ISDN modem.)

dialup line, a TA is used to provide the necessary interfaces (e.g., serial and R) to connect the non-ISDN compatible computer (or other device) to an ISDN network. In this capacity, a TA is often referred to as an ISDN modem or digital modem.

The S and T interfaces are standard ISDN digital interfaces that are electronically equivalent. They are four-wire connections that partition the two-wire access line provided by the telco into separate transmit and receive lines. All ISDN-compatible equipment have S interfaces and plug directly into an NT1 module via a T interface. A distinction between these two interfaces is made because in Europe, the T interface serves as the point of demarcation that separates the customer premises equipment (CPE) and the telco provider's equipment. In the United States, the demarcation point is the U interface. The reason for these two points of demarcation has to do with the way providers in the United States and Europe deliver ISDN service. U.S. customers are expected to purchase their own NT1 modules, which makes them CPE, while in Europe, NT1 modules reside at the customer's site, but are owned by the telcos making them part of the telco's premises equipment. NT units usually label their interfaces as S/T.

As indicated earlier, the U and V interfaces are U.S.-specific. The U interface, in addition to representing the point of demarcation, is where the telco's access line from its switch is terminated at the customer's site. This access line is commonly referred to as the *local loop* or *subscriber loop*. The V interface is used to connect the exchange and line terminations (ET and LT) within an ISDN switch.

6. OK. What else can you tell me about ISDN?

Well, for starters, as a completely digital service, ISDN eliminates the need for traditional modems, which perform analog-to-digital and digital-to-analog conversions. This provides extremely fast connections. For example, a traditional dialup modem connection over analog telephone lines requires anywhere from 30 to 60 seconds to establish a connection; an ISDN connection takes approximately 2 seconds.

ISDN also maintains a logical separation of user data (voice and non-voice) from signaling and control information. ISDN uses *bearer* or *B channels* for transmitting data, and a *signaling* or *D* (delta) *channel* for transmitting signaling and control information. Since ISDN uses a separate channel for signaling information, 100 percent of the bandwidth allocated for an ISDN *B* channel is used for data transmission. This offers an advantage over traditional T1 service where control information is in-band. For example, a 128-kbps fractional T1 line (two DS-0 circuits) provides only 112 kbps for data because 8 kbps per DS-0 channel is used for control. A 128-kbps ISDN circuit (two *B* channels), however, provides a clear 128 kbps because both *B* channels are free of any signaling overhead. (See Chapter 7 for additional information about T1 circuits.) There is also an *H* channel that is used for transmitting user data at higher transmission rates than the B channel.

ISDN is also a connection-oriented service, which implies that fixed virtual circuits are established between source and destination nodes. A virtual connection is first established between sender and receiver prior to transmission, the circuit remains in effect and dedicated exclusively to this session for the duration of the transmission, and the circuit is then disconnected after the transmission ends (see Chapter 2).

Although call setups between source and destination nodes are nailed up and torn down by the ISDN provider, the customer's access line (the connection between a customer's site and provider) is fixed and physical. To reduce call charges between a customer site and provider, most ISDN hardware devices support automatic dial-on-demand connections. Instead of continuously keeping the customer-provider circuit up, dial-on-demand establishes a connection only when data frames have to be transmitted. After a certain period of time (which is user-configurable) in which no traffic is being transmitted from the customer's site to the provider's network, the device automatically terminates the call (hangs up). This is analogous to the way we use our telephone—whenever we need to talk to someone we place the call; in all other situations the phone is kept on-hook. ISDN also supports both circuit- and packet-switched connections and is an international standard based on the concepts and principles of the OSI model (see Chapter 2).

7. Tell me more about these separate channels.

ISDN defines several different channel types. These include *B, D,* and *H* channels. The *B* channel is a 64 kbps clear channel used to transmit computer data (text and graphics), digitized voice, and digitized video. (Recall that a clear channel means that no signaling information is sent on the channel.) *B* channel transmissions are either circuit- or packet-switched. Data also can be exchanged via frame relay (Chapter 12) or through a dedicated leased line arrangement. In a leased line configuration, no call-control information needs to be transmitted on the *D* channel. Most basic ISDN services are based on multiple *B* channels.

The *D* channel is either a 16-kbps or 64-kbps channel, depending on the specific service level provided. (More on this later.) The *D* channel is used to carry signal and control information for circuit-switched user data. The *D* channel transmits information related to call initiation (call-setup) and termination (call tear-down) between an ISDN device and the telco's central office for each B channel. Thus, when a telephone call is made between two sites, the *D* channel handles all of the call-related information for the *B* channels. This is why the *B* channels are clear 64 kbps channels. The *D* channel also can be used to transmit packet-switched user data (provided that no signal or control information is needed), data from security alarm signals of remote sensing devices that detect fire or intruders, and low-speed information acquired from telemetry services such as meter reading. (*Note:* Telemetry applications involve obtaining measurements remotely and relaying them to another site for recording or display purposes.)

The *H* channel is used for transmitting user data (not signal or control information) at higher transmission rates than the *B* channel provides. Four *H* channels are defined: *H0*, *H10*, *H11*, and *H12*. *H0* comprises six *B* channels for a total capacity of 384 kbps. The *H10* channel is U.S.-specific and aggregates 23 *B* channels for a total capacity of 1.472 Mbps. The *H11* channel is the equivalent of the North American DS-1 (see Chapter 7) and consists of 24 *B* channels for an aggregate bandwidth of 1.536 Mbps. The *H12* channel, which is European-specific, comprises 30 *B* channels and has an aggregate bandwidth of 1.920 Mbps. Examples of applications that might use an *H* channel include videoconferencing high-speed fax, or high-speed packet switched data, and high-quality audio.

B and *D* channels are generally combined (i.e., multiplexed using TDM) by ISDN service providers and offered to customers in different bundled configurations. The most common package is the $2B + D$ arrangement, which consists of two *B* channels and one *D* channel. This channel structure is known as *basic rate interface* (BRI). Two other common basic interface structures are $B + D$, and *D* only. In the BRI structure, the *D* channel is 16 kbps. A second type of channel structure is called *primary rate interface* (PRI), which has a general configuration of $nB + D$. The two most common PRIs are $23B + D$, which is equivalent to the North American DS-1 rate of 1.544 Mbps, and $30B + 2D$, which is equivalent to the European E-1 rate of 2.048 Mbps. In the PRI structure, the *D* channel is 64 kbps.

8. Please elaborate on BRI and PRI.

OK. We will discuss these separately.

BRI The ISDN Basic Rate Interface (also known as *basic access*) is a 192-bit channel that consists of two 64-kbps *B* channels, one 16-kbps *D* channel, and 48 bits of overhead used for framing and other functions. (See Figure 11.4. See also Chapter 5 for more information about framing.) The two *B* channels and the *D* channel are combined into a single pair of standard copper telephone wires. Both *B* channels can support any combination of voice or data transmissions (e.g., both voice, both data, one voice and the other data). BRI provides a full-duplex data rate of 128 kbps. If call or signal information is not being carried by the *D* channel (e.g., transmitting data via packet-switching), then the rate increases to 144 kbps if the *D* channel is carrying data. Data rates also can be increased anywhere from four to eight times more through data compression (Chapter 15). (*Note:* Some telcos might still use older signaling software that requires ISDN signaling to be done in-band. In such instances, 8 kbps per *B* channel must be reserved for signaling resulting in a BRI service that provides 112 kbps instead of 128 kbps.)

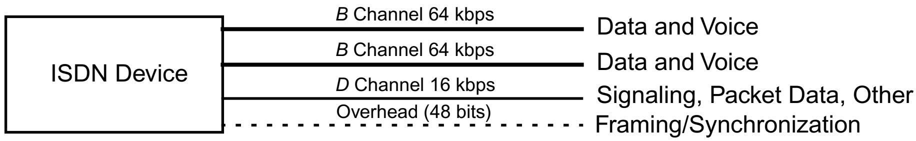


FIGURE 11.4 ISDN's basic rate interface (BRI) is a 2*B* + *D* package provided by the telcos. BRI consists of two 64-kbps *B* channels used for transmitting data or voice (or both) and one 16-kbps *D* channel used for transmitting signaling and control information. If no signal or control information is present, then the *D* channel can be used for transmitting data as well. BRI also consists of an additional 48 bits used for framing and synchronization.

ISDN BRI has two telephone numbers assigned to it—one for each *B* channel—and hence effectively provides consumers with two telephone lines via a single connection. Thus, at home, one *B* channel can be used for standard voice service (except here it is digital and not analog), and the other *B* channel for a fax machine. Alternatively, one *B* channel can be for standard voice and the other *B* channel for an Internet connection. Also at home, the *D* channel can be used for telemetry services. For example, utility companies could use this channel to obtain readings from the electric, gas, or water meter.

ISDN BRI *B* channels, through inverse multiplexing (Chapter 4), also can be combined to form a single channel with an effective bandwidth of 128 kbps. This process is called *BONDING*, a protocol named *Bandwidth ON Demand Interoperability Network Group*. An inverse multiplexer (imux), enables the two *B* channels to be used simultaneously. When a standard BRI connection is first established, the receiving device negotiates various communications parameters with the sending device for the current session. In a BRI BONDING scenario, these parameters include two connections and a 128-kbps transmission rate. Once the receiving device receives and accepts these parameters, it instructs the sending device to establish a second connection by calling its second telephone number. An alignment process then occurs that effectively combines the two lines into a single channel. An alternative to BONDING is the *multilink point-to-point protocol* (MPPP), which aggregates two or more *B* channels and runs PPP across these multiple *B* channels.

PRI As noted earlier, the primary rate interface (also know as *primary access*) has two standard configurations. The first is based on the North American DS-1 (1.544-Mbps) format and the second is based on the European E-1 (2.048-Mbps) format (see Chapter 7). PRI service is essentially the same as BRI except PRI has 23 (or 30) *B* channels instead of 2, and PRI's *D* channel operates at 64 kbps instead of 16 kbps (Figure 11.5). Unlike BRI, which is appropriate for the home or small office, PRI is more appropriate for organizations that have to provide telecommunication services to a large number of sites. For example, large corporations with various satellite offices or remote sites, corporations with PBXs, and Internet service providers are more likely to subscribe to PRI service than BRI. With PRI, it is possible to have up to 23 (or 30) separate, independent, simultaneous ISDN connections.

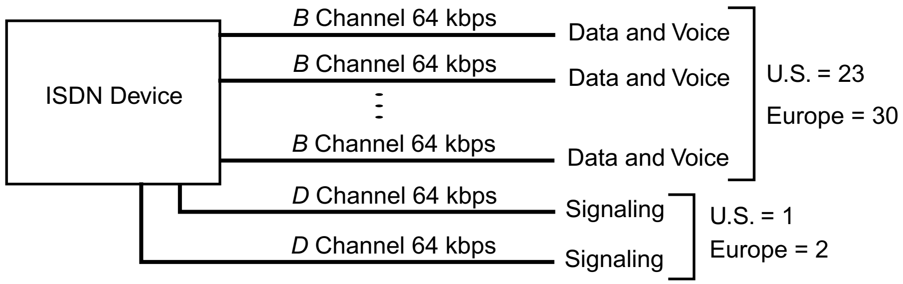


FIGURE 11.5 ISDN's primary rate interface (PRI) is packaged in two forms. A 23*B* + *D* configuration, which is based on the North American DS-1 format and rated at 1.544 Mbps, and a 30*B* + 2*D* configuration, which is based on the European E-1 format and rated at 2.048 Mbps.

9. A friend of mine has an ISDN connection and said that his phone numbers are called SPIDs. Is this true?

Not really. The telephone numbers assigned to ISDN *B* channels have *service profile identification* (SPID) numbers associated with them. SPIDs are provided by the telcos and are usually defined by adding a prefix, a suffix, or both (it depends on the telco's switch) to an assigned telephone number. For example, assume the number assigned to a *B* channel is 4075551469. If the telco uses a prefix of 05 then the SPID is 054075551469. SPIDs are used to identify the various processes of an ISDN device. By assigning a SPID to a device, the telco's ISDN switch can identify the processes associated with each device. This prevents contention among the processes. Thus, each device connected to an ISDN line must be assigned a unique SPID. Normally, if only one ISDN device is connected, a SPID is not needed. However, if more than one ISDN device is connected, then SPIDs are required. Some ISDN equipment manufacturers support an auto-SPID function that acquires the telco's assigned SPID to a particular connection and configure the end equipment automatically. SPIDs are used only in North America.

10. I have also seen the term "line set" used with ISDN. What is this?

The term *line set* is used by the North American ISDN Users' Forum (NIUF), which is an organization that provides users with a voice in the implementation of ISDN applications. Line set describes two specific characteristics of ISDN service: the number of multiplexed *B* and *D* channels, and the type of service supported. For example, line set 1 specifies a *D* configuration that supports packet switched data transmissions on a single *D* channel. Line set 4 specifies a 1*B* configuration that supports alternate voice and data transmissions on a single *B* channel. Line set 27 specifies a 2*B* + *D* configuration that supports alternate voice and data on two *B* channels and packet switched data transmissions on the *D* channel.

Associated with each line set is a *feature set*, which identifies specific ISDN features that can be ordered as part of the service. Feature set examples include call forwarding and calling number identification (i.e., caller ID).

11. Wait a minute here. I didn't know that ISDN had features like Call Forwarding and Caller ID. I thought people got an ISDN connection for the Internet. What other features does ISDN offer?

This is a common misconception. Remember, ISDN is an integrated services network. In addition to fast Internet connections, ISDN supports various telephone-related services including call forwarding, calling number identification (i.e., caller ID), call transfer, call waiting, and call hold. These services are similar to their non-ISDN counterparts that are offered through the telcos. There is one exception, though. With call waiting, when a second call comes in, you do not get interrupted, as is the case with non-ISDN call waiting. This is because the *D* channel handles all the call control information, keeping it separate from the *B* channels. Additional information about line and feature sets can be acquired via http://www.niuf.nist.gov/docs/428-94.pdf

ISDN also supports several teleservices including capability for 64-kbps Group IV fax, teletext, and videotext. *Teletext* and *videotext* are electronic information utilities that use computers or standard television sets equipped with adapters to display information. Teletext is a one-way communication system; videotext is a two-way (i.e., interactive) system. Teletext broadcasts data as part of a television signal; videotext uses cable television or telephone lines to transmit data. Typical applications available from these services include information retrieval (news weather, sports access to medical databases), electronic transactions (airline reservations, electronic funds transfer, shop-at-home services), interpersonal messaging (e-mail), computing, and telemonitoring (remote sensing, telemetry services). Clearly, many of these applications are now available via the Internet's World Wide Web. However, teletext and videotext services have been available in countries such as France, Canada, and Great Britain since the early 1980s. Furthermore, trials of these two services were conducted in the United States during the 1983–84 time period. Examples include Videotron by Knight-Ridder Newspapers in southeastern Florida, Keyfax by Keycom Electronic Publishing in Chicago, and Gateway, by Times Mirror Videotex Services in southern California in 1984.

12. Is an ISDN phone similar to a regular phone? For example, can I use one to call someone who doesn't have an ISDN phone or ISDN connection?

Yes. You can use your ISDN phone to call someone whose phone service is provided by the analog-based plain old telephone system (POTS). The reverse is also true. However, you will not be able to achieve high-quality connect sessions because only one part of the connection is digital; the other is analog. Thus, there will not be any improvement in line performance.

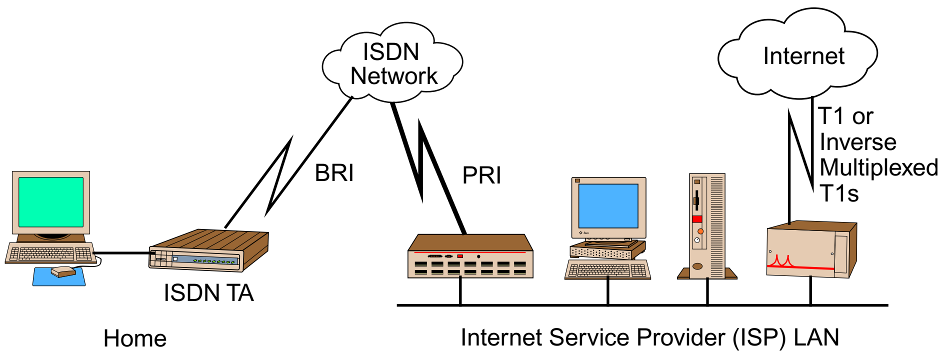
As for any differences in telephones, ISDN telephone sets have many more built-in functions and capabilities than conventional analog phones. Aside from that, both types of phones function similarly, with one exception. The analog-based POTS provides power to its telephones, which is why the telephone system is still able to operate during power outages. This is not the case with ISDN, which does not provide power to its telephones (or to any ISDN terminal equipment). Thus, unless the ISDN TE is protected by an uninterruptable power supply (UPS) or its equivalent, a power outage will also bring down your telephone connection.

13. Did the Internet and the World Wide Web make ISDN obsolete?

In some ways, yes. In other ways, no. For example, the services and capabilities available via the Internet today make ISDN less attractive. There is no need to subscribe to ISDN simply to have information retrieval, electronic transaction, or interpersonal messaging capabilities. All of these applications, and more, are available via the Web. Even ISDN's telephone-related services are now available for analog telephones via the telcos. Nevertheless, ISDN has become quite popular as an Internet dialup alternative to traditional analog modem service. Many people are subscribing to ISDN, not for its applications, but because it provides a faster gateway to the Internet. For example, an ISDN BRI service, with BONDING, provides a 128-kbps connection. With compression, this connection increases to over 500 kbps. Compared to a 56-kbps modem, which provides a data transmission rate of over 200 kbps with compression, an ISDN connection wins out.

14. Speaking of the Internet, what does a typical dialup ISDN Internet connection look like?

A dialup ISDN Internet connection requires ISDN service for both the user and the Internet service provider. This is usually BRI for the home and PRI for the ISP. At home, an ISDN terminal adapter is needed for any computer that does not have a built-in ISDN port. At the ISP side, an ISDN communications server (or equivalent) is needed to support remote ISDN connections. The ISP link to the Internet is generally a T1 circuit, inverse multiplexed T1 circuits, or T3 or fractional T3 circuits. This is illustrated in Figure 11.6. The point-to-point protocol (PPP) is the primary protocol used for data transmission. Furthermore, if the end user equipment supports Multilink PPP (MP), both B channels can be combined into a single 128-kbps channel.



IGURE 11.6 A typical home-based ISDN Internet connection consists of a BRI connection from he home to the telco's ISDN network and a PRI connection between the ISP and the telco's ISDN etwork. The entire connection, from home to ISP, is completely digital. Home-based ISDN Internet onnections are an alternative to conventional analog dialup connections using 28.8-kbps or 56-kbps odems.

15. This is the second time you mentioned Multilink PPP. Please explain what this is.

Multilink PPP (MP) is an IP protocol that combines multiple physical links (i.e., telephone lines) into a single, high capacity channel. The development of this protocol was precipitated by the desire to aggregate multiple ISDN B channels. Unlike BONDING, which is implemented in hardware, MP is achieved via software. MP is also applicable to analog dialup connections. For example, if each of three separate phone lines is connected to 33.6-kbps modems, then MP can aggregate these lines into a single 100.8-kbps channel. Both ISDN and analog MP solutions are supported by various Internet service providers. Thus, if your ISP offers MP dialup service, then you too can achieve higher capacity connections to the Internet via a dialup modem. For more information about MP, see RFC 1990. (*Note:* Although PPP is normally associated with IP, it nevertheless is a general purpose link layer protocol that supports several upper layer protocols.)

16. While we are on the subject of protocols, on what protocols is ISDN based?

ISDN relies on many different protocols, all of which conform to the layers of the OSI model. These protocols are defined in the ISDN I-series (see Table 11.1)—specifically the I.400 series—and are applicable to the first three layers of the OSI model. At the physical layer, the physical interfaces for both BRI and PRI are specified in I.430 and I.431. These specifications are applicable to *B, D,* and *H* channels. At the data link layer, the *D* channel is based on the ITU-T standard, Q.921, known as the *Link Access Protocol-D channel* (LAP-D). The *B* and *H* channels rely on frame relay protocols (see Chapter 12) for circuit-switched connections and an X.25 protocol called *Link Access Protocol-Balanced* (LAP-B) (Appendix E) for packet-switched connections. At the network layer, the *D* channel relies on the ITU-T standard, Q.931, for call control, and X.25 protocols for packet data. The *B* and *H* channels also use X.25 protocols for packet-switched connections. No layer 3 protocols are necessary for circuit-switched connections. A summary is given in Table 11.2.

17. Why does the *D* channel use different protocols at layers 2 and 3 than *B* and *H* channels?

As noted earlier, the *D* channel is a multipurpose channel. It is used for call-setup and call tear-down when the *B* channel is used to transmit circuit-switched user data. If no signaling or control information is needed—for example, a dedicated leased line connection is used or the *B* channel is transmitting packet-switched user data—then the *D* channel can be used to transmit packet-switched user data as well. Finally, the *D* channel can be used

TABLE 11.2 Summary of ISDN Protocols

OSI	*D* Channel			*B* and *H* Channels	
Layer	Call-control	Packet Data	Telemetry	Circuit-switched	Packet Data
3	ITU-T Q.931 / I.451	X.25 Protocols	——	N/A	X.25 Protocols
2	ITU-T Q.921 / (LAP-*D*) I.441			Frame Relay Protocols	LAP-*B*
1	ISDN I-series: I.430 (BRI) and I.431 (PRI)			ISDN I-series: I.430 (BRI) and I.431 (PRI)	

Source: Adapted from Stallings, 1997

to transmit information from telemetry services. These applications require different protocols at the data link and network layers than the *D* and *H* channels, which simply transmit circuit- or packet-switched data.

18. Could you expand on LAP-*D*? I've seen this before and I am interested in it.

Sure. The frame format for the Link Access Protocol for *D* channel is shown in Figure 11.7. Note that the frame carries encapsulated layer 3 data (Information field). Following is a brief explanation of the various fields:

Flag—Signals the beginning or ending of the frame

Address—Provides addressing information. The service access point identifier (SAPI) identifies where the layer 2 protocol provides service to layer 3. Specific addresses identify specific services. For example SAPI = 16 is for X.25 packet data transmissions. The command/response (C/R) bit specifies whether the frame is a command or response. The extended address (EA) bits specify the beginning and ending of the address field. If EA is 0, then another byte of address information follows. An EA of 1 implies that the current byte is the last byte of the address. Thus, given a two-byte address, EA = 0 in the higher-order byte and EA = 1 in the lower-order byte. The terminal endpoint identifier (TEI) represents the specific address or ID assigned to each ISDN terminal equipment connected to an ISDN network via an S/T interface.

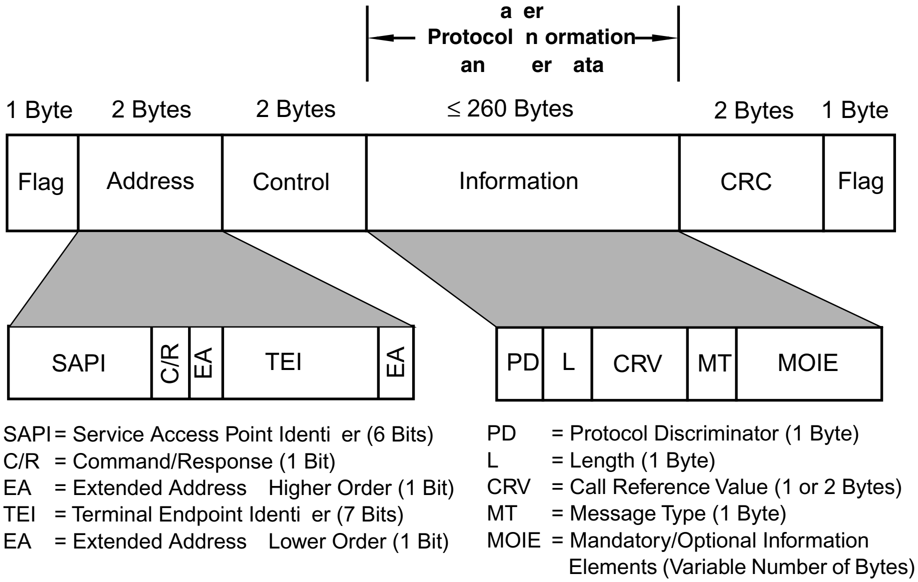


FIGURE 11.7 Frame format of ISDN's Link Access Protocol-*D* channel (LAP-*D*). Note that the information field contains encapsulated layer-3 data.

Control—Provides layer-2 control information (e.g., specifies the type of frame being transmitted, maintains frame sequence numbers).

Information—Provides layer-3 protocol information and user data. The protocol discriminator (PD) identifies the specific layer 3 protocol. The length (L) byte specifies the length of the CRV field, which is either one or two bytes. The call reference value (CRV) is the number assigned to each call. Once a call is completed, this number can be reassigned to a new call. Message type (MT) identifies specific messages related to circuit-switched connections. For example, during call-setup, the message, CONNECT, indicates that the receiving terminal equipment end node has accepted a call by the initiating TE. When a call has been completed, the message, RELEASE COMPLETE, is sent to indicate that the channel has been torn down (call tear-down). The mandatory and optional information elements (MOIE) field carries additional information specific to the message type being transmitted.

CRC—Provides for data integrity (error control) via CRC checksums (Chapter 5).

19. Are there any current ISDN initiatives that I should be aware of?

Yes. Two key ones are *always on/dynamic ISDN* (AO/DI) and *broadband ISDN* (B-ISDN). We will discuss these separately.

AO/DI Always on/dynamic ISDN is an initiative from the Vendor's ISDN Association (VIA). The concept of AO/DI is to use a portion of the *D* channel, which is always active and constantly connected to the provider's switch, to transmit user packet data. Given that most ISDN implementations assess a charge whenever a *B* channel is active—usually so much per minute for every minute the link is up—keeping a channel up can be quite expensive. For example, a 5-cent per minute charge translates to $3 per hour, which is $72 per day. With AO/DI, transmitting user data across the *D* channel is free.

AO/DI uses 9600 bps of the *D* channel's 16-kbps capacity; the remaining bandwidth (6400 bps) is used for control and signaling information. If user data transmission rates exceed the 9600 bps reserved for AO/DI, then one of the *B* channels is automatically activated (i.e., a circuit-switched connection is established) to carry the load, and the *D* channel assumes its normal role of providing control and signaling information. If additional capacity is required, then Multilink PPP is used to automatically activate and combine the second *B* channel with the first to get an aggregate bandwidth of 128 kbps. Once bandwidth requirements drop to the point where neither *B* channel is needed, then the channels automatically become inactive and the *D* channel resumes its low-capacity packet data transmission function. Thus, you can receive data without paying for any *B* channel usage until one of the *B* channels becomes active.

The primary application of AO/DI is transmitting IP packets. This offers several advantages to end users. For example, AO/DI provides users with a free, permanent connection to the Internet. As long as the IP packet transmissions can be carried satisfactorily via a 9600-bps link, then the connection is free, although your provider might assess a nominal monthly charge for AO/DI service because it will be considered part of ISDN's feature set. Examples of such transmissions include e-mail, small text files, stock quotes, and sports and headline news information. Furthermore, since the connection is permanent, users do not have to manually make a connection to the Internet every time they want to check for e-mail or stock quotes. AO/DI is also appealing to service providers because it reduces the number of *B* channels they have to nail up.

In short, AO/DI represents a technology that satisfies users and providers. Users, at best, get free service, and at worst, reduce the cost of their service. Telcos, on the other hand, still get to provide the service, but their switches become less saturated because the amount of *B* channel transmissions is reduced. AO/DI is a win-win solution for everyone.

B-ISDN Broadband ISDN is an extension of ISDN, which is sometimes called *narrowband ISDN* (N-ISDN). B-ISDN provides full-duplex data transmission at OC-12 rates (622.08 Mbps) and is designed for delivery of two primary types of services: interactive services (e.g., videoconferencing and video surveillance), and distribution services (e.g., cable TV and high definition TV). B-ISDN is also the basis for ATM (see Chapter 14). The topic of broadband is discussed in Chapter 17 on convergence.

20. I am interested in AO/DI. Is this something my ISP offers?

We can't say. You need to ask your ISP representative. AO/DI was deployed in early 1998, and AO/DI products are available from major Internet equipment manufacturers. Implementing AO/DI requires that both your telco and ISP support it. Your ISP also must have a connection to the telco's X.25 network because *D* channel packet data transmissions are carried via the telco's X.25 network (see Table 12.2). Packets are first transmitted from the end user to the telco via the *D* channel, where they are placed on the telco's X.25 network and transmitted to the ISP. The ISP then transmits these packets to their destination via the Internet. Similarly, packets arriving from the Internet are transmitted to the telco via the ISP's X.25 network connection. From there, the telco transmits the packets to the end user across the *D* channel (Figure 11.8).

21. One last question. Besides serving as an alternative to traditional Internet dialup connections, what other uses does ISDN have?

ISDN is ideal for the small office/home office (SOHO). With an NT1 module, you can connect a fax machine, a computer, and your telephone directly to an ISDN network via a single connection. Furthermore, if two *B* channels are inadequate, ISDN's flexible packaging scheme enables you to purchase additional *B* channels. ISDN also provides LAN-to-LAN connectivity, enabling an organization's remote LANs to be connected to each other or to its corporate LAN (Figure 11.9). This implementation of ISDN is a cost-effective solution if remote sites do not have to be in continuous communication with one another. If this is not the situation, then frame relay (Chapter 12) is a better alternative because ISDN phone charges can become exorbitant. Even in situations where frame relay is used for LAN-to-LAN communication, though, ISDN still can play a role. ISDN has proven to be an effective backup strategy to frame relay. ISDN can also be integrated with frame relay using fallback switches. If the frame relay network fails, the ISDN network automatically takes over. Although the transmission rate is less than what is available via frame relay, it nevertheless keeps the data flowing. Finally, ISDN provides affordable desktop-to-desktop videoconferencing. So you see, there are several niche applications that can be filled quite nicely by ISDN.

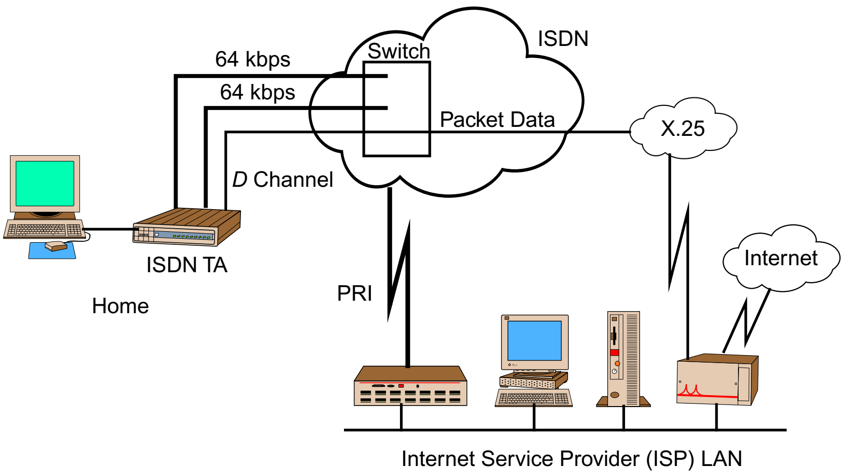


FIGURE 11.8 Always on/dynamic ISDN is ideal for Internet applications. Implementing AO/DI at home requires AO/DI support from the telco and ISP. The ISP also must have a connection to the telco's X.25 network because this is the network that handles *D* channel packet data transmissions.

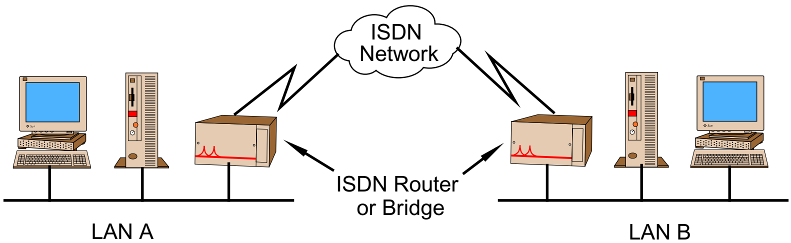


FIGURE 11.9 ISDN can be used for LAN-to-LAN connectivity. To reduce call charges, a dial-on-demand feature available in most ISDN bridges or routers will automatically call the telco's ISDN switch only when data need to be transmitted. When no traffic is seen on the port, the bridge or router will then hang up the call. This setup is exactly the same that is used for connecting a LAN to the Internet. The only difference is that one of the LANs has a link to the Internet (Figure 11.6.).

END-OF-CHAPTER COMMENTARY

In this chapter, we presented an overview of ISDN. Many of the concepts we discussed are also addressed in other chapters. For example, Chapter 1 contains a discussion of circuit- and packet-switching; Chapters 2 and 4 provide information about bandwidth and channel capacity; Chapter 3 contains information about the Internet; and Chapter 5 presents information about the data link layer. ISDN also serves as the basis for other WAN technologies and services, including frame relay (Chapter 12), SMDS (Chapter 13), and ATM Chapter 14). Finally, Chapter 15 provides information about home networking and examines various networking strategies (e.g., 56K modems, cable, and DSL) to ISDN.

Chapter 12

Frame Relay

In Chapter 11, we discussed the integrated services digital network (ISDN), which was developed in the 1970s to eventually replace the public switched telephone network (PSTN). As part of its development, ISDN was designed to provide both voice and data service. Although ISDN was considered a significant improvement to the PSTN, it was still an evolution of the PSTN and hence was circuit-switched. To support data applications in an efficient manner, a packet-switching component needed to be included with the ISDN standard. The only packet-switching technology available at the time was X.25 (see Appendix E), which did not support the ISDN model of keeping user data separate from control data. Consequently, frame relay was developed specifically as a packet-switching technology component of ISDN for data applications. Today, frame relay has emerged as a popular and robust standalone technology. In this chapter, we examine various concepts related to frame relay from both conceptual and technical perspectives. An outline of the main topics we address follows:

- Frame Relay Overview (Questions 1–3)
- Frame Relay's Physical Layer: Virtual Circuits and Committed Interface Rates (CIRs) (Questions 4–20)
- Technical Aspects and Operation of Frame Relay (Question 21)
- Frame Relay's Data Link Layer (Questions 22–28)
- Voice Over Frame Relay (Question 29)
- Frame Relay versus ATM, SMDS, and Gigabit Ethernet (Questions 30–31)
- Frame Relay in the News (Question 32)

1. What is frame relay?

Frame relay is a public, WAN packet-switching protocol that provides LAN-to-LAN connectivity. Its name implies what it does: it relays frames across a network between two sites. Frame relay is economical and efficient. It provides a single point of network access for multiple LAN-to-LAN connections. This feature offers considerable savings on local loop charges. Frame relay also can be implemented using existing bridges or routers.

2. Since it relays frames, does this make frame relay a layer-2 protocol?

Yes, and as you will learn later in the chapter, frame relay is really a "no nonsense" protocol. For example, it provides no support for error control. Instead, it relies on higher-level protocols to do error correction and to request retransmissions if packets are lost or discarded.

3. In the chapter introduction, you said that frame relay was part of the ISDN standard. How did frame relay separate itself from ISDN?

Prior to frame relay's development, LANs were primarily interconnected by dedicated (i.e., private) leased lines using point-to-point protocols or X.25. This design was acceptable if only one or two LANs required interconnectivity. However, as internetworking became more prevalent, multiport routers were needed to provide multi-LAN connectivity, and additional dedicated leased lines had to be installed (Figure 12.1a). LAN-to-LAN connectivity became a more expensive endeavor with the addition of each dedicated circuit. These costs were further escalated, and the network design became more complex, when a partially or fully meshed network design was needed (Figures 12.1b and 12.1c). In a fully-connected network, the number of links is always one less than the number of interconnected nodes or LANs. Thus, if five LANs are fully interconnected, then each LAN requires four links. Since frame relay's role in ISDN was to provide connectivity between routers, ISDN developers realized that traditional LAN-to-LAN connectivity costs could be reduced substantially if frame relay is used in place of dedicated leased lines. What frame relay provides is a single connection into a public network instead of multiple interconnections (Figure 12.2). This reduces both the cost and complexity of the network, especially in a fully meshed design. For example, in the frame relay configuration shown in Figure 12.3, each LAN only needs one link into the cloud for full interconnectivity among the five LANs. In a comparable, fully meshed private leased line network, each LAN would require four links (Figure 12.1c). Furthermore, unlike traditional private leased line service, frame relay's circuit costs are not distance based and the circuits themselves do not necessarily have to be permanent. This led to frame relay being offered and further developed as a separate protocol.

4. At the end of your answer you stated, "the circuits themselves do not necessarily have to be permanent." What do you mean by this?

In a black and white networking world, there are only two types of telecommunications links: private and virtual. Private links (also commonly referred to as standard leased lines) provide dedicated connectivity between two sites. Virtual links, on the other hand, are shared among several sites. Frame relay is a connection-oriented protocol that employs virtual links. (See Chapter 2 for additional information about virtual circuits.) As a connection-oriented protocol, frame relay must first establish a connection before two nodes can communicate. Instead of establishing and maintaining a permanent, dedicated link between a source and destination, frame relay relies on *permanent virtual circuits* (PVCs) to interconnect two sites. Thus, PVCs establish a logical connection between two sites instead of a physical one. This is what distinguishes a frame relay network from one that uses standard leased lines. Through the use of virtual circuits, data from multiple sites can

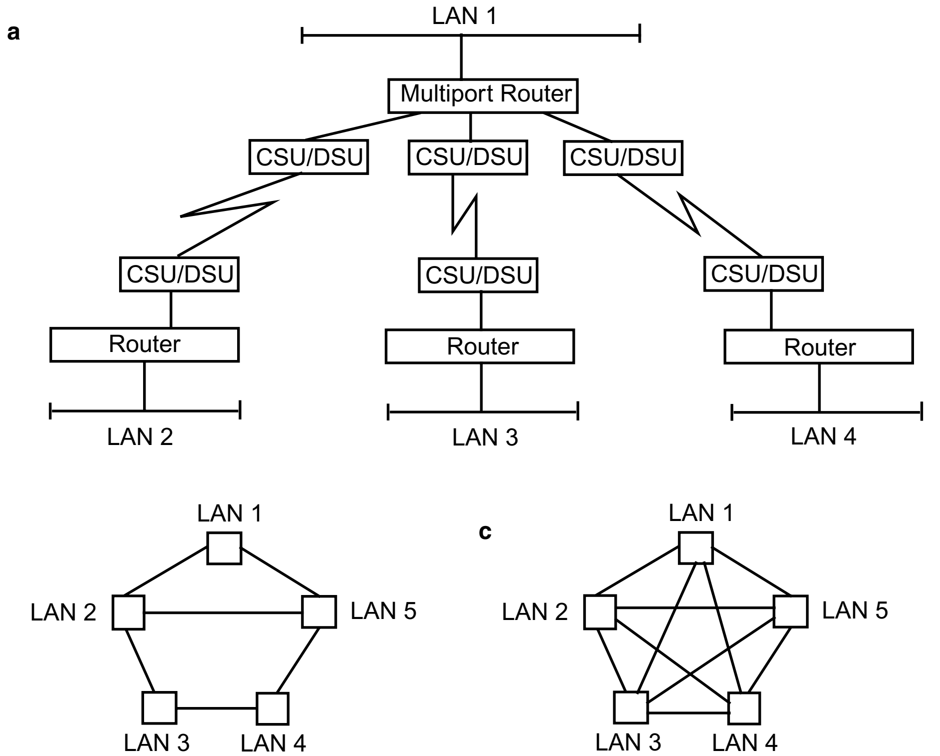


FIGURE 12.1 In a standard private line network, at least one circuit is required for LAN-to-LAN connectivity. Several topologies are possible including (a) a star configuration, (b) a partially meshed configuration, and (c) a fully meshed configuration. Note that the star configuration is the least expensive because only one LAN requires a multiport router and multiple circuits. All other LANs require a single-port router and one circuit. This configuration is also the least reliable because it has a single source of failure. An improvement in reliability calls for partially meshed or fully meshed designs. These topologies significantly increase the cost and complexity of the overall network design, though, because more than one LAN requires multiport routers and multiple circuits for LAN interconnectivity. Source: Adapted from Fitzgerald & Kraft, 1993.

be transmitted over the same link concurrently. Frame relay also supports *switched virtual circuits* (SVCs), which enable circuits between source and destination nodes to be established dynamically. We discuss SVCs later in the chapter.

5. I'm confused. How is a PVC different from a private leased line? Aren't you still using a permanent circuit with a PVC?

Sort of. PVCs do indeed have a predetermined link between a source and destination just as private leased lines. In fact, PVCs appear as private circuits because frame relay, as a connection-oriented protocol, must first establish circuits (i.e., "nail them up") between

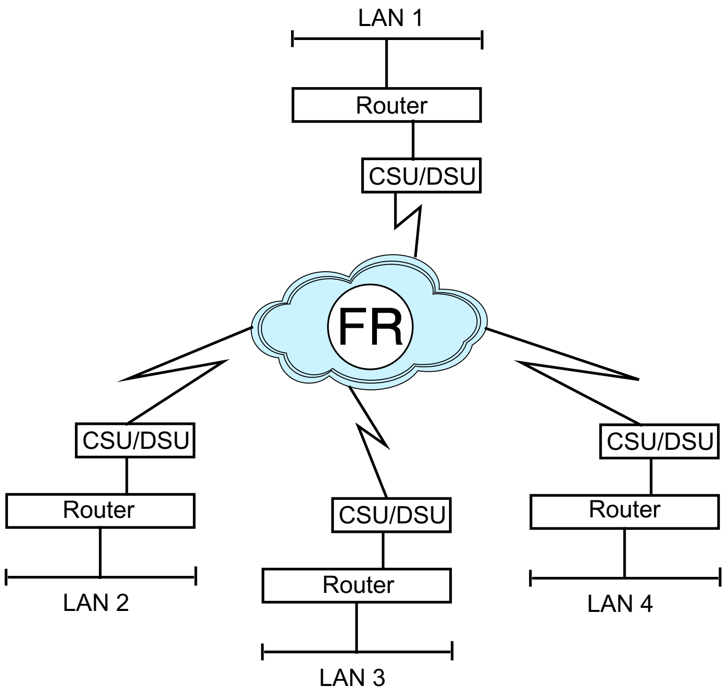


FIGURE 12.2 In a frame relay network, a single access into a frame relay "cloud" is all that is required. This simplifies the network design and makes it a less expensive endeavor.

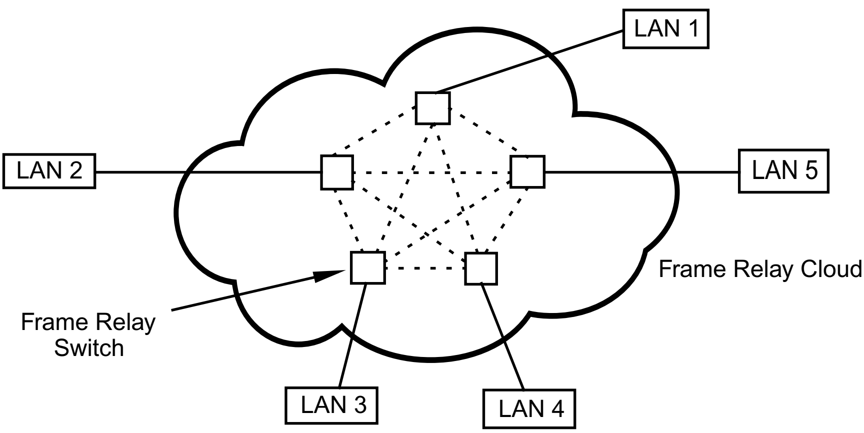


FIGURE 12.3 Frame relay provides a single connection into a public network instead of multiple connections. This translates into considerable savings in connectivity cost as well as complexity in network design, maintenance, and analysis.

end nodes prior to data communications. The difference is that PVCs are virtual circuits, not dedicated ones. This means that the bandwidth is shared among multiple sites instead of being dedicated between two sites. Thus, PVCs provide nondedicated connections through a shared medium. This is done by multiplexing a physical link so that it can be shared by multiple data transmissions (see Chapter 2).

6. Can you give me an example?

Sure. When you subscribe to frame relay service from your telecommunications carrier, a PVC is assigned between your LAN and all of your organization's other LANs that require interconnectivity. For example, in the network shown in Figure 12.4(a), four PVCs are used to provide connectivity respectively from LANs 2, 3, 4, and 5 to LAN 1. These PVCs are shown as one-way arrows, which imply that the PVCs are simplex links (i.e., data transmissions are unidirectional). Note that the links that connect a LAN to the frame relay network are not PVCs. Only the links within the cloud are PVCs. If two-way connectivity is needed, then duplex PVCs are used. A fully meshed design is also possible. This is shown in Figure 12.4(b). A fully meshed design involving five LANs is accomplished using either 10 duplex PVCs or 20 simplex PVCs. The use of duplex or simplex PVCs for meshed designs is a function of the local exchange carrier providing the frame relay service. Regardless of the assignment, though (simplex or duplex), a fully meshed design still only requires a single network connection from each LAN.

7. Are simple and fully meshed designs the only two choices one has in configuring a frame relay network?

No. You can also have a partially meshed design in which some nodes are fully interconnected and others are not. Configuration decisions are based on customer needs and network traffic requirements.

8. I can see the advantage frame relay has over private links, but aren't you still paying for all that bandwidth to interconnect the LANs?

Yes, but the cost is substantially less because PVCs have associated with them a *committed information rate* (CIR). A CIR is the amount of throughput a frame relay provider guarantees to support under normal network loads. A CIR, which can range from 16 kbps to T3 (44.8 Mbps), is assigned to a PVC as part of network configuration. It is the minimum guaranteed throughput of a PVC.

9. Does a CIR represent a fixed amount of bandwidth?

No. Unlike a private leased line, which commits a fixed amount, a frame relay provider calculates the average amount of traffic that has been transmitted across a PVC over a specified period of time (e.g., 1 second). Using this information, the provider then determines the average amount of bandwidth that has been used. This serves as the basis of the CIR on the provider's part. If a PVC's assigned CIR (which, again, was set when the network was first configured) is greater than or equal to this average, then data transmissions are guaranteed. If the assigned CIR is less than this average (i.e., the "pipe" is not big

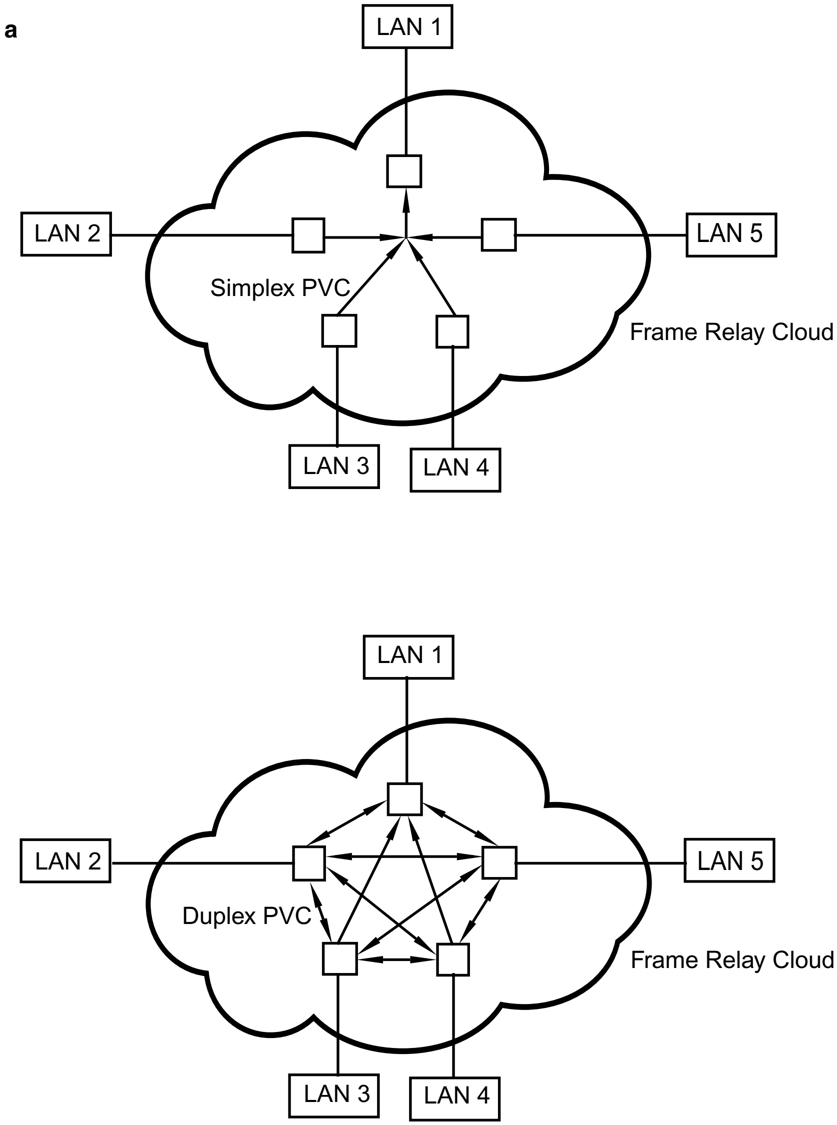


FIGURE 12.4 A frame relay network consisting of five LANs requires at least four simplex PVCs for LAN interconnectivity. In (a), simplex PVCs are used to connect LANS 2, 3, 4, and 5 to LAN 1. In this illustration, LANS 2, 3, 4, and 5 might be remote sites and LAN 1 might be corporate headquarters. In (b), a fully-meshed design using duplex PVCs is shown. Here, every LAN can communicate with every other LAN using only a single connection to the frame relay network. Note that a fully meshed design requires $n(n - 1)/2$ full-duplex PVCs, where n is the total number of LANs to interconnect. Source: Adapted from Fitzgerald, 1993.

enough), then data transmissions are not guaranteed. Thus, the assignment of a CIR to a PVC is extremely critical to both network cost and performance. If a PVC's CIR is too little, then when the network becomes congested, frames may be dropped. On the other hand, if the CIR is too high, then you are paying for excessive bandwidth. This is analogous to the way banks calculate the monthly maintenance fee of a checking account. If the monthly average of the funds in the account is less than a prescribed minimum, then a fee is assessed. If, however, the monthly average grossly exceeds the minimum average, then you probably have too much money in the account. The penalty for having too much money in a checking account is usually lost interest, since many checking accounts pay no or little interest.

10. Is the CIR of a duplex PVC the same in both directions?

Not necessarily. A CIR can be assigned to a PVC either symmetrically or asymmetrically. A symmetric CIR guarantees the same amount of bandwidth in each direction of a duplex PVC; an asymmetric CIR permits different bandwidth guarantees to be committed in each direction. This flexibility in CIR assignments is one of frame relay's greatest features. The support of asymmetric PVCs makes frame relay an ideal service for client/server applications. For example, Internet or intranet-based Web servers could be configured to have inbound CIRs two or three times (or more) greater than their outbound CIRs. This accommodation of different data transmission rates for inbound and outbound traffic can result in considerable savings for an organization (Figure 12.5).

11. What happens if a data transmission exceeds the CIR?

Data transmissions that exceed the CIR will be transmitted by the frame relay service provider on a "best effort" basis. This means the service provider will attempt to deliver the data but will not guarantee delivery. At best, data frames that require more bandwidth for delivery than what is called for by the CIR will be transmitted without any problem. At worst, the frames will be discarded and will have to be retransmitted.

12. Please expand on this concept of transmitting frames that require more bandwidth than what the CIR provides.

When a data transmission exceeds the CIR, it is referred to as a *burst*. Two types of bursts are defined in frame relay. The first, called the *committed burst* (B_c), is the maximum amount of data the provider guarantees to deliver within a specified time period, T. Note that CIR = B_c/T. Given that most providers use a 1-second time interval to calculate the average amount of bandwidth utilization, CIR is usually equal to B_c. The difference between these two parameters is their units. CIR is measured in bps; B_c is measured in bits. The second type of burst, called the *excessive burst* (B_e), is the maximum amount of *uncommitted* data a provider will attempt to deliver within a specified time period. In other words, a provider will guarantee a committed burst of B_c bits and will attempt to deliver (but not guarantee) a maximum of $B_c + B_e$ bits. For example, a PVC assigned a 128-kbps CIR might have associated with it an excessive burst rate of 64 kbps. This means that the provider will attempt to support data transmissions requiring a capacity of up to 192 kbps (Figure 12.6).

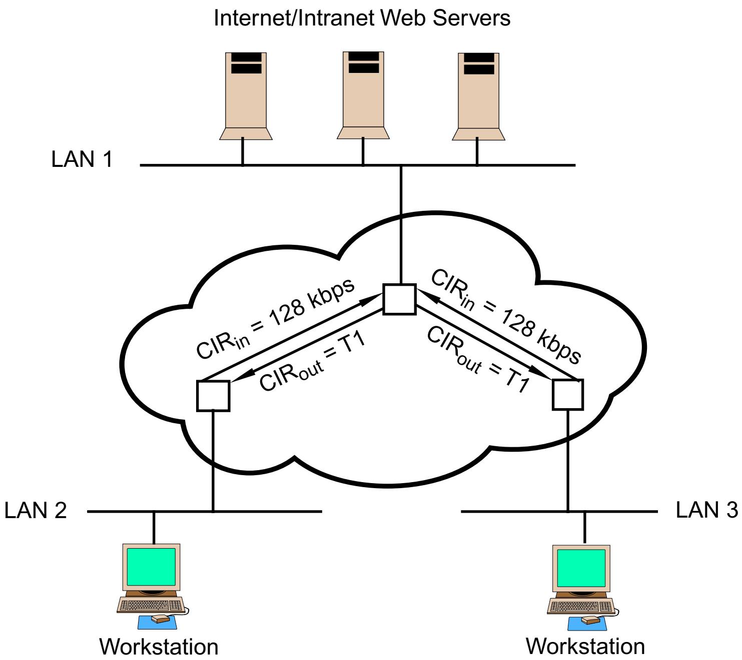


FIGURE 12.5 A frame relay network can be configured with asymmetric PVCs to accommodate different traffic flows. In this figure, LAN 1 consists of a configuration of Web servers and LANs 2 and 3 consist of workstations that access these servers. Client Web requests usually consist of small URLs, but Web server responses are usually large graphic files. Thus, LAN 1's outbound link is configured to have greater bandwidth than its inbound links. Source: Adapted from Wu, 1997

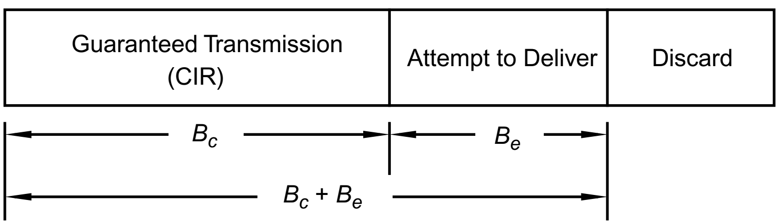


FIGURE 12.6 Data frames that fall within an agreed-upon committed information rate (CIR) are guaranteed transmission. This is called the *committed burst* (B_c) and represents the maximum amount of data the frame relay service provider guarantees to deliver within a specified time. Under normal conditions, the provider will also attempt to deliver (but not guarantee) an additional amount of data beyond the CIR. This extra amount is called the *excessive burst* (B_e). Thus, a provider will guarantee delivery of B_c bits, but will attempt to deliver a maximum of $B_c + B_e$ bits. Anything greater than this sum will be discarded.

13. I understand the concepts of B_c and B_e. What I don't understand is how you can burst to a higher rate than the CIR. Isn't the CIR the maximum link capacity?

Good question, and we can understand your confusion. We forgot to mention one other parameter—*port speed*. In addition to defining logical connections (PVCs) and bandwidth requirements (CIRs), you must also determine the appropriate port speed of the physical link that connects your LAN to your provider's frame relay network. This link, called the *port connection* or *access line*, is the local loop connection between your LAN's frame relay end node (usually a router) and the provider's frame relay switch. At the customer's site, the access line is connected to the end node's interface and is called the user-to-network interface (UNI). Depending on the carrier's policy, port speeds can be less than, equal to, or greater than the sum of the CIRs for a particular port. For example, in Figure 12.7, LAN 1 consists of Internet/Intranet Web servers, which must serve up Web pages to the clients on LANs 2 and 3. Note that the sum of LAN 1's originating CIRs is 256 kbps + 256 kbps = 512 kbps. Further note that LAN 1's port speed is T1, which is more than twice the capacity of the sum of its CIRs. LAN 2, however, demonstrates a condition known as *oversubscription*. Its port speed (128 kbps) is less than its aggregate CIRs (192 kbps). LAN 2 has oversubscribed its connection—the capacity of its connection into the frame relay network is less than the total bandwidth guaranteed by the provider. Finally, LAN 3's port speed is equal to the aggregate of its originating CIRs.

Frame relay service providers will attempt to deliver frames that exceed a CIR if two conditions are met. First, the data bursts cannot be greater than the port speed, and second, the provider must have sufficient bandwidth available within its own network to accommodate the burst. Burst rates are only supported by a carrier for a limited time period (e.g., 2 seconds). The actual time period in which burst rates are supported varies among carriers. Thus, given the configuration of Figure 12.7, the provider will attempt to support data transmissions from LAN 1 up to T1 capacity.

In a private leased line network, the issues of port speed and CIR are irrelevant—if a T1 line is provisioned, the port speed is 1.544 Mbps and it is a fixed rate. With frame relay, though, we now have considerably more flexibility in configuring the bandwidth requirements of our LAN-to-LAN connections. The trade-off for this flexibility is more detailed knowledge and understanding of the traffic patterns and flows of these connections. Given the typical bursty nature of LAN traffic, it is usually prudent to set the CIR below the port speed to accommodate for data bursts. One rule of thumb is to ensure that the total CIR of a connection does not exceed 70 percent of the port speed. Thus, if the total CIR is 256 kbps, then the port speed should be no less than 384 kbps. It is also important to note that some providers place a restriction on the maximum burst rate permitted. The bottom line is that you need to discuss these issues with your provider.

14. I am fascinated by the concept of oversubscription. How can a provider guarantee a certain amount of bandwidth but provision a link between a LAN and the frame relay network that contains less capacity than what is guaranteed?

It has to do with multiplexing and buffers. Frame relay uses statistical multiplexing (see Chapter 4) for channel allocation, and providers' frame relay switches have large buffers. Given a sufficiently large buffer size, coupled with a low probability that not every

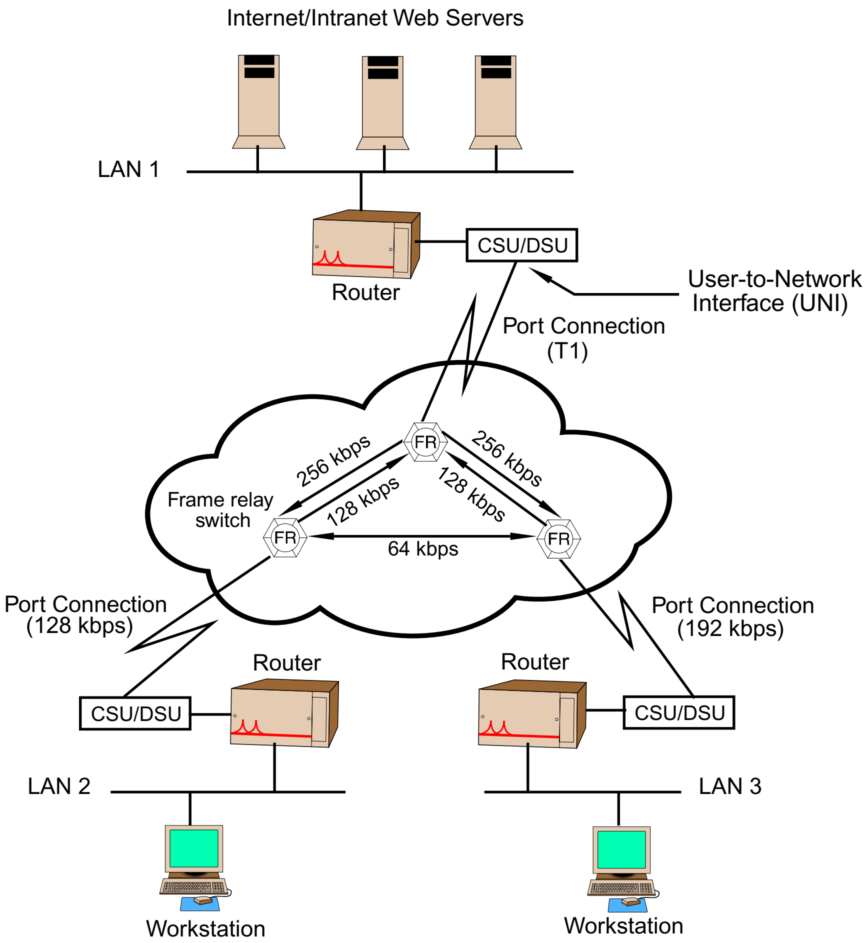


FIGURE 12.7 In a frame relay network, the port connection is the local loop circuit that connects a LAN's frame relay node (usually a router) to the service provider's switch. The capacity of this connection, called the *port speed*, can be less than, equal to, or greater than the sum of the CIRs. In this illustration, LAN 1's port speed is 1.544 Mbps, which is more than twice as great as the sum of its originating CIRs. Thus, LAN 1 can support data bursts of up to T1 speeds. In LAN 2, however, the sum of the originating CIRs (192 kbps) exceeds the port speed (128 kbps). This is a situation known as *oversubscription*. LAN 2 has oversubscribed its service since its aggregate guaranteed bandwidth is greater that its port speed. LAN 3's port speed is equal to the sum of its originating CIRs. Source: Adapted from Wu, 1997

user will need the channel at the same time, it is possible for a provider to deliver data frames successfully to a LAN that has oversubscribed its service. Not all frame relay providers permit oversubscription, though. Some mandate that the port speed be at least equal to the aggregate CIRs.

15. Given the concept of oversubscription, what's to stop a customer from establishing a CIR of zero and effectively stealing bandwidth from the provider?

Nothing. Some service providers offer a CIR of zero as an option and even encourage customers to subscribe to this service to get an accurate assessment of bandwidth needs. In some cases, this might be a little more expensive than establishing nonzero CIRs, but it also can be beneficial if traffic rates between sites are unknown. After a few months, specific traffic usage patterns will emerge and appropriate CIRs and ports speeds can then be set. Be careful though. Remember that a CIR is *contractual* bandwidth. It represents the maximum data transmission rate a service provider *guarantees* to provide. A CIR of zero implies that every date frame transmitted on the network is considered a burst and hence, delivery is not guaranteed. If the provider has surplus bandwidth (e.g., unused capacity from customers who are paying for a certain CIR but are not using all of it) available within its network, then you might not experience any network transmission problems with a CIR of zero. However, as this surplus diminishes, expect to see pronounced degradation in the performance of your LAN-to-LAN connections.

16. What happens if you need to increase or decrease the CIR of a PVC?

You contact your provider and request the change. That's the beauty about CIR: If a higher-capacity circuit is needed, only the PVC's assigned CIR has to be changed in the provider's database; a new circuit does not have to be provisioned (as is the case with private leased lines) because the circuits are virtual.

17. What other concepts or issues do I need to know about frame relay besides PVCs and CIRs?

There are several, including switched virtual circuits (SVCs), voice over frame relay, video over frame relay, ATM vs. frame relay, and the technical aspects of frame relay.

18. Let's start with SVCs. What are they and how are they different than PVCs?

Switched virtual circuits are sort of like cousins of permanent virtual circuits—they share certain similarities (e.g., they are both virtual circuits), but have enough dissimilarities to make them different. As we discussed earlier, frame relay is a connection-oriented protocol and hence, a circuit must first be established (i.e., "nailed up") between end nodes prior to data communications. With PVCs, two sites are permanently interconnected with a circuit similar to the way two sites are interconnected using private leased lines. The difference is PVCs are shared by other subscribers within a provider's frame relay network. With SVCs, however, circuits between source and destination nodes are established on the fly and then removed after data communications have ended. This makes SVCs logical *dynamic* connections instead of logical permanent connections as with PVCs. Thus, SVCs provide switched, on-demand connectivity.

An SVC-based frame relay network is analogous to the public switched telephone network. When a node on LAN *A* needs to send data to a node on LAN *B*, the frame relay switches within the frame relay network automatically nail up a circuit from source to destination prior to data communications. This circuit remains in effect only for the duration of a transmission. When the transmission is over, the switches then tear down the circuit. Thus, the path of every communication between source and destination nodes is not necessarily the same with SVCs as it is with PVCs. Furthermore, frame relay switches that support SVCs can automatically configure a path on demand between two sites, and then remove this path from their tables at the end of a transmission. With SVCs, any frame relay subscriber can communicate with any other frame relay subscriber provided both have SVC capabilities.

19. Are SVCs assigned CIRs like PVCs?

Yes. Everything we discussed earlier about CIRs applies to SVCs as well.

20. Are there any advantages or disadvantages to using PVCs or SVCs?

PVCs and SVCs have their own set of advantages and disadvantages. PVC advantages include widespread availability (every frame relay provider supports PVCs), less complex network designs (any two sites that want to communicate must have a permanent connection between them), and less expensive equipment (switches do not have to automatically configure and remove paths dynamically). On the other side of the coin, PVCs are permanent. This implies that regardless of use, you are always paying for a specific amount of bandwidth. A second disadvantage is that every time a new connection is required, a new permanent circuit must be established. Thus, the number of PVCs a customer might need may increase dramatically, making the PVCs difficult to manage. This is especially true if a fully meshed design is required. For example, an organization with 50 sites will require (50)(49)/2 = 1225 PVCs for a fully meshed network. If full connectivity is required among 100 sites, the number of PVCs needed increases to 4950.

Compared to PVCs, SVCs are more versatile. Customers do not have to establish permanent circuits between any two sites because connectivity is provided on an as-needed basis, and as a usage-based service, bandwidth is only used when needed. This can translate to considerable savings for customers. On the other hand, PVCs' advantages are currently SVCs' primary disadvantages. First, widespread SVC availability among frame relay service providers is lacking (although MCI WorldCom and Sprint deploy SVCs). SVC-based frame relay networks are also more complex in design, and switches are more sophisticated and hence expensive.

21. I think I am ready to learn about some of the technical aspects of frame relay. Can you begin with a brief technical overview of frame relay and how it works?

You bet. As you observed at the beginning of this chapter, frame relay is a data link layer protocol. It is synchronous in nature and is based on the concept of packet-switching. Thus, every frame relay frame carries source and destination addresses. As indicated earlier, frame relay also uses statistical multiplexing. This enables multiple subscribers to share the same backbone. Although frame relay operates at the data link layer, it does not provide flow control, error detection, frame sequencing, or acknowledgments. These tasks, which represent overhead, are performed by frame relay end nodes (usually a router) at the customer's site and not by the frame relay switches. If the network becomes congested and a frame relay switch's buffers get filled, the switch will discard any subsequent frames it receives until its buffers are free. Recovery from these lost frames is left to the frame relay end nodes. Transmission errors are also ignored by the switches. All frame integrity checks are, once again, performed by the customer's end nodes. Freeing the protocol from these tasks makes frame relay a very fast and highly efficient LAN-to-LAN connection.

The physical components of a frame relay network include end nodes, frame relay switches, and communication links. End nodes serve as the interface between a customer's site (LAN) and the service provider's network; switches are responsible for transmitting data from the source LAN's end node to the destination LAN's end node. End nodes are then connected to switches via an access line (i.e., a port connection), and switches are interconnected via PVCs or SVCs.

Frame relay end nodes include bridges (see Chapter 6), routers (see Chapter 7), special concentrators, and workstations or personal computers. Of these devices, the most common implementations use frame-relay capable routers or conventional routers with frame relay-capable CSUs/DSUs. Collectively, end nodes are known as frame relay access devices (FRADs). The term FRAD once meant special frame relay devices that were used for simple implementations. Today, however, the term is all-encompassing and implies any frame relay end node. Regardless of the type of device used, end nodes assemble or disassemble frame relay frames between two LANs. FRADs accept data frames from the local network (e.g., an Ethernet/802.3 frame) and assemble them into frame relay frames (Figure 12.8) by encapsulating the data contained in layers 3 through 7. This new frame is then placed on the access line for transmission across the network. When a FRAD receives a frame relay frame, it disassembles it by stripping off the frame relay headers and trailers, does any necessary reassembly of the LAN frame, performs an integrity check on the frame, and (assuming the frame is valid) places it on the local backbone for delivery to the destination node. Frame relay is also protocol independent. Thus, different LAN protocols such as Ethernet/802.3 and token ring, and different network protocols such as TCP/IP, SNA, and IPX, can run over frame relay.

22. Could you review the contents of the frame relay frame?

Sure. The frame relay frame (Figure 12.8) is of variable length; the standard does not specify a frame size limitation. The maximum size is set by (and is different among) providers and is generally a function of hardware limitations. Typical implementations range from 1600 to 4096 bytes. This implies that frame relay can fully support Ethernet/802.3 and token ring frames without having to segment (and reassemble) LAN frames for transmission across the frame relay network. Although this variable size feature makes the protocol compatible with the type of bursty traffic inherent in LANs, it also makes it susceptible to data processing delays across the network.

As shown in Figure 12.8, a frame relay frame consists of the following fields: start and end frame delimiters (SFD and EFD), address, user data, and frame check sequence (FCS). The SFD and EFD fields are one byte each and consist of the bit pattern 01111110 to indicate the beginning and ending of a frame. The address field is two bytes and can be extended to either three or four bytes. The components of a two-byte address are shown in Figure 12.8. These components include a 10-bit *data link connection identifier* (DLCI)

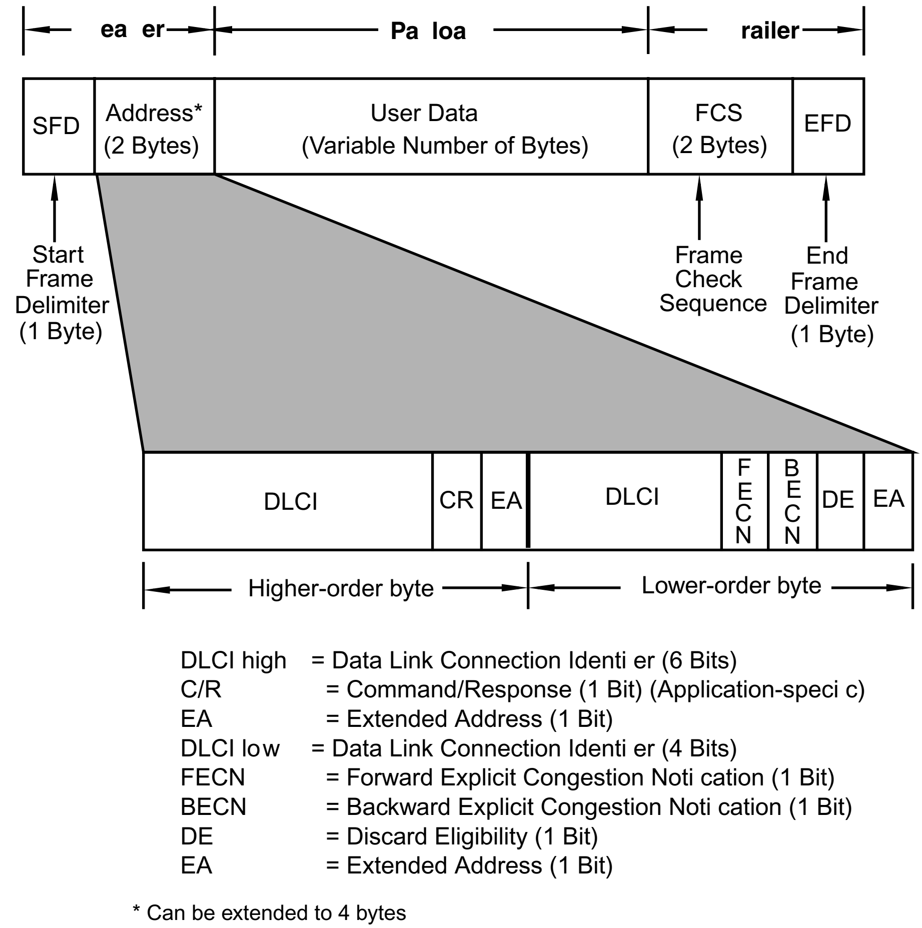


FIGURE 12.8 The contents and format of a frame relay frame. Source: Adapted from Parsons & Beach, 1996.

that is split between the two bytes, two 1-bit *extended address* (EA) fields (one per byte), and 1-bit each for *command/response* (C/R), *forward and backward explicit congestion notification* (FECN and BECN), and *discard eligibility* (DE). The user data field is of variable length. This field contains encapsulated data from the sending node. The FCS field is used to check the integrity of the frame.

Focusing on the two-byte address field, the DLCI represents the network address of the frame and includes the virtual circuit number that corresponds to the destination port of the destination LAN's end node (more on this later). The C/R bit is application-specific and not used by the protocol. It is passed transparently from switch to switch. The EA bits specify whether the address is extended to three or four bytes. If EA is 0, then another byte of address information follows; an EA of 1 implies that the current byte is the last byte of the address. Thus, a two-byte address has EA = 0 in the higher-order byte and EA = 1 in the lower-order byte. The FECN and BECN bits are used to convey congestion information to end nodes in either direction. These bits are set by frame relay switches as the frame is being transmitted across the network from source to destination. If FECN or BECN is set to 1, then a sending or receiving end node, upon receipt of a frame, will know that the frame encountered congestion and can take whatever action is necessary to enact flow control. The DE bit specifies whether or not a frame should be discarded when the network gets congested. Frames with the DE bit set to 1 are considered low priority. When the network becomes congested, if a customer's capacity exceeds its CIR, frames with DE = 1 are the first to be dropped by frame relay switches. A more detailed explanation of DLCI, FECN, BECN, and DE is given in answers to subsequent questions.

23. Please explain DLCIs a little more and give me an example of how they work.

OK. Data link connection identifiers are virtual circuit addresses assigned to PVCs or SVCs. DLCIs enable multiple virtual circuits, which represent logical connections, to be multiplexed using a single network link. For a two-byte address, DLCIs are 10 bits in length (see Figure 12.8). This implies there are $2^{10} = 1024$ possible circuit numbers. Of these, however, only 992 are available for use. The remaining 32 are reserved. For example, DLCI 0 is reserved for call control, which establishes and releases a logical connection, and DLCI 1023 is reserved for exchanging information about the virtual circuits that have been established. To accommodate larger networks, the frame's address field can be extended to three or four bytes. A three-byte address field supports a 17-bit DLCI, and a four-byte address field supports a 24-bit DLCI. Outside of the frame relay cloud, DLCIs represent the destination network's end node's port number. End nodes maintain a cross-connect table that maps its port's DLCI to a specific network address. For example, in a TCP/IP-based network, the end node (e.g., a router) is configured so that each IP address assigned to WAN interface corresponds to the correct DLCI. An illustration is shown in Figure 12.9.

DLCIs are assigned at call set-up time, or they are premapped to a destination node when PVCs are initially established with a service provider. This latter approach is used in the majority of frame relay networks. Although DLCIs must be unique within the cloud, end nodes may use the same DLCIs. For example, in Figure 12.9, the UNIs of LANs 2 and 3 are assigned the same DLCI (83). In this context, DLCIs have only local significance, and care must be exercised to ensure that the DLCI is not announced outside the local arena. In contrast to local addressing, global addressing is also available. In this context, a DLCI is assigned on a universal basis throughout the entire network. Thus, a globally-assigned DLCI would identify the same destination regardless of where the originating end node is located in the network. Although global addressing simplifies the management of DLCIs, it also reduces further the number of DLCI numbers available for use since no DLCIs can be reused by different networks.

Data transfer in a frame relay network involves first establishing (i.e., "nailing up") a logical connection between the source and destination nodes and assigning a unique DLCI to the connection. Data frames are then transferred across the network with each frame

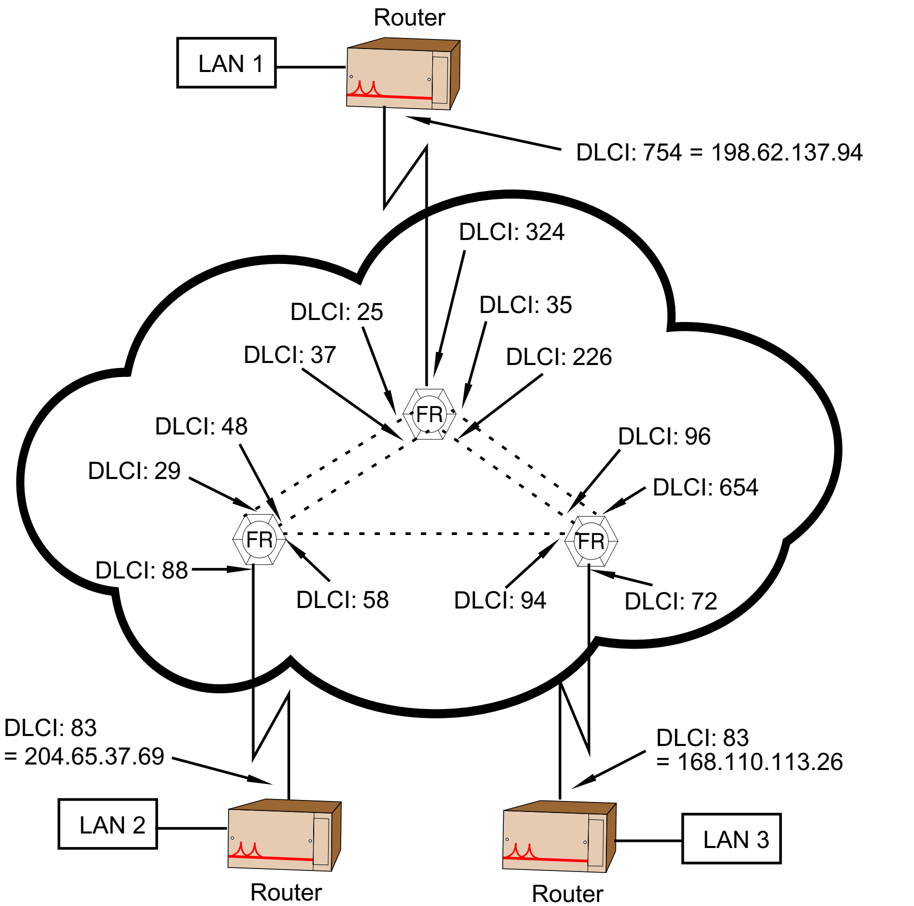


FIGURE 12.9 The virtual ports (PVCs and SVCs) of a frame relay network are uniquely identified by the data link connection identifier. Cross-connect tables within the frame relay switches map incoming DLCIs to corresponding outgoing DLCIs. For example, the virtual connection between the switches connecting LANs 2 and 3 maps DLCI 58 to DLCI 94. End nodes (routers) also must maintain tables that map port addresses (i.e., the user–network interfaces) to the correct DLCIs. Although DLCIs must be unique within the cloud, end nodes can use the same DLCIs. For example, the UNIs of LANs 2 and 3 are assigned the same DLCI (83). Thus, DLCIs have only local, not global, significance. Source: Adapted from Parsons & Bahr, 1996.

containing the assigned DLCI. The same circuit is used for the duration of the transmission. At the conclusion of the transmission, the circuit is released. Logical connections are established and released in frame relay by using frames with DLCI = 0. These frames, called *call control frames*, are exchanged between the sending and receiving nodes and include information related to the proposed connection. Among this information is the

DLCI that will be assigned to the connection. Either the sending node includes this DLCI in its connection request, or it can be assigned by the receiving node at the time it accepts the sender's connection request.

To understand the process of frame relaying, let's assume that a data frame is transmitted from LAN 2 to LAN 3 in Figure 12.9. When LAN 2's router constructs the frame for transmission, it will assign a DLCI of 83. The router places the frame on its access line and the frame is transmitted to the switch connected at the other end of the link. When the switch receives the frame, it checks the frame's integrity via the two-byte FCS contained in the frame's trailer. If any errors are found, the switch discards the frame. If the frame is valid, the switch then checks its routing table to determine the channel on which the frame should be placed. (We will assume the frame is valid throughout this illustration.) Let's agree that the switch's routing table shows frames arriving via circuit 88 must go out the port assigned DLCI = 48. The switch then rebuilds the frame to include this new DLCI and ships it out the appropriate port. The next switch (a) receives the frame on the port assigned DLCI = 37, (b) does its integrity check, (c) consults its routing table and discovers that frames arriving on circuit 37 get transmitted to the port assigned DLCI = 35, (d) rebuilds the frame to include a DLCI of 35, and then transmits the frame via circuit 35. This process continues until the destination LAN's end node (in this case, LAN 3's router) receives the frame.

24. If switches discard invalid frames, who notifies the end nodes?

No one. If acknowledgments are not received within a certain time interval, the source end node assumes that the frame was discarded. It will then retransmit the frame. One thing you can count on frame relay to do is discard frames. It is a no-nonsense protocol. Frame relay also does not operate too well in a noisy environment. If there is any line noise during a transmission, bit errors will occur and you can bet your house frame relay will drop those frames.

25. Yes, but if transmission errors occur frequently and frame relay starts discarding frames, which causes retransmissions, don't we get into a vicious circle? You know: Discarded frames lead to retransmission, which leads to more network traffic, which leads to network congestion, which causes more frames to be discarded.

Yes. Your observation is correct.

26. Well, if this is the case, and frame relay doesn't provide flow control, how, then, is congestion addressed in frame relay?

Congestion is addressed in frame relay from two perspectives. The first perspective is grounded in management and design issues. Frame relay service providers should always strive to provide an agreed-upon quality of service. To do so, though, and to do it consistently, service providers should: (a) design their networks so that sufficient bandwidth is provisioned; (b) ensure that all links are clean; (c) keep customers from establishing connections where the aggregate CIRs exceed 70 percent of the port speed; (d) ensure that any one end node is kept from monopolizing the network at the expense of other end nodes; and (e) distribute resources across the network in a fair and equitable manner. Unfortunately, these considerations are carrier dependent and vary from one provider to another.

The second perspective is standards-based. The frame relay protocol provides a mechanism for addressing congestion. This is done via the forward and backward explicit congestion notification bits (see Figure 12.8). FECN and BECN are designed to inform (i.e., notify) end nodes that a frame has experienced network congestion. If congestion emerges along a virtual circuit, these bits are set to 1 by the frame relay switch connected to that circuit as the frame passes through the switch. The FECN bit is set in frames going toward the destination node; the BECN bit is set in frames going toward the sending node.

Of the two bits, congestion notification via BECN produces the most direct action. If the sending end node receives a frame with BECN = 1, it can simply reduce (or temporarily suspend) its transmission rate. This action is direct and effective. It's kind of like a form of punishment. If you scold or spank your child the moment he or she "acts up," the effect is a very rapid inhibition of an unwanted behavior. A different strategy needs to be employed by the destination end node if it receives frames with FECN = 1. Since the destination node cannot directly slow down the sender's data transmission rate, it must rely on indirect methods. One method is to increase the length of time receipt-of-frame acknowledgments take when sent to the sender. This will slow down acknowledgment transmissions. After the sending node's acknowledgment timers have expired several times, it will eventually increase the timer interval to accommodate the new rate being used by destination node. This action indirectly causes the sender to reduce its transmission rate. A second method is to effect flow control in the higher layers. For example, if a TCP-based application such as FTP is running over TCP, and IP is being used as the network layer protocol (layer 3), then the destination node can reset its transmission window value to zero. (Recall from Chapter 3 that a TCP/IP transmission window represents the maximum number of bytes a remote node can accept.) Doing so informs the sender that the maximum number of bytes the receiver can accept is zero. Thus, a zero window informs the sender to stop transmitting data to the receiver until a nonzero window size is received from the receiver.

Incorporating the FECN and BECN bits in the frame relay frame produces shared responsibility between the network and the end nodes for congestion control: The network's role is to monitor itself for congestion and then notify the end nodes that congestion exists. Upon notification, the end nodes then take the necessary action to reduce the flow of frames onto the network.

27. Where does the discard eligibility bit fit into all of this?

Ah! The discard eligibility bit. We forgot all about it. Thanks for reminding us. Besides FECN and BECN, the DE bit can be used as another strategy for congestion control. Its role is to give the network guidance in determining which frames should be discarded. This is important because setting FECN and BECN bits does not guarantee that end nodes will respond to this notification, or if they do, that the response will be in a timely manner. Furthermore, the implementation of FECN and BECN in frame relay is not mandatory. Without any guidance, the network (i.e., frame relay switches) will discard frames arbitrarily at the onset of congestion. Although discarding frames randomly produces the desired effect—it helps reduce congestion—it does not give any consideration to frames that have a "contractual right" to remain. For example, frames transmitted within the parameters of a contracted CIR are just as likely to be dropped as frames transmitted at a burst rate above the CIR. This situation can be resolved by configuring end nodes to set the DE bit to 1 for all frames transmitted at a rate higher than the CIR. Now, in the presence of congestion, network switches will discard frames with DE = 1 before they discard frames that do not have their DE bit set. Discarded frames will then be retransmitted at a later time when congestion has subsided. (*Note:* Retransmission occurs after the sending node has detected that the packet was dropped, that is, after it failed to receive an acknowledgment.) The DE bit also can be used to distinguish between high- and low-priority frames. For example, if e-mail data are considered more important than Web data, then the end node can be configured to set the DE bit of all Web-based frames. This gives e-mail frames a higher transmission priority than Web frames, because if congestion occurs, switches will drop Web frames first. The DE bit enables frame relay customers to adopt a frame-discard strategy that is predicated on their preferences. Although this provides a more fair and equitable approach to discarding frames than arbitrary selection, it does not guarantee that only DE-flagged frames will be dropped.

The topics of congestion and congestion management strategies are part of a bigger picture called network performance, which includes metrics such as network uptime, delays, and frame or packet loss. These metrics vary from one provider to another. For example, network uptime guarantees can range from 99 percent to 99.95 percent to 99.99 percent to 99.999 percent among different providers. Although these rates are relatively close to one another, the difference in the number of minutes (or hours) of downtime is staggering. For example, uptimes of 99 percent, 99.5 percent, and 99.99 percent translate to downtimes of 1 percent, 0.5 percent, and 0.01 percent, respectively. Table 12.1 shows the differences in these rates measured daily, monthly, and yearly. Thus, a network provider that guarantees a 99 percent uptime measured on a monthly basis means that the guarantee does not become effective until the network has been down for more than 432 minutes (7.2 hours) over a 30-day period. If the guarantee is on an annual basis, then it does not take effect unless downtime exceeds 5256 minutes (87.6 hours) over the 365-day period. These same downtimes for a 99.99 percent guarantee are 4.32 minutes per month and 52.56 minutes per year. As you can see, there is quite a difference. The bottom line is customers who are subscribing to frame relay service need to investigate, probe, and negotiate network performance issues with their provider.

28. As a data link layer protocol, what provisions does frame relay have for link management, which is a layer 2 function?

The frame relay protocol specifies a *link management interface* (LMI) for link control. LMI provides an interface for link status information to be exchanged between an end node (e.g., router or switch) and the network. LMI's functions are quite basic, though, and limited to activities such as PVC notification (end nodes are notified whenever PVCs are added to or removed from the network), PVC monitoring (circuits are monitored so end nodes know which ones are available), and link establishment between an end node and

TABLE 12.1 Network Downtime Comparisons at Different Levels of Uptime Guarantees

	99% Uptime	**99.5% Uptime**	**99.99% Uptime**
Measured Daily 24 hr/day × 60 min/hr = 1440 minutes/day	1440 × .01 = 14.4 minutes downtime/day	1440 × .005 = 7.2 minutes downtime/day	1440 × .0001 = 0.144 minute downtime/day
Measured Monthly 1440 min/day × 30 days/month = 43,200 minutes/month	43,200 × .01 = 432 minutes = 7.2 hours downtime/month	43,200 × .005 = 216 minutes = 3.6 hours downtime/month	43,200 × .0001 = 4.32 minutes downtime/month
Measured Yearly 1440 min/day × 365 days/year = 525,600 minutes/year	525,600 × .01 = 5256 minutes = 87.6 hours downtime/year	525,600 × .005 = 2628 minutes = 43.8 hours downtime/year	525,600 × .0001 = 52.56 minutes downtime/year

the network. Given frame relay's ISDN roots, link management is provided out-of-band, thus a separate virtual circuit is used for transmitting link status information messages.

29. OK. Thanks for the information. Let's change gears and get back to some of the other aspects of frame relay. For example, you mentioned voice over frame relay. What is that all about?

Frame relay has evolved from a data only service to one that can support voice, fax, and video transmissions. *Voice over frame relay* (VOFR) has benefited from the development of voice-capable FRADs, which have been designed with advanced technologies to accommodate the nuances associated with transmitting voice traffic. These technologies include voice compression, echo cancellation, and delay control techniques. Voice compression eliminates pauses and redundant information typical of human communication. This reduces the amount of bandwidth required to transmit voice signals. Voice compression also permits voice transmissions over lower-speed channels. Voice compression methods include two international standards, ITU G.729 and ITU G.728, as well as proprietary solutions. Echo cancellation eliminates voice echoing, which occurs when propagation delays reflect voice traffic back to its point of transmission. In addition to echoing, delays can also cause voice distortion. For example, if jitter is high, then the receiving-end equipment will not be able to satisfactorily regenerate the voice signal. Delay control techniques that have been developed to eliminate problems related to delay include traffic prioritization and fragmentation. With prioritization, voice transmissions are assigned a higher priority than data frames, which are buffered until voice traffic has been transmitted. Fragmentation minimizes delay by segmenting large data frames into smaller-sized frames so that voice transmissions are not impeded by the transmission of large data frames. In a typical VOFR application, a customer's PBX is connected to a FRAD, which is connected to the data network. In most cases, additional PVCs are needed, and CIRs will have to be increased to accommodate voice traffic. Frame relay service provider switches also must be capable of transmitting voice traffic. A typical implementation is shown in Figure 12.10. In addition to voice, several vendors have also produced FRADs

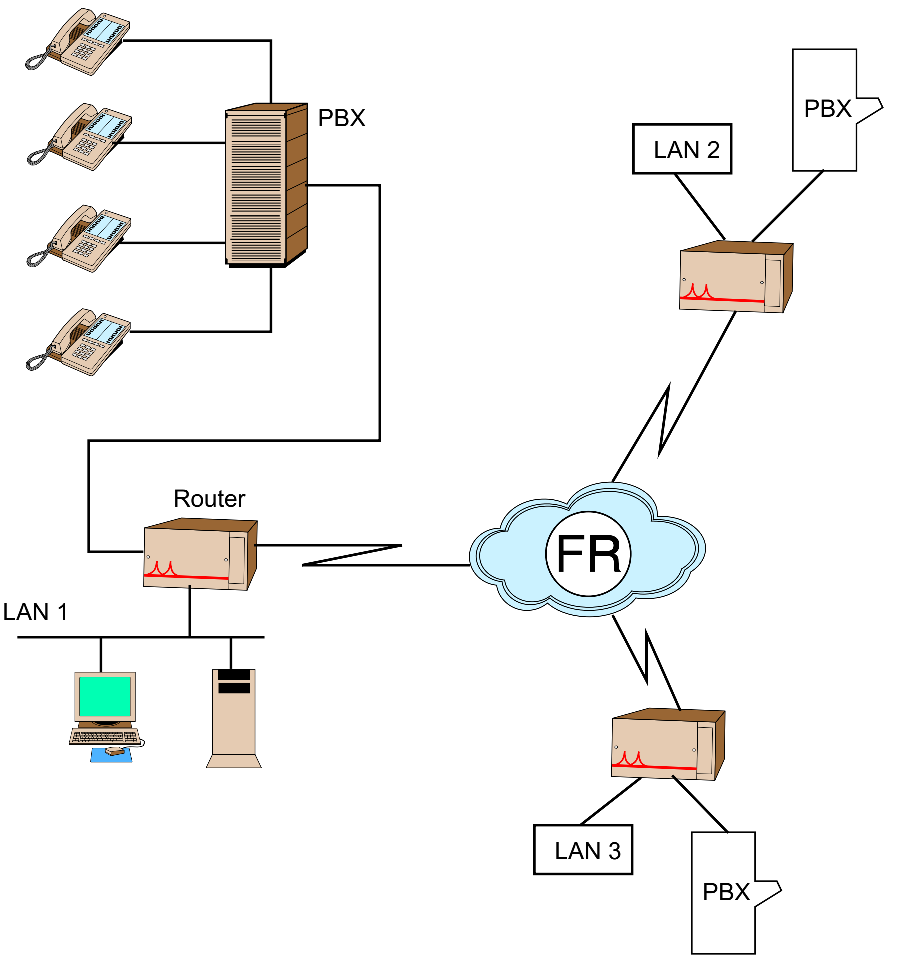


FIGURE 12.10 A typical implementation of *voice over frame relay* (VOFR) involves a PBX system connected to a voice-capable FRAD, which also links a LAN.

that are capable of supporting real-time video. Once again, attention needs to be paid to adequate bandwidth and CIRs. (See Chapter 17 for additional information about convergence technologies.)

30. How does frame relay compare to SMDS or ATM?

During the late 1980s to early to mid-1990s, frame relay, SMDS, ATM, and broadband ISDN (B-ISDN) were all in the spotlight as new fast packet-switching technologies or services. Frame relay, which originally was a subset of ISDN, was defined initially to transmit data at rates ranging from 56 kbps to T1/E-1. It was considered an interim technology, lying somewhere between X.25 and ATM relative to their respective data transmission rates. Frame relay data rates dovetailed nicely with the local loop—the link between a customer's site and the telco's nearest point of presence (POP). SMDS and B-ISDN were initially designed to transmit data, fax, graphics, video, and voice at speeds from 45 Mbps to 155 Mbps. ATM was designed initially to offer the same kinds of services as SMDS and B-ISDN, but its transmission rates were from 45 Mbps to 2.4 Gbps.

Much has changed in such a short period of time. In the United States, frame relay has essentially pushed SMDS and B-ISDN out of the spotlight (see Chapters 11 and 13). Frame relay now operates at T3, and telcos are upgrading their networks to support rates greater than DS-3. In fact, frame relay's evolution took another giant step forward when the ITU approved a new version of frame relay recommendations in 2000. The new recommendations—Recommendation X.36, which defines the user–network interface, and Recommendation X.76, which defines the network–network interface—are applicable to data rates up to OC-12 (622 Mbps), making frame relay competitive with ATM for WAN connections. Additionally, frame relay is now capable of supporting services such as on-demand connections, multicast communications, data compression, and CoS/QoS, all of which enable it to transmit multimedia data such as voice and video. This makes frame relay a viable option for LAN-to-LAN and WAN connections, as well as a broadband service. Furthermore, frame relay frames can be encapsulated within ATM's 53-byte cell and transmitted across ATM links. Perhaps more important, though, given frame relay's variable frame size, a single frame can carry a minimum-sized Ethernet/802.3 frame (64 bytes), whereas ATM needs two cells for the same transmission. Thus, when it comes to transmitting Ethernet frames, frame relay is more efficient than ATM because it uses less bandwidth. Collectively, all of these issues and developments ensure that frame relay will either compete or coexist with ATM.

31. Speaking of Ethernet, how will frame relay stack up against the new 10 Gigabit Ethernet standard?

This is a good question. In the near future, we do not envision 10 Gigabit Ethernet (see Chapter 8) pushing aside established frame relay networks. However, frame relay might follow a similar path to that of FDDI. That is, as 10 Gigabit Ethernet's presence in the MAN/WAN environments becomes more pronounced, we probably will not see any new frame relay networks being established. It also might serve as a cost-effective backup network to Gigabit Ethernet MANs/WANs similar to the role ISDN serves to frame relay (see Chapter 11 and below).

32. One last question. Wasn't that AT&T network that crashed back in April 1998 a frame relay network? What happened?

Yes it was. In fact, in the recent past, two frame relay outages made the news. The first occurred on April 13, 1998, when AT&T's frame relay network crashed as a result of a firmware upgrade to a core switch. According to a report in the June 1998 issue of *Data Communications*, a technician performed the upgrade on a live switch that was connected to the network and then entered incorrect commands. This generated spurious administrative messages, which would not have been a problem if the switch was off-line. However, because the switch was on-line, these messages propagated to other switches throughout the network, and within a half-hour, the entire network was down. This left AT&T's frame relay customers without connectivity for more than a day. Note that if a customer's service uptime was guaranteed at either 99 percent to 99.5 percent and based on a yearly rate, the guarantee would not have applied (see Table 12.1). The second outage was on August 5, 1999, and involved MCI WorldCom's frame relay network. In this incident, only about 15 percent of the network infrastructure was affected, but this still translated to approximately 30 percent of MCI WorldCom's customers experiencing network failure. Some customers' connectivity was disrupted for as long as 10 days. The MCI WorldCom outage was the result of a software error.

Although frame relay is regarded as a highly reliable and cost-effective network, the AT&T and MCI WorldCom crashes demonstrated that as little as a single hardware or software error can disrupt frame relay connections. Consequently, it is prudent for frame relay customers (and all users of network services) to give serious attention to the issue of disaster recovery. For example, AT&T and MCI WorldCom customers who had secondary (i.e., backup) ISDN circuits in place, or redundant frame relay circuits from a different provider, were able to maintain connectivity. Given the additional cost of these measures, one might be tempted to ask, "Why bother?" The answer to this question can be given by paraphrasing the edict used to explain why George Bush (the father) lost his bid for a second term as president of the United States in 1992: "It's technology, stupid."

END-OF-CHAPTER COMMENTARY

In this chapter several concepts and technologies related to frame relay were discussed, including circuit- and packet-switching, virtual circuits (PVCs and SVCs), bandwidth (CIRs), design and topology issues, and ISDN. Additional information about these concepts can be found in earlier chapters within this book. For example: circuit- and packet-switching are discussed in Chapter 1; network design and topology issues, as well as virtual circuits, are discussed in Chapter 2; bandwidth is discussed in Chapters 2 and 4; and ISDN is discussed in Chapter 11. Other technologies and services that were also discussed are addressed in subsequent chapters. These are SMDS, which is the topic of the next chapter (see Chapter 13), and ATM (see Chapter 14), which follows our discussion of SMDS.

This Page Intentionally Left Blank

Chapter 13

Switched Multimegabit Data Service (SMDS)

In this chapter, we present information about switched multimegabit data service (SMDS), a data service that never achieved popularity in the United States. Our main purpose in discussing SMDS is twofold. First, the service is interesting in its own right and deserves attention. Perhaps more important, though, is SMDS's technical design and specifications. SMDS is based on IEEE 802.6, a data link layer protocol for metropolitan area networks (MANs). Called *Distributed Queue Dual Bus* (DQDB), this protocol is quite different from other protocols we have considered and merits examination. An outline of the main topics we address follows:

- SMDS Overview (Questions 1–2, 6)
- SMDS and Local and Interexchange Carriers (Questions 3–5)
- The DQDB and SMDS Protocols (Questions 7–9)
- SMDS Addressing (Questions 10–11)
- SMDS Applications (Question 12)
- SMDS vs. Frame Relay and ATM (Questions 13–14)
- Current Status and Future of SMDS (Question 15)

1. What is SMDS?

Switched multimegabit data service is a cell-based, connectionless, high-speed, public, packet-switched, broadband, metropolitan area data network service designed for LAN-to-LAN connectivity. It is also a data service, which means that it can only transmit data (although it is capable of transmitting voice and video).

2. What do you mean by "cell-based?"

SMDS uses fixed-length *cells* as its basic unit for transmitting data. These cells are similar to ATM cells, namely, they contain 53 bytes—a 44-byte payload plus a 7-byte header and a 2-byte trailer. (See Chapter 14 for more information about ATM.) As a connectionless data service, SMDS does not establish a connection between sending and receiving hosts prior to data transmission. Instead, an SMDS switch establishes a virtual circuit between sending and receiving machines. Cells are placed on the medium when they are ready for transmission and then transmitted independently of each other. Thus, data cells are transmitted without delay and in no particular order. This provides bandwidth on demand for the kind of bursty traffic inherent in LAN applications. (In contrast, ATM and frame relay are connection-oriented.) SMDS is available from telco providers as a shared, public network that uses an international standards-based addressing plan known as E.164. Operating like a shared LAN (e.g., Ethernet/802.3), each SMDS cell contains a destination address enabling any SMDS subscriber to exchange data with any other SMDS subscriber—only those nodes with the correct destination address respond to the transmission.

SMDS can support several data rates including DS-1, DS-3, and SONET OC-3. Low-speed SMDS (56 kbps/64 kbps) is also available from some telcos. Since SMDS is a technology-independent service, customers can subscribe to SMDS at a specific data rate for guaranteed throughput. The broadband aspect of SMDS comes from its compatibility with broadband ISDN (B-ISDN) and its potential for supporting voice and video. (See Chapter 11 for additional information about B-ISDN.) SMDS also is based on and compatible with the IEEE 802.6 metropolitan area network (MAN) standard.

3. Who developed SMDS?

SMDS was developed by Bellcore and the Regional Bell Operating Companies (RBOCs) to address customer demand for interconnected multimegabit LANs within a major metropolitan area. It was designed to be a high performance, reasonably priced, switched service to provide connectivity and extend LAN-like performance. SMDS is implemented by local exchange carriers (LECs), which are companies that provide local telephone and telecommunications services within the same local access and transport area (LATA).

4. Hold the phone. What's a LATA?

The concept of a LATA was introduced in 1984 when the U.S. Justice Department broke up AT&T. An area of concern that emerged from the breakup was the issue of revenue from the long distance calling market. AT&T did not want the LECs to have control over both the local and long distance calling markets. This problem was resolved by establishing 195 *local access and transport areas* (LATAs), which comprised specific geographical and administrative areas controlled by the LECs. Federal regulations restricted LECs to providing only local telephone and telecommunications services within the same LATA. Calls that cross a LATA were handled by *interexchange carriers* IECs or IXCs), which provided long-distance telephone and telecommunications services (e.g., AT&T, Sprint, and MCI WorldCom). In other words, an LEC provided intra-LATA services and an IEC provided inter-LATA services. Thus, voice calls originating in one LATA and destined for another LATA were initially handled by the first LATA's LEC, which then handed the call (i.e., circuit) over to an IEC for inter-LATA transport. The IEC, in turn, handed the call over to the LEC that had control over the destination LATA. Tariffs related to long-distance calls were based on which networks the calls were carried (the LEC's or the IEC's). Thus, it was not uncommon for a long-distance call from New York City to Los Angeles to be less expensive than one placed between two cities within the same state and only 50 miles from each other. The NY–LA call might be carried exclusively by an IEC, but the intrastate call might have to cross one or more LATA boundaries.

This situation was not restricted to voice calls. Since the telephone network is commonly used for data communications, the LECs and IECs play the same roles for data communications as they do for voice transmissions. As a result, circuit costs associated with data communication networks are a function of LATAs. Circuits established within the same LATA are less expensive than those that have to cross LATA boundaries. The 1984 breakup of AT&T also established clearly defined points of demarcation, called DEMARCs, which separated the customer's premises from the telephone company's network. The DEMARC also specified who owns what equipment—the customer or telephone company—where the equipment is placed, and where tariffs and service rates begin and end. (This is discussed in a little more detail later in the chapter.)

Much of this is changing, though, as a result of the U.S. Telecommunications Act of 1996, which permits LECs to provide long distance service and IECs to provide local calling service. This change effectively blurs the distinction between an LEC and IEC. The Telecommunications Act also fostered the creation of other telecommunications companies to spur competition for telecommunications services. The logic behind this is that competition reduces prices and provides choices for consumers.

Along with the new law came new terminology. The LECs that existed prior to the Telecommunications Act are now referred to as *Incumbent LECs* (ILECs), and the new telecommunications service providers formed as a result of the Telecommunications Act are known as *competitive LECs* (CLECs). Today, ILECs and CLECs offer a wide range of telecommunications services, including local dial tone, long-distance calling, Internet access, and cable TV. The CLECs are also permitted to colocate their equipment with ILECs' equipment and typically lease ILECs' circuits for delivering their services. Some CLECs have developed their own networks independent of ILEC or IEC networks.

5. If SMDS is offered through LECs, what happens if the LANs that are to be connected are in different LATAs?

Good question. Prior to the Telecommunications Act of 1996, this was a problem because LECs were prohibited by law from carrying any type of traffic across LATA boundaries. To address this issue, a service called *Exchange Access SMDS* (XA-SMDS) was deployed in 1993. Through XA-SMDS, LECs offered SMDS service to IECs for delivery across LATAs. Connectivity was usually established via a private link where standard SMDS routers transferred traffic from one network to another. XA-SMDS also specified a standard Inter-Carrier Interface (ICI) for the LEC–IEC connection.

Today, as a result of the Telecommunications Act of 1996, LECs and IECs can compete in each other's business so this is no longer an issue. The only restriction is that LECs are regulated separately by each state's public utilities commissions. Thus, if the LEC providing SMDS service operates in two different states, then the types, cost, and rules of the service might vary considerably. MCI WorldCom also has established a national SMDS network that provides connectivity among SMDS networks located in different parts of the country. In this context, SMDS is considered both a MAN and WAN service.

6. OK. Thanks. Why was SMDS developed?

Prior to SMDS' development in the late 1980s and early 1990s, the only method network administrators had for interconnecting their LANs within a metropolitan area was via leased lines at either 56 kbps or T1 rates. Given the LAN data rates of the time (4 Mbps/16 Mbps for token ring and 10 Mbps for Ethernet/802.3), the leased line approach was insufficient from three perspectives. First, the leased lines represented a bottleneck for LAN-to-LAN transmissions. Data frames being transferred between two Ethernet/802.3 nodes residing on separate LANs interconnected by a 56 kbps or T1 line resulted in congestion and transmission delays. The LAN data rates simply overwhelmed the capacity of the leased circuit. Second, leased line costs were nontrivial. If an organization wanted to interconnect multiple LANs located at different locations across a metropolitan area (e.g., within a local county), it would need multiple leased lines. Depending on the topology used, the cost of multiple T1 service can be exorbitant. Although 56-kbps service would be more affordable, the inherent delays and congestion would make it unacceptable. Finally, private leased lines did not scale up well economically. If congestion and delays of T1 service were unacceptable, the next step up was T3 service, which was too expensive for many organizations. Figures 13.1 and 13.2 compare traditionally designed MANs to an SMDS network. With SMDS, however, an organization can create a corporate LAN internet (i.e., a corporate network of LANs) across a metropolitan area cost-effectively.

7. You said SMDS is a service, not a technology. On what technology is SMDS based?

SMDS is based on a subset of the IEEE 802.6 physical layer and MAC sublayer standard, which specifies a high-speed network protocol similar to token ring. At the physical layer, IEEE 802.6 specifies a dual bus design using fiber-optic cable (Figure 13.3). The buses, which are labeled *A* and *B*, transmit in only one, but opposite, directions. This is similar to FDDI's counterrotating ring (see Chapter 10). Together, the buses provide full-duplex operation. IEEE 802.6 networks can be designed as either an open or looped bus. The difference between the two topologies is where each bus begins and ends. In the open design, the buses begin and end at different nodes; in the loop design, the buses begin and end at the same node. Both designs are illustrated in Figures 13.4(a) and 13.4(b), respectively. SMDS uses the open bus topology.

At the data link layer, access to the SMDS network is governed by the IEEE 802.6 *Distributed Queue Dual Bus (DQDB)* protocol. This protocol partitions each bus into time slots, which are used to transmit data. Each slot contains a busy bit and a request bit. If the slot contains data, then the busy bit is set to 1. A slot with its request bit set implies that a node is requesting to send data. If the busy bit is not set (i.e., busy bit = 0), then the slot is considered empty. Nodes can only transmit data using empty slots.

The DQDB protocol works as follows: Prior to sending data, a node must first reserve slots on one bus to be used on the second bus. For example, in Figure 13.3, if node 1 wants to send data to node 3, it must use bus *A* because *A* transmits in the direction of node 3. This implies that node 1 must reserve a slot on bus *B*. Similarly, if node 3 wants to send data to node 1, it must use bus *B*, which implies it must reserve a slot on bus *A*. As mentioned above, slots are reserved by setting the request bit of an empty slot.

Reserving slots on one bus to be used on the second bus enables nodes to notify their neighbors that they have data to transmit. The process of reserving slots also prevents nodes from monopolizing the bus. Without a reservation system, upstream nodes can continuously fill empty slots with data and hence deny downstream slots from ever seeing an

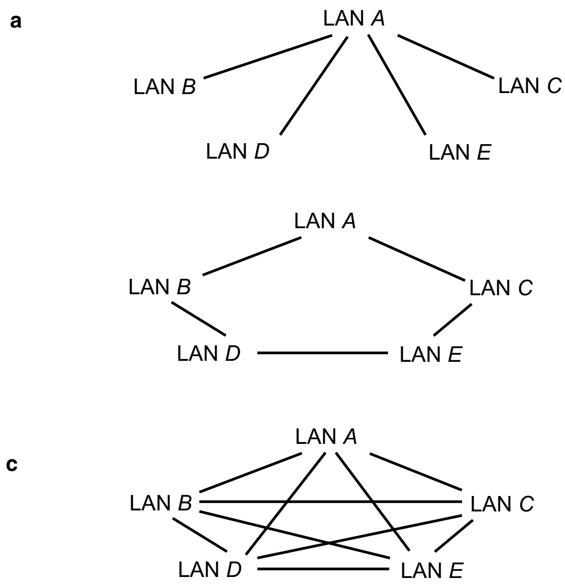


FIGURE 13.1 The traditional method of establishing metropolitan area networks (MANs) is to interconnect LANs within a metropolitan area using dedicated leased circuits. Several methods are possible, and each has its own advantages and disadvantages. In (a), LANs *B, C, D,* and *E* are indirectly connected via LAN *A*. One problem with this design is that if LAN *A* fails, then the remaining LANs cannot communicate with one another. In (b), the single source of failure of design (a) is removed and all neighboring LANs are directly interconnected. However, nonneighboring LANs are still indirectly connected. The design in (c) represents a fully meshed MAN in which every LAN is directly connected to every other LAN. Although more robust, this design is extremely expensive.

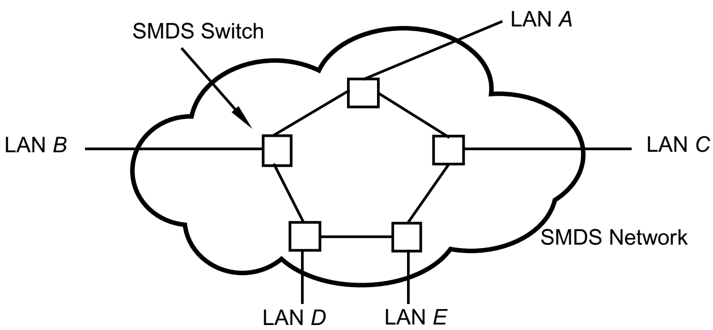


FIGURE 13.2 In contrast to traditionally designed MANs (Figure 13.1), an SMDS network provides direct connectivity among LANs within a geographical region using a single link for each LAN.

empty slot. For example, in Figure 13.3, slots are generated at the head end of bus *A* and flow from left-to-right. Thus, node 1 will always see an empty slot first, followed by node 2, then node 3, and finally node 4. Similarly, on bus *B*, slots are generated and flow from right-to-left. Thus, node 4 will always see an empty slot on this bus before any of the other nodes.

Once a node has requested a slot, it then monitors both buses and maintains a request counter. The purpose of the counter is to maintain an accurate count of the number of unfilled data transmission requests made by downstream nodes. (Unfilled requests are identified whenever a slot has its request bit set.) This counter specifies the number of empty slots a requesting node must let pass before it can access an empty slot itself. Each time a slot that has its request bit set passes by, the counter is increased by 1; each time an empty slot passes on the second bus, the counter is decreased by 1. When the counter reaches 0, the node is permitted to transmit data—that is, it can access the next empty slot on the other bus. This procedure permits nodes to essentially transmit data on a first-come-first-served basis, and enables nodes to know the state of queued requests throughout the entire network. This process also represents a form of CSMA/CA, *carrier sense multiple access with collision avoidance* (see Chapter 5), which prevents nodes from sending data at the same time.

Time slots are sampled by connected nodes at a rate of 8000 per second. This makes the timing of an SMDS network consistent with the T1/E-1 and SONET/SDH hierarchies (see Chapter 7). The number of slots on a network is also a function of bus speed. For example, if the transmission rate is 34 Mbps, then there are eight slots available per frame. DQDB supports both connectionless and connection-oriented service and is capable of transmitting data, voice and video. As a subset of IEEE 802.6, though, SMDS is connectionless based and transmits only data.

8. OK. So SMDS is based on a subset of IEEE 802.6. Isn't there a specific SMDS protocol, though?

Yes there is. The *SMDS Interface Protocol* (SIP) was defined by Bellcore, which is the research branch of the RBOCs. SIP consists of three protocol levels, formally called SIP level 3, SIP level 2, and SIP level 1. As we describe these protocol levels you will note a similarity between their functions and the first three layers of the OSI model. It is important to note that although these protocol levels are based on the first three OSI layers, they do not directly correspond to these layers. Instead, the three protocol levels represent SMDS's MAC sublayer and hence operate at the data link layer. Before we get into any detail about SIP, it might help to first identify the components of an SMDS network. A simplified diagram is provided for this purpose in Figure 13.5. LANs are interconnected via SMDS using three components: SMDS routers, SMDS DSUs (SDSUs), and SMDS switches. Collectively, these three components form the basis of an SMDS network and support the DQDB protocol. Router, DSU, and switch connections are established as in any LAN–WAN configuration, and the major router manufacturers (e.g., Cisco) support SMDS routing. The point of demarcation, called the *subscriber network interface* (SNI), separates the router and SDSU, which are called the *customer premises equipment* (CPE), from the telco's SMDS network equipment. A SMDS switch represents the head end of

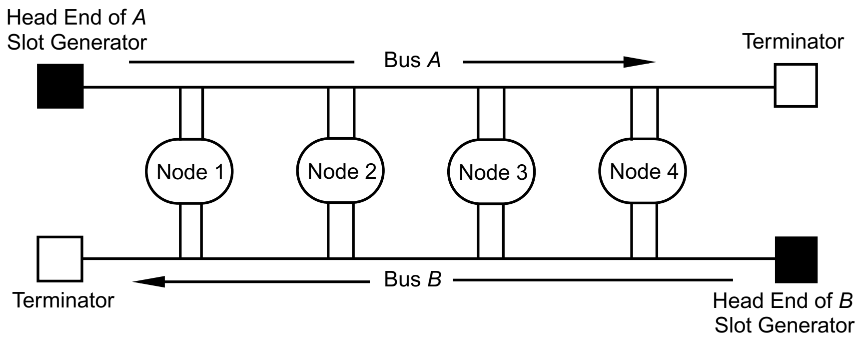


FIGURE 13.3 The physical layer of an SMDS network has two fiber-optic buses, *A* and *B,* which transmit data in opposite directions. The direction of data flow is from the head of the bus, which acts as a slot generator, to the terminator. Nodes are connected to both buses in a logically adjacent manner and read all transmitted data. Source: Adapted from Bates & Gregory, 1998.

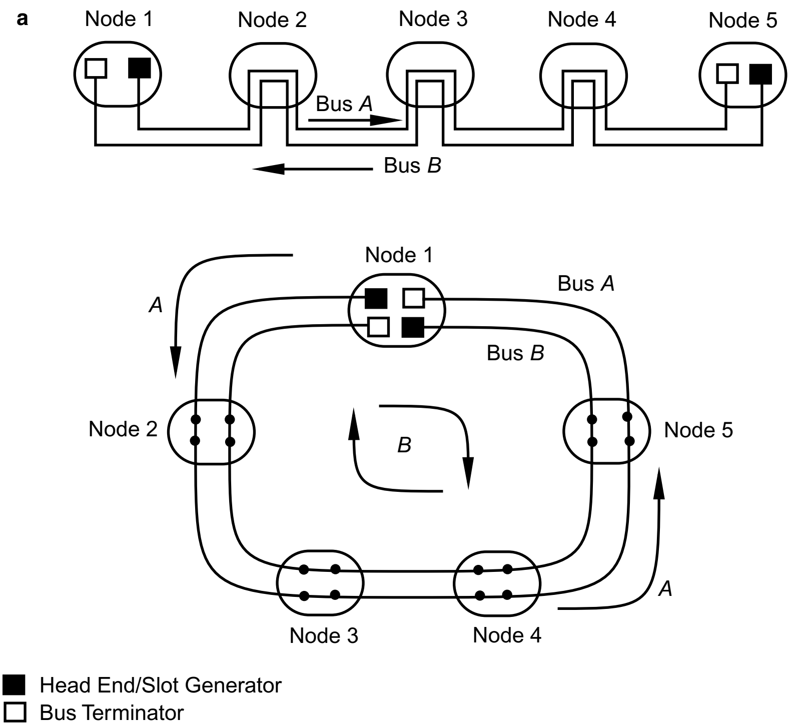


FIGURE 13.4 In an open bus design (a), the head end of the buses is located in two different nodes. In the closed loop design (b), the head of both buses is located in the same node. Source: Adapted from Bates & Gregory, 1998.

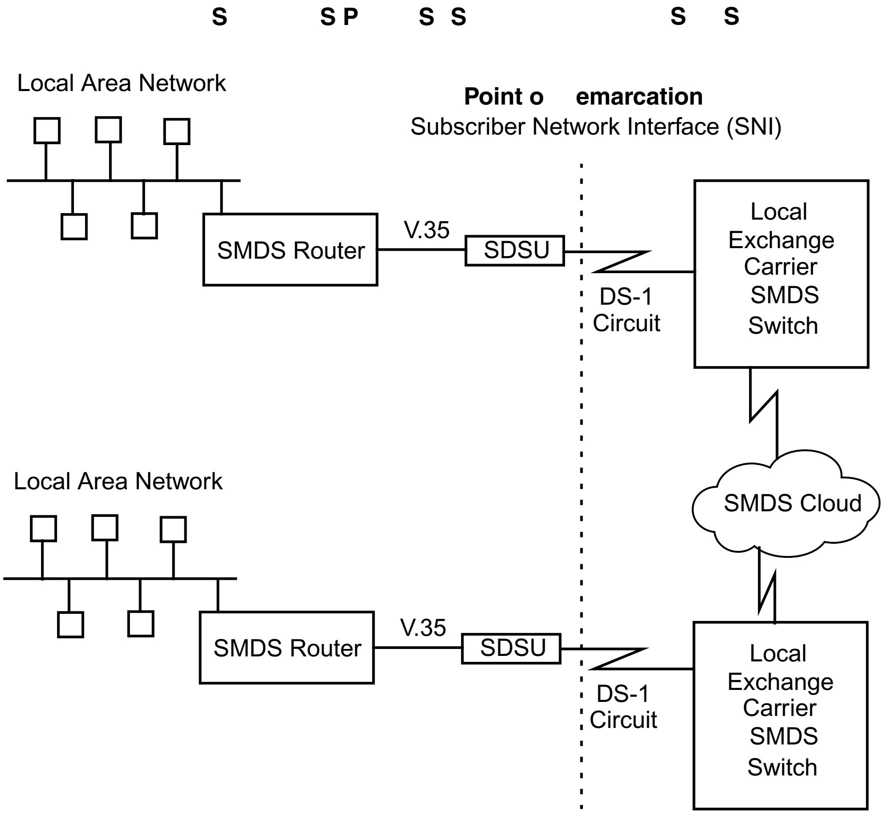


FIGURE 13.5 The components of an SMDS network consist of a router that supports the SMDS Interface Protocol, an SMDS CSU/DSU, a circuit, and an SMDS switch. The point that separates the customer's equipment from the SMDS network is called the point of demarcation and is the interface between the circuit and the SDSU. This is usually a punch-down block located at the customer's telecommunications office. This interface is formally called the subscriber network interface (SNI). Source: Adapted from Sharer, 1993.

one of the buses (e.g., bus *A*), and all other nodes are CPE nodes. Thus, in Figure 13.4(a), node 1 is the SMDS switch, and nodes 2 through 5 are CPE nodes. The end CPE node, which is node 5 in Figure 13.4(a), serves as the head end of the second bus (i.e., bus *B*).

When data frames destined for a remote LAN are received by the SMDS router from the local LAN, the router encapsulates these frames into special SMDS frames via SIP level 3. Called SIP level-3 PDUs (short for *protocol data units*), these frames are then transmitted across the V.35 interface between the router and SDSU. SIP level-3 PDUs provide the connectionless service of SMDS and contain up to 9188 bytes of user information, plus related header and trailer information (Figure 13.6). At the SDSU, SIP level 2 accepts the Level 3 PDUs and partitions them into fixed-length segments of 53 bytes each (Figure 13.7). The various length and tag fields contained in the level-3 PDUs provide the

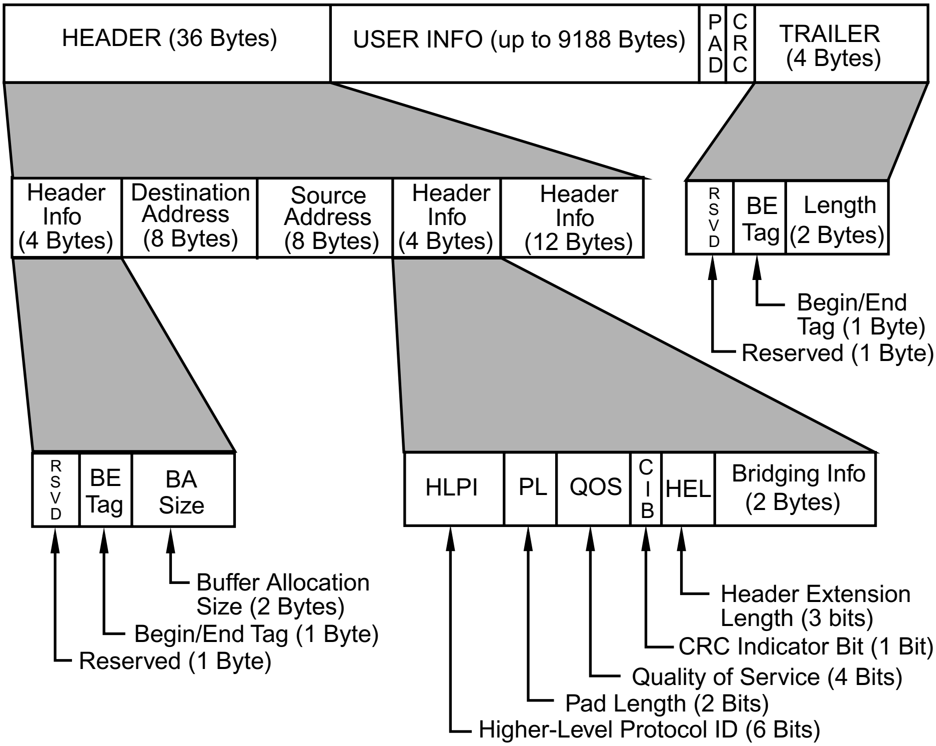


FIGURE 13.6 Contents of an SMDS Interface Protocol level-3 PDU. This frame is partitioned into a 53-byte fixed-length cell by SIP level 2 (see also Figures 13.7 and 13.8). The PAD and CRC fields vary from 0 to 3 bytes and 0 to 4 bytes, respectively. Source: Adapted from Bates & Gregory, 1998.

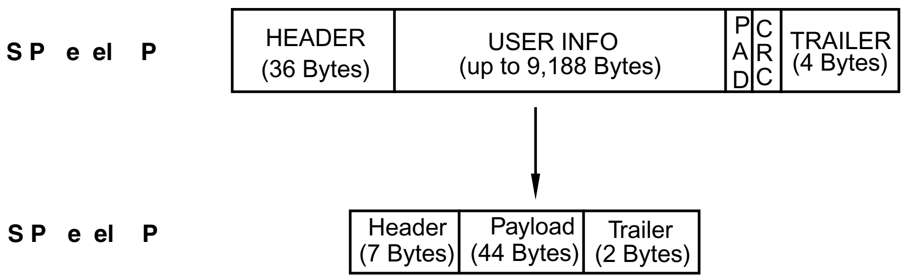


FIGURE 13.7 SIP level-3 PDUs are fragmented into 53-byte cells by SIP level 2. These level-2 PDUs are the slots that are placed on the medium. SMDS nodes read all slots placed on the bus and write to empty slots. The number of slots placed on the bus is a function of the bus's data rate. The expanded version of an SIP level-2 PDU is shown in Figure 13.8.

necessary information for this partitioning (and eventual reassembly). These 53-byte cells, called SIP level-2 PDUs, consist of 44 bytes of user information (called the payload or segmentation unit), a 7-byte header, and a 2-byte trailer (Figure 13.8). SIP level-2 PDUs represent the basic data unit on an SMDS network. These data units are then processed by SIP level 1, which provides the physical interface to the SMDS network. This last level of SIP consists of two sublayers: the *transmission system* and the *physical layer convergence protocol*. The transmission system specifies how cells are to be placed on the medium; the physical layer convergence protocol formats the 53-byte cells for actual delivery across the network. The cells are then reassembled into level-3 PDUs at the receiving end. Thus, an SMDS router provides SIP-3, and the SMDS DSU provides SIP-2 and SIP-1.

When SMDS was initially deployed, the LECs required their customers' SMDS interface equipment to deliver level-2 PDUs to the LECs' SMDS switches. This meant that customers could not use standard CSU/DSUs; they had to purchase special SMDS DSUs. This made SMDS an expensive service because SDSUs, in some cases, cost three times more than conventional CSU/DSUs. Today, however, the level-3 to level-2 conversion is performed within the LEC's SMDS switch, enabling subscribers to use conventional CSU/DSUs.

9. I see in Figure 13.8 that the Access Control field of the header contains the request and busy bits. I also notice that there are three different request bits. What's that all about?

SMDS supports three request bits: Req0, Req1, and Req2. Each one corresponds to a different level of priority. Priority levels are similar to that of token ring. Regardless of which priority is used, request bits indicate that a node has data to transmit.

10. What do SMDS addresses look like?

SMDS addresses are based on the standard ISDN global numbering addressing format specified by ITU-T standard E.164. These addresses are just like telephone numbers. They are 15 decimal digits long and include a country code, area or city code, and a local number. Country codes are two or three digits long and consist of a zone code followed by a one- or two-digit national identifier. Area or city codes are up to four digits long. For example, country codes for the United Kingdom (UK), Australia, and Taiwan are 44, 61, and 886, respectively. Sample city codes within the UK include 171 and 181 for London, 141 for Glasgow, 151 for Liverpool, and 1232 for Belfast. Sample city codes in Taiwan include 2 for Taipei and 37 for Chunan. In the United States and Canada, country codes are not used. Instead, these countries use the zone code 1, which is followed by a three-digit area code and a seven digit local number.

There are also two types of SMDS addresses. *Individual addresses* are used for unicast transmissions; *group addresses* are used for multicast traffic. To distinguish between the two, individual addresses begin with a hexadecimal C and group addresses begin with a hexadecimal E. If an address contains less than 15 digits, then it is padded with hexadecimal Fs. Thus, the SMDS address, C14075557235FFFF, identifies an individual location in the 407 area code within the United States, and E160462284422961 is a group address in British Columbia, Canada. The SMDS service provider (i.e., the LEC) assigns a block of up to 16 individual addresses to each subscriber network interface (SNI), which can

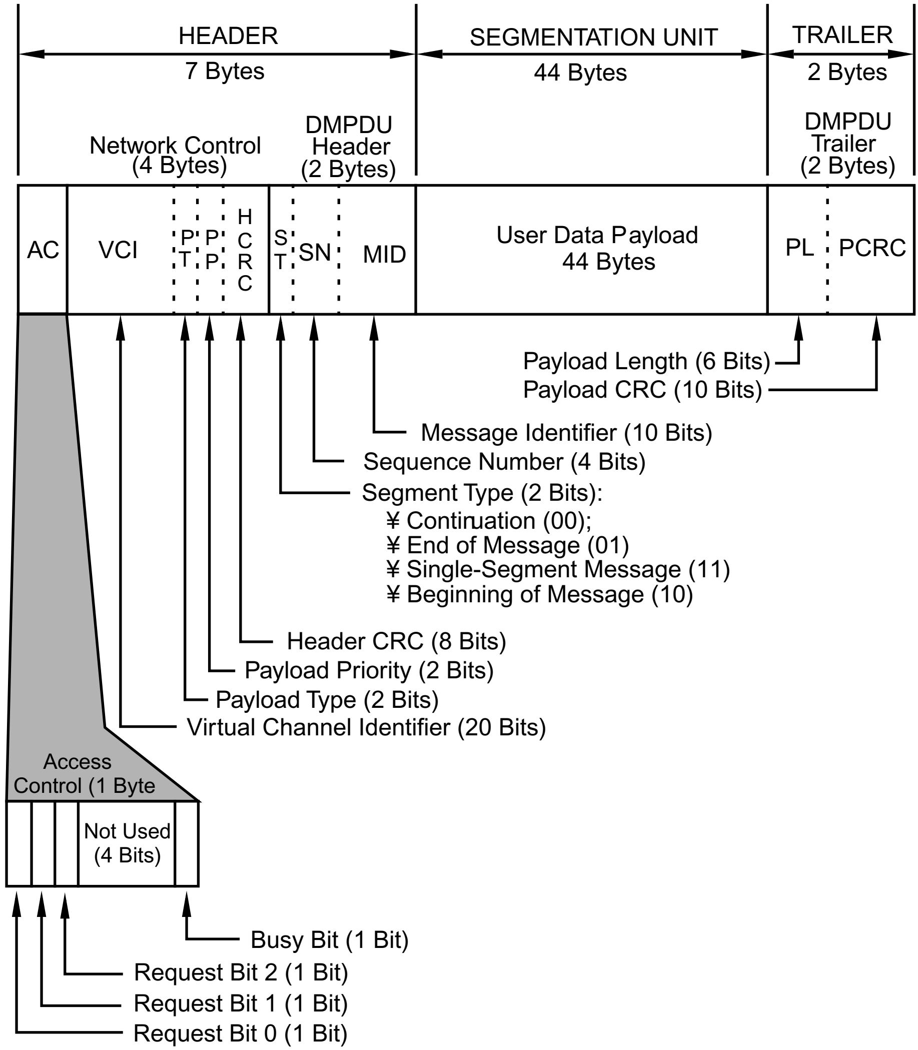


FIGURE 13.8 The basic data unit on an SMDS network is a 53-byte cell. This cell, which consists of a 7-byte header, a 44-byte payload, and a 2-byte trailer, is the result of a fragmentation process that is performed by the SMDS Interface Protocol level 2 (see Figure 14.7). SIP level 2 receives from a router (via SIP level 3) an initial MAC frame that contains up to 9188 bytes of user data plus related header and trailer information. This frame, called the *initial MAC protocol data unit* (IMPDU), is then partitioned into 53-byte cells for transmission over the network via SIP level 1. At the receiving end, cells are then reassembled into their original form. Key to this reassembly process is the header and trailer information of the IMPDU. This information is also partitioned as part of the fragmentation process and carried via the 2-byte derived MAC protocol data units (DMPDU), which are included as part of the 53-byte cell (see also Figure 13.6). Source: Adapted from Bates & Gregory, 1998.

exchange data among 128 individual or group addresses. More than one SMDS node can be assigned the same group address in addition to an individual address. There are restrictions, though. For example, an individual address cannot belong to more than 32 groups, and a group address cannot represent more than 128 individual addresses.

Group addressing also can be used as a filtering mechanism. For example, assume five LANs (*A, B, C, D,* and *E*) within a local county are interconnected via SMDS. Further assume that all five LANs are running IP; *A, B,* and *C* are also running IPX; and *D* and *E* are also running AppleTalk. Using group addressing, IPX traffic can be isolated from LANs *D* and *E*, and AppleTalk traffic can be isolated from LANs *A, B,* and *C*. Group addressing also can be used to create virtual ports over SMDS.

11. Since data transmission in an SMDS network occurs via the subscriber network interface, who configures what addresses get to exchange data, the customer or the LEC?

The service provider does this configuration based on customer specifications. Thus, the customer simply informs the LEC which addresses are to communicate with each other, and at what rate, and the LEC does all the configurations.

12. How do companies know if SMDS is an appropriate service for them?

Companies that can benefit from SMDS include organizations that have at least four remote sites across a metropolitan area that need to be interconnected in a seamless, cost-effective manner; require high-speed, bulk data transfers at a reasonable cost; need to transfer quickly large files including video and high-resolution images such as blueprint schematics, MRIs, CAT scans, and X-rays. In short, SMDS is appropriate for any enterprise with distributed sites that uses bandwidth-intensive applications. Prime candidates include hospitals, publishing companies, graphic design houses, insurance companies, police departments, automobile manufacturers, colleges and universities, and municipal governments. Applications include CAD/CAM, LAN-to-LAN connectivity, telemedicine and teleradiology, collaborative printing and publishing, image and multimedia file transfers, distance education, videoconferencing, and high-speed access to the Internet.

13. How does SMDS compare to frame relay and ATM?

When comparing SMDS to other LAN-to-LAN technologies such as frame relay and ATM, it is important to understand that SMDS is a service, not a technology; frame relay and ATM are technologies. Since it is technology independent, SMDS can operate over frame relay or ATM. Technology independence also implies protocol independence. Thus, SMDS can support any LAN protocol such as token ring and Ethernet/802.3, as well as various network protocols including TCP/IP, OSI, IPX, and AppleTalk. Second, SMDS is a packet-switching, connectionless service; frame relay and ATM are connection-oriented. This means that it is not necessary to establish a connection through the network prior to data transmission. Instead, packets are transmitted without any delay for setting up or tearing down a circuit. There also is no need to define private virtual circuits (PVCs) or committed interface rates (CIRs), as is the case with frame relay. Thus, an SMDS network is simpler to design than frame relay. Third, SMDS's bandwidth range (56 kbps to SONET

TABLE 13.1 SMDS Features and Related Benefits

Feature	Related Benefit
1. Service, not a technology	**1.** Can operate over frame relay or ATM
2. Protocol-independent	**2.** Supports multiple LAN or network protocols
3. 56/64 kbps to SONET speeds	**3.** Complete range of speeds for all applications
4. Connectionless	**4**. Obviates need to define PVCs as with frame relay
5. Packet-switched	**5**. Packets transmitted without delay
6. Logical, fully meshed connectivity	**6.** Reliable and robust LAN-to-LAN connectivity
7. Shared public network	**7.** Subscribers can exchange data with each other
8. Uses 53-byte cells similar to ATM	**8**. Smooth migration to ATM
9. No CIRs as with frame relay	**9.** Guaranteed bandwidth; pay only for usage
10. Built-in management	**10.** Provides usage-based billing and statistics

TABLE 13.2 SMDS Disadvantages

1. Restricted availability—the number of LECs offering SMDS is very limited.
2. Limited nationwide/global service—MCI WorldCom is the only IEC to offer inter-LATA SMDS.
3. Perceived as an expensive service.
4. Overshadowed by frame relay and ATM.
5. Although capable, SMDS does not provide voice or video support.
6. Need for private networks via public backbone is provided by the Internet and VPNs.

rates) provides a better range at the low end than ATM and a better range at the high end than frame relay. Furthermore, unlike frame relay, an SMDS circuit is fully committed to its specified rate, and you only pay for the bandwidth used. Fourth, SMDS provides several network management features including usage-based billing and users' usage statistics. SMDS's addressing scheme also provides a type of built-in security measure by restricting data transfers to nodes assigned a specific group address. Many of these features are not available from SMDS's competition. Table 13.1 summarizes many of the features and benefits of SMDS; Table 13.2 lists some of SMDS's disadvantages.

14. Am I missing something? If SMDS has all of these features and benefits, how come it hasn't received the kind of attention frame relay and ATM receive?

You are right in your observation that SMDS seems to have been overshadowed by frame relay and ATM. However, this is only true in the United States. In Europe, SMDS is popular. (More on this later.) Many reasons have been cited in the trade publications explaining why SMDS got off to a slow start in the United States and then never did materialize into a universally compelling and accepted service. We will summarize them here for you.

First, as is the case in trying to sell any new technology or service, success depends on proper prior planning and marketing. It has been speculated that the LECs failed in both of these areas when it came to SMDS. Poor planning on the LECs' part resulted in expensive equipment, which gave SMDS the reputation of being an expensive service. Contributing to this reputation was SMDS's initial lack of support for low data rates (less than DS-1). Many organizations that could have benefited from SMDS service were locked out because they could not afford (or justify) the high cost associated with high speeds and the potential wasted bandwidth. Although some LECs now offer low-speed SMDS, which makes the service more attractive to companies with limited resources and bandwidth requirements, it's a matter of too little too late since these companies are opting instead for frame relay.

From a technology perspective, insufficient attention also was given to how neighboring LECs would interconnect their SMDS networks. This led to incompatible methods that made customers with LANs that crossed LATA boundaries hesitant to subscribe to SMDS. Although this issue also has been addressed, it is once again a matter of too little too late. Since the SMDS market never materialized, commitment to SMDS among the LECs has been spotty at best. LECs have been opting instead to concentrate on frame relay and ATM. For example, in the United States, among the various LECs, Verizon Communications (formerly Bell Atlantic) offers SMDS service. Furthermore, among the major IXCs (AT&T, MCI WorldCom, and Sprint), only MCI WorldCom offers nationwide SMDS service. This lack of support makes customers wary of the service and less likely to subscribe to it. In fact, the final nail in SMDS's coffin was AT&T's decision to provide frame relay and ATM instead of SMDS.

Another reason for SMDS's lack of market share in the broadband community is it was never designed to support isochronous data, which is needed to transmit real time digital voice and video applications. SMDS was to eventually support voice and video in addition to data. Its DQDB access method provides the necessary technology for this support. However, it doesn't appear that SMDS will be further developed to incorporate support for isochronous services. Finally, although SMDS has built-in security that enables it to use a shared, public network as the backbone for a private network, this concept has been overshadowed by the Internet and virtual private networks (VPNs), as we discussed in Chapter 7.

15. What is the current status and future of SMDS?

From all accounts, SMDS is probably "dead" in the United States. Consider, for example, the following:

1. The SMDS Interest Group, which was the biggest promoter of SMDS, folded in 1997.
2. In *Data Communications'* 25th anniversary issue (October 21, 1997), SMDS was identified as one of the top 25 flops. "Switched multimegabit data services were designed to deliver LAN-like performances over the wide area. And it's just what they did, to the dismay of users who got high bandwidth, lots of flexibility—and the variable delays associated with Ethernet. Now the SMDS Interest Group has folded, presumably for lack of interest" (p. 143).
3. MCI WorldCom is the only interexchange carrier that offers SMDS.

4. Some local exchange carriers have discontinued offering SMDS service. US West, for example, cited a dearth of customer demand as its reason for canceling SMDS.

On the other hand, since it is a service, SMDS can coexist with ATM and frame relay. That is, SMDS subscribers can migrate to ATM or use ATM or frame relay as the underlying technology for their SMDS service. It is uncertain, though, whether this will ever come to fruition on a large-scale basis in the United States. From all accounts it is unlikely that SMDS will grow in market share or be developed further. It might, however, survive as a niche market.

Although SMDS is not in widespread use in the United States, it is still popular in Europe. Providers include British Telecommunications (London), Telecom Eireann (Irish Republic), France Telecom (Paris), Belgacom (Brussels), and Deutsche Telekom (Bonn). Other countries in which SMDS deployment is high include Denmark, Switzerland, Austria, Italy, Sweden, and Australia.

END-OF-CHAPTER COMMENTARY

This concludes our brief discussion on SMDS. For additional information about this service, you are encouraged to consult the references listed in the appendix. You also might want to review Chapter 12 (Frame Relay) and Chapter 14 (ATM) to compare SMDS to these two technologies.

This Page Intentionally Left Blank

Chapter 14

Asynchronous Transfer Mode (ATM)

In this chapter, we provide an introduction to asynchronous transfer mode (ATM), which is a high-speed switching network architecture created in the late 1980s/early 1990s. ATM was designed expressly to combine the delivery of a wide range of services (data, voice, and video) over a single network. We briefly examined ATM from various perspectives in nearly every chapter of this book. It is now time to present a more detailed discussion of this technology. An outline of the major topics we discuss follows:

- Definition and History of ATM (Questions 1–3)
- General Concepts and Operation of ATM (Questions 4–7)
- ATM Interfaces and the Anchorage Accord (Questions 8–10)
- ATM Components and Addressing (Questions 11–13, 22)
- ATM Cells, Switches, and Virtual Connections (Questions 14–22)
- ATM Adaptation Layer (AAL) (Questions 23–26)
- Data Types Insights (Questions 27–29)
- ATM vs. Gigabit Ethernet (Questions 30–32)
- ATM in LAN Environments (Questions 33–34)
- ATM, Frame Relay, and SONET (Questions 35–36)

1. What is ATM?

Asynchronous transfer mode (ATM) is a sophisticated, multispeed network environment that provides a variety of complex network services for applications requiring various types of network solutions. It can be used to carry data, voice and video—separately or simultaneously—over the same network path, and is one of the most complex communications technologies available today for public or private network infrastructures. ATM can be used in LANs, MANs, and WANs, all at the same time if needed. Using terminology developed in previous chapters, ATM might also be considered a "hyphenated" protocol—it is connection-oriented, full-duplex, point-to-point, and cell-switched.

2. I know about connection-oriented, full-duplex, and point-to-point. What is "cell-switched"?

The concept of cell-switched is very similar to frame relay (Chapter 12), which uses switches to transfer variable-length frames within the frame relay cloud from source to destination. Instead of using frames, ATM uses fixed-length *cells*, which contain exactly 53 bytes—48 bytes for user data and 5 bytes for overhead. As a result, ATM is sometimes referred to as *cell relay*, which dates back to the late 1960s. The concept of cell relay is predicated on time domain multiplexing and packet-switching. The term *asynchronous TDM* (ATDM) was coined in 1968. We will discuss ATM cells in more detail later in the chapter.

3. How did ATM get started?

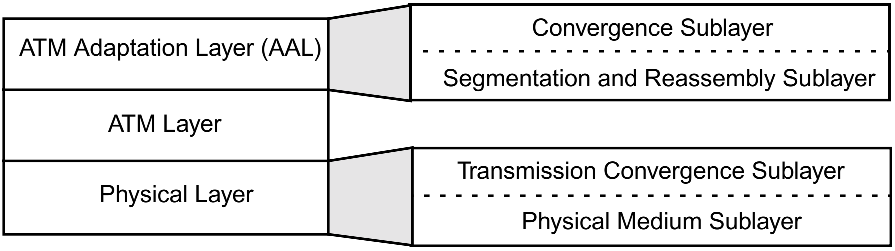
ATM has its roots in broadband ISDN (B-ISDN), which some people refer to as second-generation ISDN. In the presence of B-ISDN, "standard" or "first-generation" ISDN is called *narrowband ISDN* (see Chapter 11). In 1986, the standards organization, CCITT (now part of ITU), decided to make cell relay the transfer mode of B-ISDN. The CCITT also decided to change the name of this cell-based technology from ATDM to ATM. CCITT's decision meant that broadband networks throughout the world would be based on ATM. Two years later, in 1988, a three-layered referenced model for ATM was defined by CCITT. These layers, which represent the first three layers of the B-ISDN reference model, include the physical layer, the ATM layer, and the ATM adaptation layer. It is important to note that the ATM layers do not necessarily correspond to the layers of the OSI model. The ATM layers are summarized in Figure 14.1. (Much of the information presented in Figure 14.1 is discussed later in this chapter.) After much discussion, the CCITT also defined ATM's cell format to consist of 53 bytes, with 48 bytes for user data and 5 bytes for overhead. Finally, in 1990, the CCITT issued its first set of recommendations that specified the details of ATM for B-ISDN. For a good overview of ATM's history, see Gould (1994).

4. I keep hearing about how complex ATM is. Could you provide me a conceptual description of ATM so I have a better understanding about the technology?

Sure. ATM operates as a "network within a network" concept. It has its own internal handshaking and management protocols, quality of service (QoS) facilities, performance and flow control facilities, and many other components that are usually separated on network technologies such as frame relay. ATM is very much like building a subway system that can support other transportation systems on top of it. The "upper" transportation systems do not know that ATM is acting as an independent freeway and are oblivious of the complex nature of ATM networking.

5. That's a start. Tell me more about this "network in a network" concept.

OK. One of ATM's features is that it can guarantee delivery of time-sensitive information over the designed network. For instance, typical video as seen on a television set transmits 30 frames of video per second over the airwaves. Not 29, not 31—30 only. That's it. No more, no less. Therefore, it is critical for the network to be able to move



P ical a er

¥Transports cells from one interface to another via a communications channel.
¥ Supports both optical and electrical communications channels.
¥ LAN support ranges from 25 to 155 Mbps for copper and ber.
¥WAN support includes SONET/SDH rates over ber.
¥ Contains two sublayers: The *physical medium sublayer* function is restricted to only medium-dependent functions such as bit transfer, bit alignment, and electrical-to-optical conversions. The *transmission convergence sublayer* performs functions related to converting cells from the ATM layer into bits on the sending node and converting bits into cells on the receiving node.

A a er

¥ Performs cell multiplexing/demultiplexing and switching.
¥ Provides virtual connections between endpoints.
¥ Generates appropriate cell headers on a sending node based on information received from higher layers; extracts cell header on a receiving node and passes cell payload to higher layers.

A A aptation a er AA

¥ Partitions higher-level user data into 48-byte cells plus necessary overhead.
¥ De nes ve different adaptation types to support different service classes:
— ***AAL1:*** Supports Class A services, which are connection-oriented constant-bit rate (CBR) (e.g., voice transmissions).
— ***AAL2:*** Supports Class B services, which are connection-oriented variable-bit rate (VBR) (e.g., synchronized data, packet-based video).
— ***AAL3/AAL4:*** Supports Class C and Class D services, which are, respectively, connection-oriented VBR (e.g., bursty data used for le transfers) and connectionless VBR (e.g., LAN data).
— ***AAL5:*** Supports Class C and Class D services; known as the *simple and efficient adaptation layer* (SEAL) and used for bursty data transfers in which higher-layer protocols perform error control. AAL5 is a modi cation of connection-oriented VBR.
¥ Consists of two sublayers. The *convergence sublayer* (CS) provides a speci c AAL service at an AAL network service access point (NSAP). The *segmentation and reassembly sublayer* (SAR) segments higher-level messages into an ATM cell 48-byte information eld on the sending node and reassembles this information for delivery to the higher layers on the receiving node.

FIGURE 14.1 The ATM layers. The physical layer corresponds to OSI's physical layer; the ATM layer generally corresponds to OSI's data link and network layers; the AAL generally corresponds to OSI's higher-level layers (transport, session, and application). Source: Adapted from Atkins & Norris, 1995; and Bates & Gregory, 1998.

exactly 30 frames of video information per second over the network. This means that the network must be able to properly reserve bandwidth space from source to destination to accommodate delivery of the 30 frames per second without degrading the throughput between the source and destination locations. To do this, all intermediary locations (typically ATM switch devices) between the source and the destination must allocate network "space" to move the information efficiently in the required time frame. (This is called *isochronous* communications.) This requires various handshaking between systems, bandwidth allocation and management facilities, QoS delivery facilities, and a host of other technologies, algorithms, and techniques to ensure the path between source and destination is clean, fast, and efficient. All this adds up to the need to have a "network in a network" so that ATM devices can properly manage the path of activity between the source and destination systems on the network.

6. Does this mean that communication protocols such as TCP/IP and SNA do not know all of this work is taking place in the ATM hardware?

Yes, Grasshopper. You have searched for enlightenment and found a little.

7. So, is ATM like Ethernet or any other network we have discussed?

Yes and no, unfortunately. If viewed from the perspective of other LAN technologies such as Ethernet/802.3, ATM can provide many services that are somewhat equivalent to Ethernet/802.3. It can also be used in a LAN for desktop connectivity instead of Ethernet/802.3. In this context, networking software "thinks" that the ATM environment reacts like OSI layer 2 hardware, as do like Ethernet/802.3, token ring, or FDDI. In terms of functionality, ATM can "be" like Ethernet/802.3 or, more specifically, can be used where Ethernet/802.3 might be used. How it works, however, is a completely different story and is not at all similar in function to other standard LAN technologies.

8. Before we get into some of the technical details of how ATM works, can you give a simple example of an ATM network?

Sure. An ATM network fundamentally resembles a frame relay network. Cells are transmitted from source to destination via an ATM switched subnet. End nodes communicate with an ATM device via a user-to-network interface (UNI), and ATM switches communicate with each other via a network-to-network interface (NNI). The UNI and NNI components are just two of a wide variety of ATM component standards. A typical configuration is shown in Figure 14.2. Also, a simplified example of how ATM facilitates multiservice networking is shown in Figure 14.3.

9. What other component standards are there besides UNI and NNI?

In ATM, there is a hierarchy of standardized interfaces called the *Anchorage Accord*. The basic hierarchy, as defined by the accord, is shown in Figure 14.4. Separate standards define the manner in which network components are interconnected and how they interact with each other. Working from left to right in Figure 14.4, the various components of the ATM environment are defined by the functional specifications listed. At this writing, the various numbers indicate current versions of the specifications. These version numbers, however, will be ever increasing over time as the standards mature and change. Shaded components are presently under development; all others are complete.

Note that the standards are divided into two basic groups: (a) foundation specifications and (b) application and service specifications. The former define ATM base function sets;

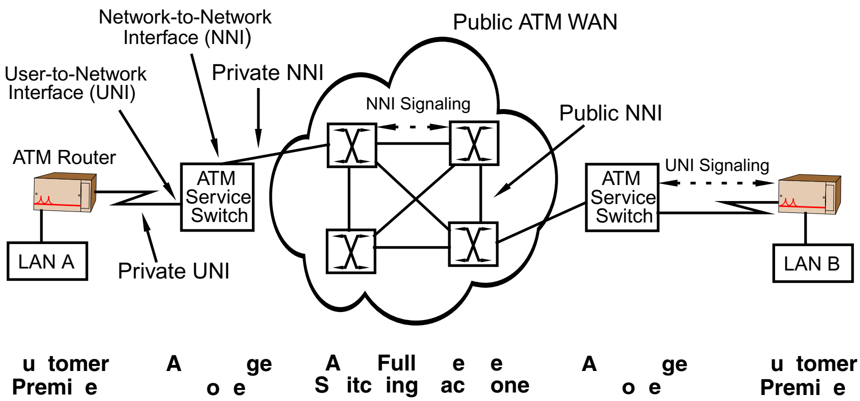


FIGURE 14.2 In an ATM connection, end nodes communicate with an ATM device via a user-to-network interface (UNI), and ATM devices communicate with each other via a network-to-network interface (NNI). Source: Adapted from Mehta, 1996.

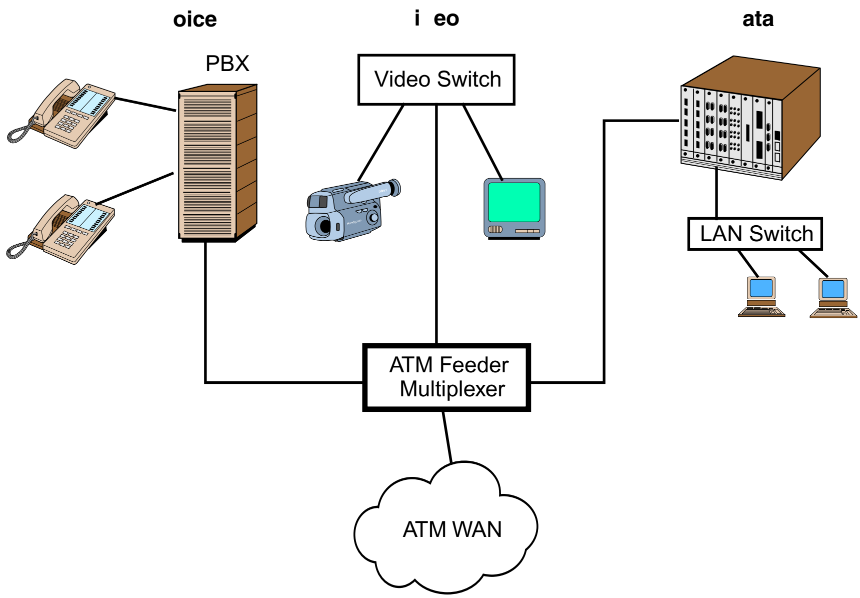
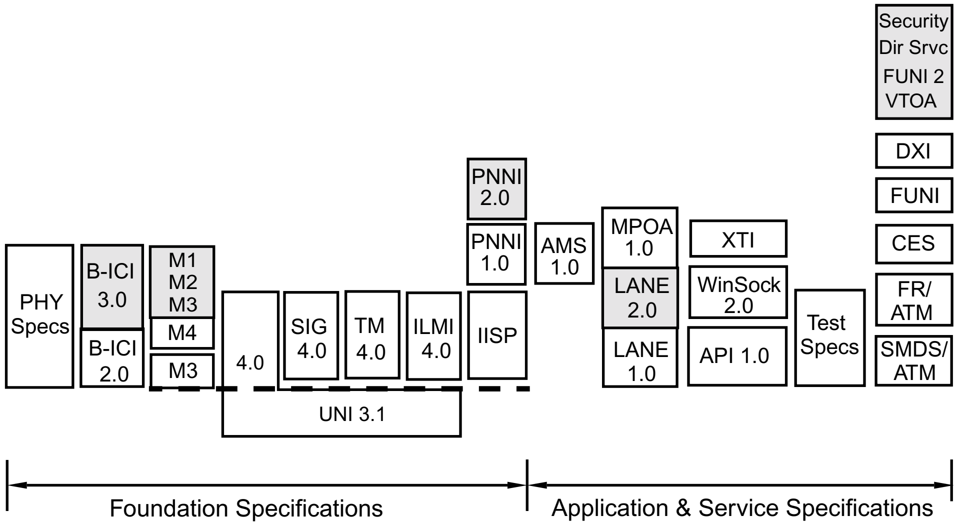


FIGURE 14.3 A simplified example of how ATM facilitates multiservice networking. ATM is the only network that was built from the ground up to support data, voice, and video at the same time. Therefore, an ATM network can be used for almost any type of network environment in use today and in the future.



A S	Audiovisual Multimedia Services
	Broadband ISDN Carrier Interface
S	Circuit Emulation Services
	Data Exchange Interface
F A	Frame Relay over ATM
F	Frame User Network Interface
SP	Interim Interswitch Signaling Protocol
	Integrated Local Management Interface
A	Local Area Network Emulation
	Multiplexing (e.g., M1 is multiplexed T1)
P A	Multiprotocol over ATM
P	Private Network-to-Network Interface
S G	Signaling Speci cations for UNI
S S A	Switched Multimegabit Data Services over ATM
	Traf c Management
	User-Network Interface
A	Voice and Telephony over ATM

FIGURE 14.4 The Anchorage Accord hierarchy of standardized ATM interfaces. (Shaded components are presently under development; all others are complete.)

the latter define how other networks and applications interoperate with ATM network components. It is outside the scope of this book to get into the massive details of each functional component due to their complexity and also the amount of time it takes to properly define and illustrate every item. However, we do need to spend a little time on some of the components to understand how ATM works and why the functional layers are necessary.

10. Is this how ATM is able to provide for the various types of services?

In a general sense, yes. In an ATM network, the type of computer being used is irrelevant to the network transportation capabilities. This means that PBXs and data networks from computer-to-computer or video-to-display facilities can all share the same network infrastructure simultaneously. This is done by using all the functional standards shown in Figure 14.4 to provide exact interfaces for different data types to the network infrastructure. In this manner, the network supports all the different types of interconnection services required as well as all the different types of transmission requirements.

11. How does an ATM network keep all the traffic sorted out so that the right messages get to the right places?

Just like any other type of network, ATM uses addresses for each connection point. Two types of addresses are used. For private networks, ATM uses OSI's (see Chapter 2) network service access points (NSAPs) for its addressing mechanism. NSAPs are 20-byte addresses and include a 13-byte prefix that can be used to identify a specific location including a country, region, or end system. Public ATM networks, however, use E.164, the standard ISDN global numbering addressing format specified by ITU-T (see Chapter 13). There is a move underfoot to standardize on NSAP-based addresses for all devices, public or private.

12. Could you expand on the components of an ATM network and explain how they interconnect with each other?

Certainly. Let's start with the source node. For a device to be connected to an ATM network, it must first have an ATM-based network interface card (NIC) installed. (See Chapter 6 for more information about NICs.) These NICs, which are layer-2 devices and contain on-board ATM circuitry, are then connected to a local switch that is usually provided by the corporate networking people. The physical connection between the workstation and the local switch (also called a "private switch") can be copper (typically UTP category 5 running at 25 Mbps or 155 Mbps), fiber (at various speeds from 155 Mbps to terabit ranges), or wireless (10 Mbps or greater). The specification for a NIC to "talk" to a switch is the user-to-network interface (UNI) functional specification, which was mentioned earlier. The purpose of the UNI specification is to define exactly how an end station communicates to the next level of network interface, which is a private switch at a site. This specification is quite large and complex. Among others, it specifies the handshaking between the NIC and switch, the quality of service requirements for traffic between the NIC and switch, and all the other connection-oriented facilities ATM requires to ensure that the NIC and switch are content with each other when connected on the network. Furthermore, UNI also specifies sections of the hardware physical attributes like copper interface issues, fiber interface issues and wireless component functionalities. These collective specifications and functions are generally referred to as UNI signaling (see Figure 14.2). Thus, UNI is specifically intended for end-point communication to the ATM switch environment.

13. Does this mean that switches communicate with each other differently?

Yes. Switch-to-switch communication in a private (premises) network is called *Private network-to-network interface* (PNNI). PNNI provides some specific functionality that switches need between each other but that end stations could not care less about. Other specifications exist as well. Two in particular are the broadband ISDN intercarrier interface (B-ICI) and the LAN user-to-network interface (LUNI). B-ICI is used for connecting two public ATM networks between different network service providers; LUNI specifies how an emulated LAN connects to an ATM environment. For example, if you have an existing Ethernet/802.3 or other LAN, then there is a specification called LANE (local area network emulation) to help you connect to an ATM network. Similarly, if you have an existing frame relay WAN facility that needs to interoperate with ATM, then the frame user network interface (FUNI) is used. There are many others. Thus, wherever there is a network connection point on an ATM network that connects to a dissimilar connection point, a functional specification (and related abbreviation, of course) associated with the connection method is defined (see Figure 14.4).

14. Tell me more about ATM switches.

ATM switches are responsible for transmitting data within an ATM network (see Figure 14.2). An ATM switch implements a routing method called a switch fabric. Switch fabric algorithms come in a wide variety of implementations, but the most common types are called *Batcher-Banyan* and *delta* switching environments. The algorithms in the switch exchange information with other interconnected switches to learn where source and destination SAP locations are in the overall network. By using these high-speed methods, a very large network of components can quickly be mapped and discovered when traffic hits the physical network and needs to be forwarded to the proper location. When first powered-up, switches configure themselves by going into an autodiscovery mode, which enables them to automatically learn about other connected switches. They also provide temporary NSAP addresses to ATM connection points that may need one to interoperate and help manage the overall network environment from a configuration and traffic management perspective.

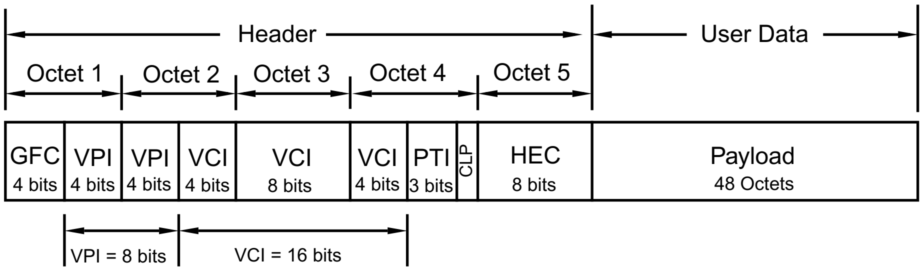
15. How does ATM transport data across the network via switches?

To answer this question, we need to discuss, from a more technical perspective, the three main concepts of an ATM network: ATM cells, virtual connections, and addressing. These three concepts represent the essential building blocks of all ATM networks and are highly interrelated. Let's begin with cells. Feel free to interrupt us at any time.

As we stated earlier in the chapter, ATM uses 53-byte (octet) cells comprising a 5-byte header and a 48-byte payload (user data). ATM cells are the basic unit of data that cross the UNI, and they are fixed length. Thus, cells are *always* the same size—never larger, never smaller. It is set up this way for throughput and buffer management reasons (video gets real unhappy when the network is slow). The structure of a cell is shown in Figure 14.5. The cell's header carries the information needed to transport a cell across an ATM network.

16. Time out. I need a point of clarification. Why is the word "octet" suddenly being used instead of "byte?"

The term *octet* means eight bits. The term *byte*, however, does not necessarily mean eight bits to some people. During the early days of computing, byte was used to represent



GF Generic Flow Control controls the ow of data across the user-to-network interface (UNI) permitting multiple ATM devices to be attached to the same network interface. GFC bits are reassigned and become VPI bits at the network-to-network interface (NNI), thus providing additional support for more virtual paths.

P Virtual Path Identi er part of the network address used to identify a grouping of channels between network entities. The rst 4 bits of the VPI are in the rst octet; the remaining bits are in the second octet.

Virtual Channel Identi er a pointer that identi es the virtual channel the system is using on a particular path. The VPI/VCI combination makes up the data link running between two network nodes. The VCI is 16 bits long and uses the second 4 bits of octet 2, all 8 bits of octet 3, and the rst 4 bits of octet 4.

P Payload Type indicates the type of information contained in the cell. Because cells are used for transporting different types of information, several types of payload type indicators have been de ned:

A 0 in the most signi cant bit (000, 001, 010, and 011) denotes that the cell is carrying *user information*. User information cells with a middle bit of 1 (010 and 011) indicate that the cell has experienced congestion.

A 1 in the most signi cant bit (100, 101, 110, and 111) denotes that the cell is carrying *control* or *resource management information*.

¥ 100 and 101 represent network maintenance and control.

¥ 110 is for resource management.

¥ 111 is reserved for future de nition.

P Cell Loss Payload speci es whether or not to discard the cell in the presence of congestion. If set to 1, the network can discard the cell. If set to 0, then the cell might not be discarded. (Similar to BECN and FECN in frame relay.)

Header Error Control provides error correction for single-bit errors and error detection for multiple-bit errors in the cell header.

FIGURE 14.5 Structure and contents of an ATM UNI cell. Note that an NNI cell is nearly identical to the UNI cell. The only difference is that there is no need for the GFC field. In the NNI cell, GFC is subsumed by VPI, which totals 12 bits instead of 8. Source: Adapted from Atkins & Norris, 1995.

a unit of information composed of some number of consecutive bits. Thus, depending on the system, you could have 4-bit bytes, 8-bit bytes, 16-bit bytes, 32-bit bytes, and so forth. As the computer industry matured, most people accepted a byte to have exactly 8 bits. The use of octet in ATM instead of byte can be viewed from two perspectives. The first perspective is one of definition: Octet is the better term since it is well-defined—it has always represented exactly eight bits of information. The second perspective is one of snobbishness: Some people use the word "octet" to remind others that they have been around systems a long time and that some systems used bytes of varying sizes.

17. OK. Thanks. Let's get back to the ATM cell. Given its size, how can a cell carry routing information? For example, LAN addresses used for Ethernet or token ring are 6 bytes. Thus, source and destination addresses must occupy 12 bytes. How does this fit into a cell?

This is true *if* ATM were using LAN-type addressing schemes. Remember: ATM networks are based on a "network within a network" concept and, therefore, can do some clever things that don't require each cell to carry complete source and destination addresses in them like a LAN frame does. To better understand this, we first need to introduce the second of ATM's basic building blocks, namely, virtual connections.

18. Please explain.

In ATM, two types of virtual connections are defined: *virtual channel connection* (VCC) and *virtual path connection* (VPC). A virtual channel connection is a virtual circuit that provides a logical connection between a source and destination. These virtual circuits can be either permanent (PVC) or switched (SVC). PVCs and SVCs are discussed in detail in Chapter 12. A *virtual channel identifier* (VCI), which is contained within the header of a cell (Figure 14.5), is used to identify the channel. VCIs are similar to frame relay's data link connection identifiers (DLCIs), which were discussed in Chapter 12.

A virtual path connection is a semipermanent connection that provides a logical collection of virtual channels that have the same endpoints. In other words, a VPC carries a group of virtual channels all with the same endpoints. For example, a multimedia application between two physicians' workstations might consist of a single path that supports two virtual channels: one for computer conferencing and a second channel for transferring large image files such as x-rays. Thus, in this illustration, a single virtual path supports two virtual channels. For another application, consider the common situation of an organization providing connectivity to a remote site. In a non-ATM environment, separate links are needed for voice, data, and video support. With ATM, though, a single virtual path can be established that supports several virtual connections. For example, one VCC could be for telephone service, a second VCC could be used for a frame relay connection, and a third VCC could be used for videoconferencing. Thus, virtual paths enable any connection that uses the same network path from source to destination to be bundled into a single unit. A virtual path identifier (VPI) is used denote a virtual path and is included in a cell's header (see Figure 14.5). A virtual channel can only be established after a virtual path connection is in place. An illustration of VCCs, VPCs, VPIs, and VCIs is given in Figure 14.6. Note that on each physical link a VPI specifies a virtual path and a VCI specifies a specific virtual channel.

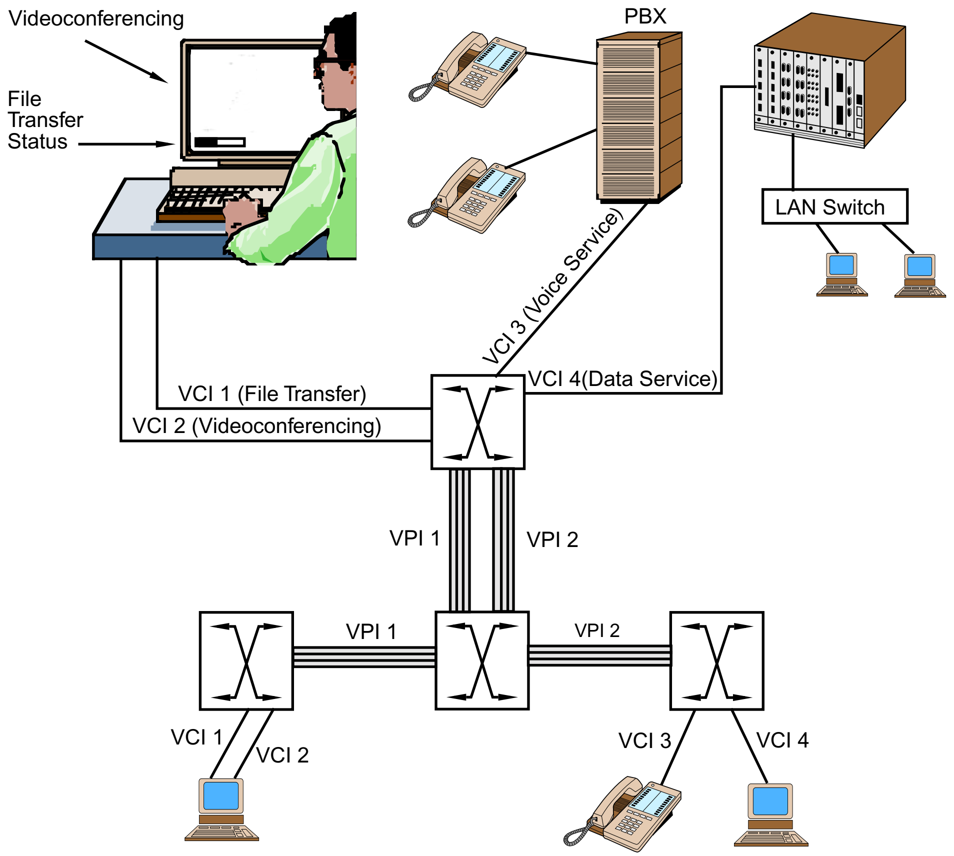


FIGURE 14.6 In an ATM network, a virtual path connection is a logical collection of virtual channels that have the same endpoints. Virtual paths indicators (VPIs) and virtual connection identifiers (VCIs) are used to label these respective connections. In this illustration, virtual channels 1 and 2 are carried by the same virtual path (VPI 1), and virtual channels 3 and 4 are carried by the same virtual path (VPI 2). (VPI 1 and VPI 2 between the two central ATM switches *could* be combined into a single path and still carry all four virtual circuits and their corresponding VCIs). Source: Adapted from Minoli, 1992b.

Further note that VCIs only have local significance. Thus, different virtual paths may reuse VCIs. However, VCIs on the same virtual path must be different. Finally, virtual paths can be used to establish a logical network topology independent of the physical topology.

19. Referencing your last statement, does this imply that there doesn't have to be a physical link between switches?

That's correct. Two switches can be interconnected via a VPC without having a direct physical link between them. Remember: ATM is connection-oriented, which means that a circuit is first established prior to data transmissions. Thus, as part of a call setup, two nonadjacent switches without a physical link can be included as part of a virtual path for a particular session. Note further that virtual paths can also provide a form of traffic control by logically (not physically) partitioning network traffic based on the type of data being carried and associated quality of service.

20. How do VPIs and VCIs help with addressing?

By using virtual path information to move data between connection points on the network, a full addressing scheme is not necessary. This means that much smaller "addresses" can be used to move information between connecting points on the network. By using such a scheme, full addressing information for the source and destination is not required in each cell and this reduces overall overhead in the cell formats. The process (albeit simplified) is as follows: An ATM switch receives a cell at an incoming port. The switch examines the cell's header to identify the values contained in the VPI/VCI fields and then consults its translation table to determine the appropriate outgoing port based on the VPI/VCI values. Once the correct port is identified, the switch transmits the cell out the port.

21. So instead of carrying source and destination addresses as TCP/IP packets do, ATM uses VPIs and VCIs to transport cells across the network. Is this correct?

Yes it is. In a nutshell, ATM switches are switched based on information contained in the cell header. In other words, ATM transports data via fixed-length cells based on the VPI/VCI values contained within a cell's header. During call setup, a route from source to destination is established by ATM end nodes and switches. As this path is being established, VPIs and VCIs are assigned. Thus, the determination of a route is made only once. Furthermore, unlike TCP/IP internetworking devices (routers or layer-3 switches), ATM switches do not have to decode large and relatively complicated headers in order to determine the appropriate output port. A direct consequence of this simplification is that forwarding decisions are implemented in silicon, thus enabling them to be implemented at wire speed. At one time this was considered an ATM advantage. However, ASIC technology has since enabled routers to perform traditional table lookups and packet forwarding at hardware speeds, thus providing them with switch-like performance (see Chapter 7). ATM also avoids collisions by having users negotiate throughput and quality of service in advance. Using these parameters, ATM can multiplex several users, each on a separate virtual channel, onto a single virtual path. Additional virtual paths can then be added by increasing the size of the switch.

22. Can you summarize the various concepts related to ATM addressing for me?

Sure. Each ATM station has a unique hardware address—called a *station identifier*—that is part of its network interface card. A station identifier is similar to a MAC sublayer address used in Ethernet/802.3 and token ring LANs. In addition, an ATM station also has a network layer address (or equivalent) that provides the station with a global address. The addressing method used for these network layer address equivalents is either NSAP (for private ATMs) or E.164 (for public ATMs). These two addresses are the only addresses that actually bind an ATM station to a physical link. Finally, when a connection is established from source to destination, VPIs and VCIs assignments are made that specify the virtual connection path. It is these VPI/VCI assignments that get incorporated into a cell's header and are used for route determination. Note that this is a rather simplified summary of ATM addressing concepts and does not include issues such as looping, multicasting, or routing protocols.

23. OK. I think I have a fairly good grasp of some of the basic ATM concepts. What I still do not understand, though, is how ATM can support different types of data simultaneously.

ATM accommodates different types of traffic via the *ATM Adaptation Layer* (AAL) protocol (see Figure 14.1), which consists of two sublayers: convergence; and segmentation and reassembly (SAR). On the sending side, the convergence sublayer accepts data messages from higher-layer application protocols, interprets the type and format of the messages, and then prepares them for processing by the SAR sublayer. The SAR sublayer then translates the messages into ATM format by packaging them into the 48-byte payload portion of an ATM cell. The reverse is done on the receiving side. The AAL protocol actually does more than what is described here. For example, the AAL also resolves transmission errors, takes care of lost or misinserted cells, and provides flow and timing control. We will not discuss these services.

The convergence sublayer's interpretation of data type and format is based on the specific *class of service* (CoS) assigned to the data by the application. The AAL provides support for four different service classes. These are summarized in Table 14.1. When the SAR sublayer receives data from the convergence sublayer, it processes the data so they are consistent (i.e., meets the transmission needs) with the specified type and format. Depending on the data type, the AAL protocol provides five different AAL types to accommodate a particular service class. For example, AAL1 is used for data that require connection-oriented, constant-bit rate transmissions. An example of this data type is traditional voice service. AAL2 is used for data types that require connection-oriented variable-bit rate data transmissions. An example of this data type is packet video, which might be used in a videoconferencing application. AAL3 and AAL4 are used for connection-oriented or connectionless variable-bit rate transmissions. Examples of these data types include bursty data typical of LAN applications such as those found on frame relay (connection-oriented) and SMDS (connectionless) networks. Lastly, AAL5, which initially was labeled the simple and efficient adaptation layer (SEAL), is used for transmissions in which higher layer protocols provide error recovery. AAL5 is an improvement to AAL3. As a result, through the AAL protocol and its adaptation layers, an ATM transmission is able to accommodate different types of transmissions simultaneously on the same network. The correspondence between AAL types and service classes is summarized in Table 14.2.

24. There almost seems to be a one-to-one correspondence between service class and AAL type. Is this the way it was designed?

The initial design was to have a different AAL type for each service class. Specifically, AAL1 through AAL4 were designed, respectively, to support service classes A through D. As in most cases, though, reality got in the way. When AAL3 and AAL4 were being

TABLE 14.1 ATM Service Classes and Descriptions

Service Class	Description
Class A	Constant-bit rate (CBR) connection-oriented transmissions that require a strict timing relationship between source and destination nodes (e.g., circuit emulation and voice transmissions).
Class B	Variable-bit rate (VBR) connection-oriented transmissions that require a strict timing relationship between source and destination nodes (e.g., synchronized data transmissions or packet-mode video for videoconferencing).
Class C	Connection-oriented VBR transmissions that do not require a strict timing relationship between source and destination nodes (e.g., LAN data transfer applications such as frame relay).
Class D	Connectionless VBR transmissions that do not require a strict timing relationship between source and destination nodes (e.g., LAN data transfer applications such as SMDS).

TABLE 14.2 ATM Adaptation Layers (AAL) Types and Corresponding Service Classes

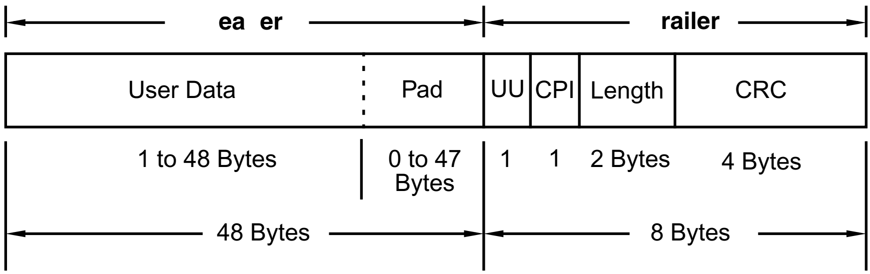
AAL Type	Service Class
AAL1	Class A
AAL2	Class B
AAL3	Classes C and D
AAL4	Classes C and D
AAL5	Classes C and D

developed, it was observed that there was considerable overlap between the two specifications. Consequently, they were combined into a single type, which is now referred to as AAL Type 3/4.

25. So how did AAL5 get created?

AAL5 was developed to provide a less complex Type 3/4 AAL. Many of the services supported in classes C and D did not warrant the level of sophistication that was incorporated into Type 3/4. As a result, a more simple and efficient adaptation layer (hence AAL5's original name, SEAL) emerged.

Presently, only AAL5 is widely implemented and used. The reason is that one of the ATM standards bodies, the ATM Forum, originally defined AAL5 for efficient transmission of TCP/IP. Because AAL5 uses a very small amount of the cell for its overhead, it is very popular in its use by vendors and ATM technology implementors. Whether AAL is



er ata	This eld is the actual user information being transported across the network. Although it can range in length from 1 byte to 65,535 bytes, it is segmented into 48-byte chunks.
Pa	This eld is used to pad the data unit to 48 bytes. Its length varies from 0 to 47 bytes.
	This 1-byte *user-to-user* eld allows 1 byte of information to be conveyed transparently between users.
P	This 1-byte *common part indicator* is intended to identify subsequent elds, but it is not currently being used.
engt	This 2-byte eld stores the length of the transmitted data. This eld is necessary to distinguish between user data and padding.
	The *cyclic redundancy check* eld uses a 32-bit checksum to provide data integrity of the entire PDU.

FIGURE 14.7 Format and contents of the AAL5 segmentation and reassembly (SAR) protocol data unit (PDU). Source: Adapted from Atkins & Norris, 1995.

used or not in a connection is up to the vendor of the protocol stack that uses the ATM hardware in a network topology. Given the popularity of the AAL5 protocol, the format of its protocol data unit (PDU) at the segmentation and reassembly (SAR) level is shown in Figure 14.7. Recall that a block of higher-level data is encapsulated in a PDU at the convergence sublayer. This PDU is then passed to the SAR sublayer where it is segmented into a 48-byte "chunks" so that it can be transported in the payload area of an ATM cell. What is shown in Figure 14.7 is the format of AAL5's SAR PDU. As can be seen, a cell can have many additional fields in it besides the general ATM format.

26. So all the other AAL layers really are not used?

Right. Some of them will probably be implemented over time, new layers probably will be invented, and older ones discarded. It's the Zen of Networking at work.

27. OK. Now that I have a basic understanding of ATM, why do I care about it?

As with any network environment, there is change. If you look at the direction of overall network trends, you'll see a strong movement toward the convergence of data, voice, and video applications. (See Chapter 17 for more information about the concept of convergence technology.) To satisfy all three network types, some basic functions must be created to allow all three types of network technologies to coexist.

28. What kind of technology is needed for data, voice, and video?

Data networking allows traffic characteristics that can be bursty in nature, and data can have variable-length packets or frames. Data traffic also can tolerate a certain amount of transmission delay, especially for non-real-time traffic. This implies that data arrival rates can be variable in nature, which suggests that *variable-bit rate* (VBR) transmissions are acceptable to data applications.

Voice traffic, on the other hand, is more sensitive to the arrival time of traffic. It's a good idea for packets carrying part of a conversation to arrive quickly enough to avoid blank spots of time ("dead air") in the conversation. This suggests a *constant-bit rate* (CBR) transmission be used for voice. There also is a problem with full duplex speech, where two or more people speak at exactly the same time and can be heard simultaneously. Networks don't necessarily agree with this concept very well. Voice, however, is not a big bandwidth hog, as a rule, so fast data networks with VBR can support voice communications as long as the network is swift and does not suffer congestion or loss delays.

Video is even more sensitive to timing. Such transmissions expect the number of frames transmitted from one site to another to arrive in order and in a very specific time frame (usually measured in milliseconds). Technologies such as constant-bit rate, where the number of arrival bits in a transmission are constant and consistent, are essential to making a standard traffic arrival rate possible. However, CBR is not enough. The frames must be transmitted in the proper order and must arrive at the correct speed within a specific time frame. To do this, the network must reserve bandwidth in the path from the source to the destination to ensure that all the bits arrive in order and on time. This general method of providing CBR with a guaranteed delivery sequence in a specific time frame while reserving path "space" is called *isochronous communications* and is common to all ATM networks.

When you add to the discussion that different network applications will ultimately require more and more bandwidth in the future to satisfy consumer needs, ATM is the only network presently on the drawing board that provides the transmission technologies, transmission rates, and quality of service (QoS) required to address user needs . . . at least for now (e.g., it is still too early to determine 10 Gigabit Ethernet's impact).

29. What you're saying, then, is that the convergence of network functions, which is the "wave of the future," requires technologies analogous to ATM. Is this correct?

Yes.

30. What about Gigabit Ethernet and 10 Gigabit Ethernet? Aren't these two technologies (especially 10 GbE) fast enough for all of this?

Although 1 GbE and 10 GbE are capable of transmitting data and voice at acceptable levels, they are still a VBR technology and have problems with video equipment. This becomes acutely evident in the presence of network congestion, or when a specific delivery time frame is required. A good example of this is High Definition Television (HDTV), which will eventually appear on global networks as a standard transmission method.

31. Even so, I thought protocols such as IEEE 802.1p, IEEE 802.1q (discussed in Chapter 5), and RSVP (discussed in Chapter 7) and RTP were supposed to remedy Gigabit Ethernet's QoS shortcomings.

Presumably, the *Resource Reservation Protocol* (RSVP) and the *Realtime Transport Protocol* (RTP) as well as IEEE 802.1p/802.1q will remedy 1 GbE and 10 GbE's QoS shortcomings. Such protocols permit applications to reserve a specific amount of bandwidth for data transmission. However, when you consider what the incorporation of these protocols within an Ethernet LAN brings to the table, and then compare their functions to ATM, several problems emerge. First, Ethernet/802.3 frames are of variable length ranging from 64 to 1518 bytes. This alone suggests that the delivery rate will not be consistent. ATM uses fixed-size cells, which does guarantee a constant delivery rate. Second, in an Ethernet/802.3 transmission, frames are queued in a switch on a first-in first-out basis. (See Chapters 6 and 8 for information about Ethernet switches.) Furthermore, before a switch transmits queued frame, n, the entire contents of queued frame, $n - 1$, must be transmitted. Thus, a switch transmits queued frames sequentially and in the order in which they were buffered. The problem with this scheme is that if two frames, both of which are reserving bandwidth, arrive at a switch port simultaneously (within a few microseconds of each other), the frame arriving first gets transmitted first, and the frame arriving second gets buffered. ATM switches, on the other hand, can simultaneously create and service multiple, independent queues having different priorities (class of service) and different transmission needs based on data type (i.e., quality of service). Moreover, these simultaneous and multiple transmissions are performed with a constant delivery rate. Additional information about the Gigabit Ethernet-ATM debate is provided in Chapter 8.

32. So, just making the network faster will not solve the convergence problems?

That's right. Even the work on Terabit Ethernet, currently underway at this writing, will yield a network faster than the currently fastest commercially available ATM network (today). However, it is not slated to have the proper traffic controls and network reliability factors that are inherent in an ATM environment. It is also not expected to provide a network larger than that used in a campus environment or possibly around a city (in a metropolitan network environment).

33. Well, what about the campus environment? Isn't ATM a bit pricey? Doesn't its data rate of 25 Mbps fall short when compared to Fast Ethernet or Gigabit Ethernet LANs?

Good question. ATM was initially designed as a WAN technology for use in B-ISDN. This has not stopped its encroachment, however, as a technology for LANs (or for MANs). ATM's infiltration on the LAN front was designed to serve as a 155-Mbps backbone technology and deliver 25 Mbps (called "low-speed" ATM) to the desktop. This grand scheme of a unified ATM LAN/MAN/WAN, however, got hijacked by Ethernet, specifically, Fast Ethernet, which delivers 100 Mbps to the desktop, and Gigabit Ethernet, which provides a 1000/10,000-Mbps backbone. Thus, compared to Fast and 1 and 10 Gigabit Ethernet, ATM data rates are low, and the deployment of ATM is quite expensive.

34. What else does ATM have available for LAN support?

For starters, there's ATM's *local area network emulation* (LANE) interface (see Figure 14.4), which provides a technology that enables ATM to emulate Ethernet/802.3 or token ring. LANs that incorporate LANE are called *emulated local area networks* (ELANs). In ATM's protocol hierarchy, LANE is above AAL5 in the ATM adaptation layer. The LANE protocol defines a service interface for the network layer that functions identically to the one used by Ethernet/802.3 and token ring LANs. Data that cross this interface are encapsulated in the appropriate MAC sublayer format. In an ELAN environment, LAN end nodes are connected to a special LAN emulation device that runs a LAN emulation client (LEC) process. The LEC functions as a proxy ATM end node. In addition, a native ATM end node runs a LAN emulation server (LES) process, which is responsible for resolving MAC hardware addresses to ATM addresses.

As an example of this process consider the situation where an Ethernet/802.3 source node wants to transmit a data frame to an Ethernet/802.3 destination node across an ATM switch fabric. To do this, the source node transmits a frame of data to the LEC process that resides on a LAN emulation device. The LEC then issues an ARP (address resolution protocol) broadcast requesting a MAC-to-ATM address resolution. (See Chapters 3 and 7 for more information about ARP.) The MAC address is the hardware address of the transmitting Ethernet/802.3 end node. An LES process, which resides on a native ATM device, responds to the ARP broadcast and returns to the LEC the ATM address of the remote LAN emulation device (and residing LEC) to which the destination node is connected. The source LEC then establishes a virtual circuit to the destination LEC. The LAN emulation device then translates the Ethernet/802.3 frame into an ATM cell via the SAR sublayer (Figure 14.8). Note that implementing this scheme does not involve modifying any higher layer protocols. Note further that in this implementation, the ATM network is effectively operating as a fast data link layer for the Ethernet/802.3 network.

In addition to resolving MAC-ATM address issues, another issue that needs attention is connection type. LAN protocols are connectionless; ATM is connection-oriented. Thus, if ATM is going to emulate an Ethernet/802.3 or token ring LAN, then it must be able to carry full source and destination addresses in every frame transmitted. This poses quite a challenge given ATM's five-byte header. Furthermore, a connection-oriented technology does not adequately support bidirectional broadcast or multicast transmissions, which are inherent in Ethernet/802.3 and token ring LANs. To address these issues, a special UNI (user-to-network interface) was developed for LANs. This LAN UNI (called LUNI) enables ATM to emulate the connectionless nature of LANs across the ATM switch fabric.

A second strategy ATM provides for desktop support is a technology called *cells in frames* (CIF), which defines a method for transporting ATM protocols over Ethernet and token ring LANs. CIF is a LAN technology that provides LANs with ATM features including QoS and the seamless integration of data, voice, and video. CIF extends ATM's virtual connections to the desktop via a special CIF attachment device, which provides an interface similar to ATM's frame user-to-network interface (FUNI). This device performs most of the segmentation and reassembly (SAR), but the Ethernet/802.3 or token ring node still must build the ATM Layer's PDU. See the Web site at http://cif.cornell.edu for more information available from the CIF Alliance. A second resource for CIF is the document http://www.ziplink.net/~lroberts/Atmf-961104.html.

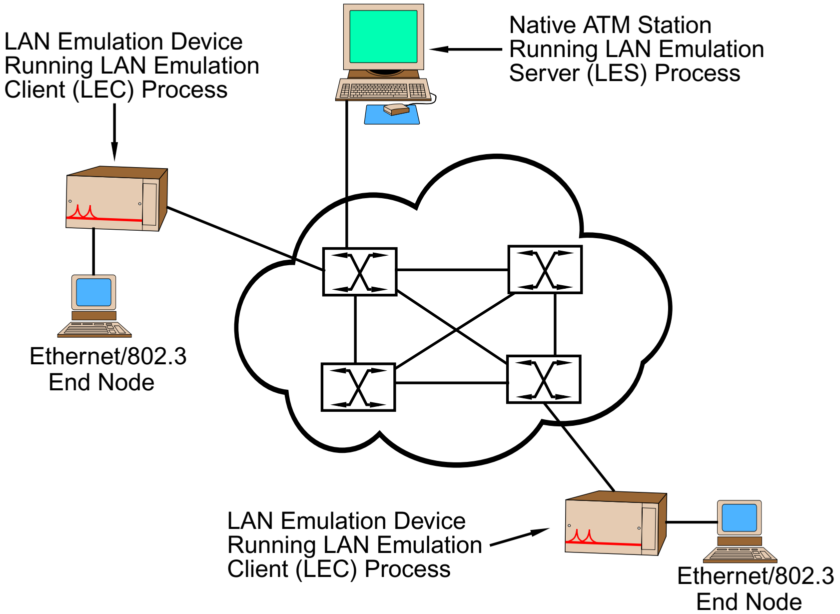


FIGURE 14.8 A typical ATM local area network emulation (LANE) configuration in an Ethernet/802.3 environment. Source: Adapted from King, 1994.

35. OK. I think I have had my fill of LUNI and FUNI to the point where I'm going loony now. Let's wrap things up with a few more "clean-up"–type questions. First, how does frame relay stack up against ATM?

Recall that ATM is similar to frame relay. In fact, ATM is sometime referred to as cell relay to distinguish it from frame relay. The key difference is that frame relay uses variable-length frames as its main transmission unit, whereas ATM uses fixed-length packets of exactly 53 bytes known as cells. Although ATM's use of smaller, fixed-length cells results in higher overhead than frame relay incurs, it also provides two critical advantages over frame relay, namely, speed and traffic type. Since all ATM cells are exactly the same size, they are much easier (and hence, faster) to process. Second, by using short cells with predictable transmission delay, ATM can combine cells carrying delay-sensitive traffic like interactive video and voice along with data cells. This concept, called *interleaving*, isn't possible with frame relay because longer data frames create longer and unpredictable delays when processing voice and video traffic. Thus, frame relay is less suitable for real-time videoconferencing, for example.

36. Second, please explain the difference between ATM and SONET.

When discussing ATM and SONET, it is important to realize two things. First, SONET is nothing more than a transport mechanism (see Chapter 7), and second, ATM does not require the use of any specific physical layer protocol. As a high-bandwidth carrier service, SONET can serve as the transport facility for any network technology or service, including ATM, FDDI, SMDS, and ISDN. SONET also can support various topologies including point-to-point, star, and ring. Frequently, though, SONET is used to carry ATM traffic. The two are kind of linked to each other to the point where they can be considered as "words that come in pairs" (e.g., hue and cry, or table and chair)—ATM and SONET. There is a reason for this. Instead of developing a new physical layer, the designers of ATM borrowed SONET's link-level technology and used it for ATM switching. Furthermore, the ATM Forum has defined 622-Mbps ATM (OC-12) (and higher) to run over only SONET. This does not mean though, that OC-12 is ATM. As we discussed in Chapter 7, OC-12 is simply the label given to denote the concatenation of 12 DS-3 channels, which provides an aggregate bandwidth of 622.08 Mbps. So, in a nutshell, SONET is a carrier service that transports bits from a source to destination, and ATM is a technology and protocol that was designed to use SONET as its carrier service.

END-OF-CHAPTER COMMENTARY

This concludes our discussion of ATM. The information contained in this chapter provides only a working overview of the technology. For detailed technical information, you are encouraged to visit the ATM Forum Web site (http://www.atmforum.com), or consult some of the references given for this chapter. You might also want to revisit other chapters within this book that included ATM as part of the discussion, most notably: Chapter 7, which includes a discussion of SONET; Chapter 8, which examines Gigabit Ethernet (1 GbE and 10 GbE) and its relationship to ATM; and Chapters 12 and 13, which provide a comparison of frame relay, SMDS, and ATM. The Glossary also provides definitions of additional ATM terms, and the Web site http://www.techfest.com/networking/atm.htm is a terrific resource for ATM information. Finally, as is the case with many technologies, ATM continues to evolve, and there is one evolving technology that has caught the eye of WAN operators. It is CIF, which was mentioned earlier as a technology for transporting ATM protocols over Ethernet and token ring LANs. The CIF Alliance is actively modifying CIF to work over SONET and PPP links. This proposed WAN version of CIF presumably maintains all of native ATM's key features but requires less overhead. The ATM Forum is working with the CIF Alliance to bring the concept of using CIF to carry ATM protocols to fruition.

Chapter 15

Dialup and Home Networking

In this chapter we discuss the concepts and methods of extending networking to the home or small office/home office (SOHO) environment. This discussion involves dialup networking, which entails using the telephone system to dial into a network, dedicated, "always on" connections via DSL or cable services, and home-based LANs. An outline of the major topics we address follows:

- Dialup Networking Concepts and Issues (Questions 1–3)
- Modem Concepts: Analog and 56K Modems (Questions 4–14)
- *x*DSL Connections (Questions 15–17)
- Cable Modem Connections (Questions 18-23)
- Home-Based Internet Connections (Questions 24–27)
- Home-Based LANs: Concepts and Issues (Questions 28–34)

1. What is dialup networking?

Dialup networking refers to a network connection that is established by "dialing" into the network through the public telephone system. Dialup connections can be analog or digital. Analog connections involve the use of a modem; digital connections require end-to-end digital connectivity. (See Chapter 11 on ISDN for more information about end-to-end digital connections.) Dialup connections generally can be classified as either terminal-based or network-based.

2. What's the difference between a terminal- and network-based connection?

A *terminal dialup connection* involves the use of special terminal emulation software, which makes the local system a terminal of the remote machine. This type of connection is sometimes called a *tty* connection, which is an acronym for *teletype*, a term used in the early days of computing to denote a terminal connection between a device and a centralized host. In a tty dialup connection, terminal emulation software is used to make the local system (the one dialing in) appear as a terminal to a centralized host. There are many different kinds of terminal emulation software available, each with its own capabilities. Two in particular are Kermit and ProComm. Windows 3.1, Windows 95/98/2000 also come bundled with terminal emulation software, and modem manufacturers usually include a version of their own emulation software with their modems. Commonly emulated terminals are vt100 and vt102. In a dialup tty environment, control of the local machine is transferred to that of the remote system. The emulation software converts the local node into a terminal of the centralized host. A consequence of this conversion is some keyboard functions or mouse capabilities do not work correctly if they are not mapped properly and supported by the remote system.

In a tty dialup connection, the remote system (the one being dialed into) can be another computer as shown in Figure 15.1(a), or it can be a specialized device called a *terminal server* or communications server that is connected to a LAN, as in Figure 15.1(b). When a terminal or communications server is used, the server acts as an intermediary node that establishes a terminal connection between the local system and a networked machine. For example, in Figure 15.1(b), the PC client dials into the terminal server, which then enables the PC to connect to any machine on the local network, or (assuming proper authorizations are in place) any machine connected to the Internet. This is the basis of what is referred to as a UNIX "shell account" available from some Internet service providers.

Although it is possible to dial into a computer that is connected to a network and gain access to the network via this system, a dialup tty connection is not a network connection. This is because the local machine is considered nothing more than a terminal of the remote system. Anything you do through this connection (e.g., read e-mail, transfer files) is all done relative to the remote system, not the local host. For example, let's assume that in Figure 15.1(b) a terminal connection is established from the local PC to one of the networked hosts via the terminal server. If you read e-mail on this host and save it, it will be saved on the host, not the PC. Similarly, if you transfer a file from the Internet and save it, it will be saved on the host, not the PC.

In a *network dialup connection*, special networking software makes the local machine become a true networked host. A network dialup connection transforms the local machine into a networked node and provides it with the same capabilities of any other networked node exactly as if it were directly connected to a LAN.

3. OK. But I use a PPP account from home to connect to the Internet. Is this considered a networked connection or a terminal connection?

A Point-to-Point Protocol (PPP) link, which is common among home-based Internet users, is an example of a networked-based dialup connection. PPP enables a local machine to become a directly connected Internet host via a telephone line. Thus, in Figure 15.1(b), a networked dialup connection effectively makes the local PC appear as if it were directly connected to the LAN. Furthermore, all actions performed on the local machine are now done relative to the local machine and not to some remote node, as is the case with a tty connection. Networked dialup connections are established using special networking software such as PPP or SLIP (Serial Line Internet Protocol). On the local side, the client version of PPP or SLIP makes a connection to a remote machine running the server version of PPP or SLIP. The remote machine can be another computer or it can be a terminal server. Once the connection is established, the local machine is assigned a network address.

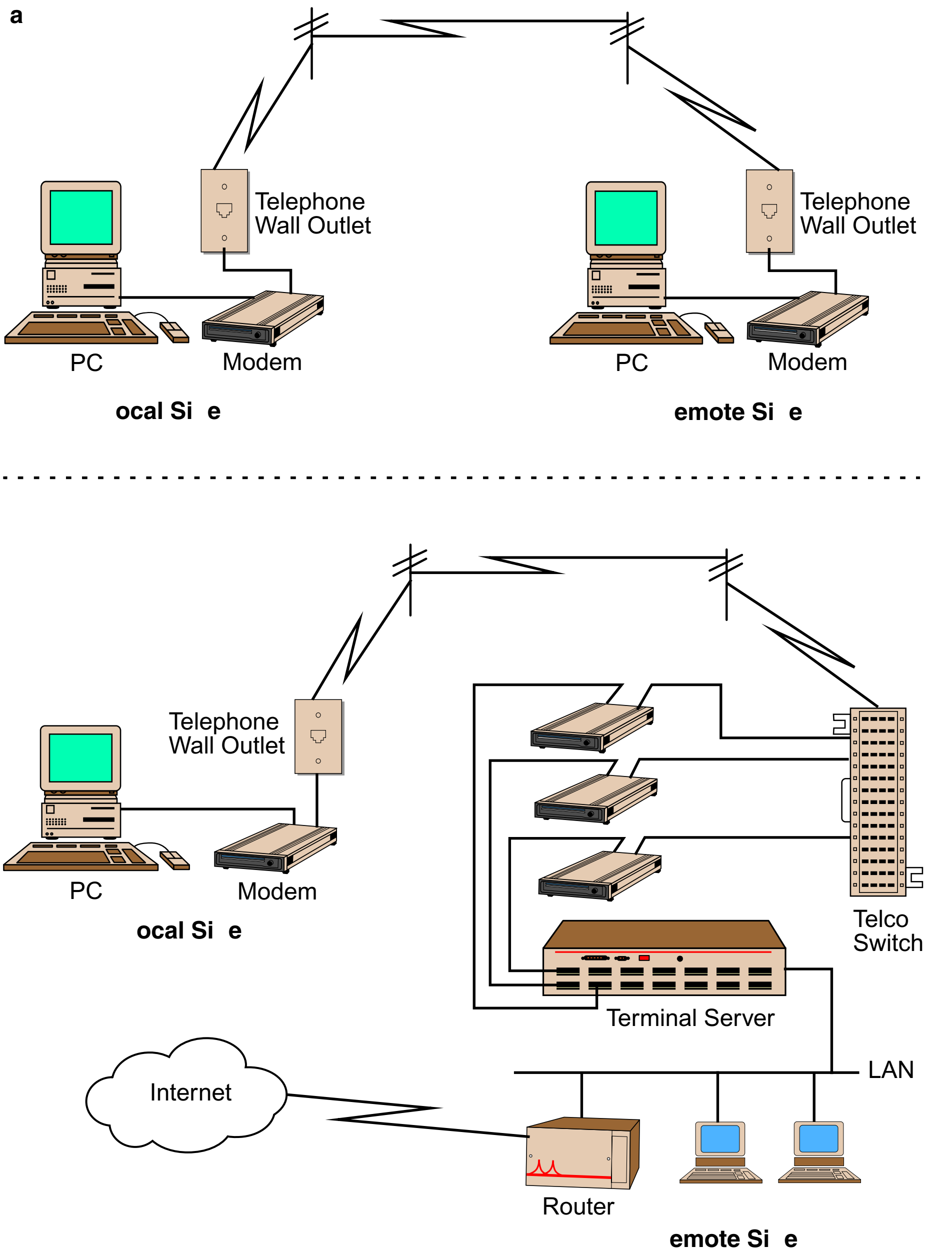


FIGURE 15.1 A typical dialup connection uses a standard telephone line as the medium for providing a computer-to-computer connection (a) or a computer-to-network connection (b) via a terminal server.

Depending on the configuration, the server either selects from a pool of available addresses and dynamically assigns the remote machine a network address, or the server issues a preassigned address. Regardless of the type of assignment, the network address remains in effect for the duration of the session.

4. Regardless of the type of dialup connection, though, is a modem is required? Also, could you review the concept of a modem?

That's correct. Since a dialup network connection involves "dialing" into the network through the public telephone system, dialup connections require modems—one at each end of a transmission line. This is because to transmit digital data (i.e., output from a computer) across an analog-based communications network (the plain old telephone system, POTS) the digital data must be represented in analog form. A conventional modem, which stands for modulator/demodulator, transforms a computer's digital signal into analog form at the sending side so the signal can be carried across a standard telephone line. On the receiving side, the modem demodulates the signal. That is, it reconverts the transmitted analog signal from the phone line to digital form before it is passed to the computer.

5. Are there any modem standards as there are for other networking equipment?

Yes. There are many industry standards for modems. Some are defined by international committees, others by modem manufacturers. For universal interoperability, though, modems should be compliant with standards formalized by the telecommunications sector of the International Telecommunications (ITU-T), which establishes worldwide communications standards. ITU-T is the former Consultative Committee for International Telephony and Telegraphy (CCITT). Modem standards defined by ITU-T are known as the V-series (ITU-T prefaces these standards with the letter "V"), and specify techniques for modulation, error control, and compression. A brief description of some of these standards is given in Table 15.1. Note that a second or revised version is denoted by *bis*, and a third version is denoted by *ter*.

6. How do modems work?

The actual technical details involving modem operation are beyond the scope and objectives of this book. Conceptually, though, modems work in the following way: When a dialup connection is first initialized, the two modems at each end of the connection begin screaming at each other—a negotiation process, called *handshaking,* that the modems engage in so they can come to some agreement on certain communication parameters. One parameter is the data transmission rate, which is formally known as the *DCE rate*, and informally referred to as "speed." DCE stands for *data communications equipment*.

7. Wait a minute. Explain this DCE rate please.

OK. In a dialup connection, the modem is a DCE device, and the telephone line connects the DCE device to the phone jack. Furthermore, the DCE-to-DCE rate is the data transmission rate between the two end modems. That is, it is the speed at which the two modems will exchange data. The ITU-T protocols specify standards for modem speeds.

TABLE 15.1 Selected V.*x* Modem Protocols

Protocol	Description
V.21	Standard for 300-bps modems using full-duplex transmission over a dialup line.
V.22	Standard for 600-bps and 1200-bps full-duplex modems over dialup and two-wire leased lines. Compatible with the Bell 212A standard used in the United States.
V.22 *bis*	Standard for 2400-bps full-duplex modems over dialup and two-wire leased lines; cycles to 1200- and 600-bps operation.
V.23	Standard for 600-bps or 1200-bps synchronous or asynchronous half-duplex modems used on dialup lines. Used in the United Kingdom.
V.29	Standard for 9600-bps facsimile service.
V.32	Standard for 9600-bps modems, cycles to 4800 bps when line quality degrades, and cycles forward when line quality improves.
V.32 *bis*	Standard that extends V.32 to 7200, 12,000, and 14,400 bps; cycles to lower rate when line quality degrades; cycles forward when line quality improves.
V.32 *ter*	Pseudostandard that extends v.32 *bis* to 19,200 bps and 21,600 bps.
V.34	Standard for 28,800-bps modems. (Note: Some V.34 modems were enhanced with new software that provided them with the capability to achieve data rates of 31,200 bps or 33,600 bps.)
V.FAST	Proprietary, pseudostandard from Hayes and Rockwell for modems transmitting at data rates up to 28,800 bps; served as a migration path for V.34.
V.42	Standard for error correction instead of for a modem. Uses LAPM as the primary error-correcting protocol, with MNP Classes 1 through 4 as an alternative (see Table 15.2).
V.42 *bis*	Standard that enhances V.42 by incorporating the British Telecom Lempel Ziv data compression technique to V.42 error correction. Most V.32, V.32 *bis*, and V.34-compliant modems come with V.42 or V.42 *bis* or MNP.
V.44	A compression technology originally developed for the satellite industry to maximize available bandwidth. Approved by ITU as an industry standard in mid-2000, V.44 can generally yield download speed improvements of 20% to 60% when compared to V.42 *bis*. In cases where data are highly compressible, a 200% improvement in download speeds can be expected. The actual amount of improvement is dynamic and depends on the data content.
V.90	Standard for 57,600-bps modems (commonly called “56K modems”) in which asymmetric data rates apply (i.e., the send and receive rates are different). Depending on telephone line conditions, upstream rates (send) are restricted to 33,600 bps, and downstream rates (receive) are restricted to 57,600 bps. V.90 modems are designed for connections that are digital at one end and have one digital-to-analog conversion.
V.92	A new 56K modem standard approved by ITU in mid-2000. V.92 offers a maximum upstream rate of 48,000 bps, faster connection time, and call-waiting. The 48-kbps upstream rate is an improvement over V.90’s 33.6-kbps rate, although achieving it might be as problematic as achieving a 56K downstream rate. The faster connection time feature will reduce the time it takes a modem to complete its handshaking process by one-half. The call-waiting feature will allow the modem to place an Internet connection on hold while you take another call on the same line. The total wait-time will be controlled by your ISP and can range from 0 to 16 minutes, or indefinitely.

The most widely used speeds (and their corresponding protocols) today are 14,400 bps (V.32 *bis*), 28,800 bps (V.34), 33,600 bps (V.34 with software enhancement), and 57,600 bps (V.90) (see Table 15.1). Note that the DCE-to-DCE rate is fixed because modems must agree on a specific modulation technique.

Although modems will always try to negotiate the fastest link possible, speeds are dependent on line quality. For example, if two V.34 compliant modems are trying to communicate with each other, they will try to negotiate a 33,600 bps connection. If this is not possible, the modems will then try to connect at the next highest rate the line can support. If the line quality deteriorates during an established connection, then the modems will cycle down ("fallback") to a lower speed. When line quality improves, the modems will fall forward to a higher speed. Modem speed is also limited by the slowest modem in the connection. Thus, if a V.34 modem is trying to establish a connection with a V.32 *bis* modem, the fastest link possible is 14,400 bps regardless of line quality.

8. I have heard that if you have a 28,800 bps modem, then you can attain transfer rates as high as 115,200 bps. Is this true, and if so, how is it accomplished?

Yes, this is true. It is accomplished with *compression*, which is another one of those parameters modems negotiate when a dialup connection is first initialized. Compression is a process that codes repetitive patterns within a data set. For example, if a text message contains the string "XXXXXX," then instead of using six bytes to represent the data (one for each character), a compression technique might code the string so that only two bytes are used to represent it, with one byte to identify the repetitive character and one byte to specify the number of times it is repeated. Compressed files can be sent at a faster rate than uncompressed files. For example, a 1 Mbyte uncompressed file transferred at 28,800 bps takes nearly 5 minutes, depending on the file transfer protocol used. This is calculated as follows: Assuming 8 bits per byte, 28,800 bits per second = 28,800/8 = 3600 bytes per second. Since 1 Kilobyte = 1000 bytes, then 1 Megabyte = 1000 × 1000 = 1,000,000 bytes. Thus, to transfer 1 Megabyte at 28,800 bps requires 1,000,000/3600 = 278 seconds, which is approximately 5 minutes. If this same file is compressed by a 4 to 1 ratio, then the time it takes to transmit the compressed file is one-quarter of the original time, which effectively quadruples the data transfer rate to 115,200 bps. Remember, though, compression is only achieved if there is redundancy in the data set. The three primary standards used for data compression involving modems is V.42 *bis*, V.44, and the Microcom Networking Protocol (MNP) Level 5. V.42 *bis* defines a technique that can generate a 4 to 1 data compression ratio, depending on the type of file being transmitted, and V.44, which is a relatively new ITU standard, provides a 6 to 1 compression ratio. MNP 5 uses a data compression algorithm that compresses data by a factor of 2 to 1. MNP was initially a proprietary protocol developed by Microcom Incorporated that became an industry standard for data compression and error control in the 1980s. A brief summary of the various MNP levels are given in Table 15-2.

To benefit from compression and the potentially higher effective throughput rates, the computer must be able to transmit with its connected modem at a rate that is equal to the possible speed that can be achieved by the compression ratio. This speed is called the *DTE-to-DCE rate* and is commonly referred to as the computer's "port speed" or "serial port rate." On PCs, serial ports are more commonly known as the *universal asynchronous*

TABLE 15.2 The MNP Protocols

Protocol	Description
MNP 1–4	Used for hardware error control. All four levels are incorporated into V.42.
MNP 5	Incorporates the first four levels of MNP. Also uses a data compression algorithm that compresses data by a factor of 2 to 1.
MNP 6	Supports both V.22 *bis* and V.29.
MNP 7	Same as MNP 5 except it employs a data compression algorithm that compresses data by a factor of 3 to 1.
MNP 8	Extension of MNP 7; lets half-duplex devices operate in full-duplex mode.
MNP 9	Used for a variety of different circuits.
MNP 10	Used in cellular modems and poor line quality settings.

receiver/transmitter, or *UART.* DTE stands for *data terminal equipment*. In a dialup connection, the computer is the DTE device. Thus, the DTE-to-DCE rate is how fast the computer can talk to its modem, and the *DCE-to-DCE rate* is how fast the two modems can talk to each other. The DCE-to-DCE rate is a function of the modem standard and cannot be changed. However, the DTE-to-DCE rate is user configurable through the modem's communications software. Thus, a computer connected to a V.34 modem, should maintain a port speed of 115,200 bps to benefit from a 4 to 1 compression ratio, and a port speed of 172,800 bps to benefit from a 6 to1 compression ratio. Similarly, a computer using a V.90 modem should maintain a port speed of 230,400 bps if using V.42 *bis* and 345,600 bps if using V.44.

9. Where does error control come into play?

Error control (see Chapter 5) refers to error detection and correction. Modem standards for error control are specified by V.42 and MNP levels 1 through 4. Note that V.42 includes MNP 1–4. V.42 also includes something called *link access procedure for modems* (LAPM), which uses cyclic redundancy check (CRC) and automatic repeat request (ARQ) for error control. CRC is used for error detection, and ARQ prevents the modem from accepting any more data until the defective frame has been retransmitted successfully. V.42's default is LAPM. Thus, if a connection is being initialized between two V.42 compliant modems, they will use LAPM for error control. If one of the modems is not V.42 compliant, then the modems will negotiate to use MNP 1–4.

10. Would you summarize all of this modem information for me?

Be glad to. Modems are DCE devices that convert between analog and digital signals. Modem standards, as defined by ITU-T, specify modem speeds (called the DCE-to-DCE rate), the type of error correction methods they will use (V.42), and the data compression technique (V.42 *bis* or V.44). Given redundancy in the data set, the effective throughput of a modem connection can be doubled, tripled, or even quadrupled, depending on the compression technique used. To take advantage of this increase in effective throughput, though, the speed at which the PC talks to the modem (called the DTE-to-DCE rate) must match the potential compression ratio. Thus, if the DCE-to-DCE rate is 57,600 bps, then the DTE-to-DCE rate must be 230,400 bps to benefit from a 4 to 1 compression rate.

11. You really cleared up a lot of things for me. Let's talk about 56K modems for a moment. How different are they from other modems like V.34 and V.32 *bis*?

A 56K modem is a hybrid modem that involves a path consisting of both analog and digital connections. To understand this, let's consider a standard analog modem. According to Shannon's limit (see Chapter 4), the highest possible speed a modem can achieve over an analog line is somewhere between 33,600 bps and 38,400 bps, depending on line conditions. Analog modems also require four conversions in each direction. At the sending side, a digital to analog conversion converts digital data from the PC to analog form for transmission over the phone line to the telephone company's nearest switching station. Once there, the signal is then converted back to digital form where it is transmitted across the telephone company's digital network. When the signal reaches the switching station that serves the destination site, it is reconverted back to analog form and carried to the destination site. Finally, the modem at the destination site demodulates the signal from analog to digital form before it is passed to the receiving computer. This conversion process is illustrated in Figure 15.2.

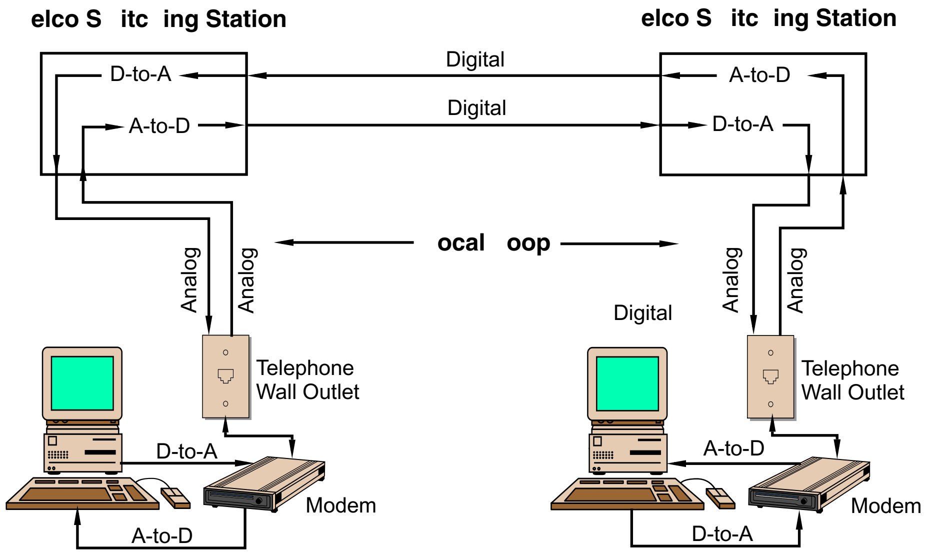


FIGURE 15.2 A typical analog dialup connection between two nodes involves four analog–digital conversions in each direction. (A-to-D = analog-to-digital conversion; D-to-A = digital-to-analog conversion.)

12. Time out. Why are there so many conversions? I thought the telephone company's network is completely digital.

It is. However, unless an end-to-end digital circuit is installed (as is the case with ISDN), the circuit between a home and the telephone company's nearest switching station is analog. This circuit is known by many different names. Formally, it is called the *digital subscriber loop* but it is more commonly referred to as the *local loop*. It has also become figuratively known as "the last mile/first mile" because, depending on one's perspective, it is either the last part of the network data must traverse to reach a customer's site or it is the first part of the network data must traverse to access the telco's switch. The local loop is predominately copper-based and represents the bottleneck to high-speed, home-based dialup networking. Given the cost constraints involved in rewiring local loops, the focus of resolving the bottleneck issue has been addressed by technology. One technology currently being deployed is *x*DSL service; a second is cable modems. Each is discussed later in the chapter.

13. How many conversions are needed with a 56K modem?

In contrast to a standard analog modem, a 56K modem only requires two conversions in each direction. Both conversions occur only at the local site. This is because the path is analog from the client site to the central office switching station, but it is completely digital from the switching station to the server site (see Figure 15.3). Thus, a digital to analog conversion first converts digital data from the PC to analog form for local loop transmission, and then an analog to digital conversion converts the analog data back to digital form at the telco's local switching station. From here, the remaining part of the journey is digital. Thus, a 56K modem involves a hybrid analog-digital path. The path is analog from the client site to the central office switching station, but the path from the switching station to the server site is completely digital. This hybrid analog–digital circuit restricts "upstream" transmission rates (i.e., from client to server) to 33,600 bps for V.90 modems and 48,000 bps for the new V.92 modems (see Table 15.1). "Downstream" rates (i.e., from server to client) can be as high as 57,600 bps depending on line conditions. Note that in the United States, an FCC regulation limits the highest data rate to a little less than 56K.

14. I have heard people talk about x2 and K56flex. What are these?

Prior to standardization, two early, competing, and incompatible proprietary 56K modem standards were developed: x2 from USRobotics (3Com) and K56flex from Lucent Technologies. Both standards have since been absorbed by the de jure V.90 ITU-T standard, which was recently upgraded to V.92. Thus, as long as both ends of a transmission link have V.90- or V.92-compliant 56K modems, you do not have to be concerned with x2 or 56Kflex. If this is a concern because the type of modem at the remote end is unknown, some 56K modems employ all four technologies: V.92, V.90, x2, and 56Kflex. This way, when the modems begin their handshaking process, if the remote modem uses one of the proprietary 56K standards, then the local modem will cycle through and connect with a compatible standard. You should note that 56K modems cannot be used for global connectivity because the signaling schemes used on international connections reduce speeds to a maximum of 33.6 kbps.

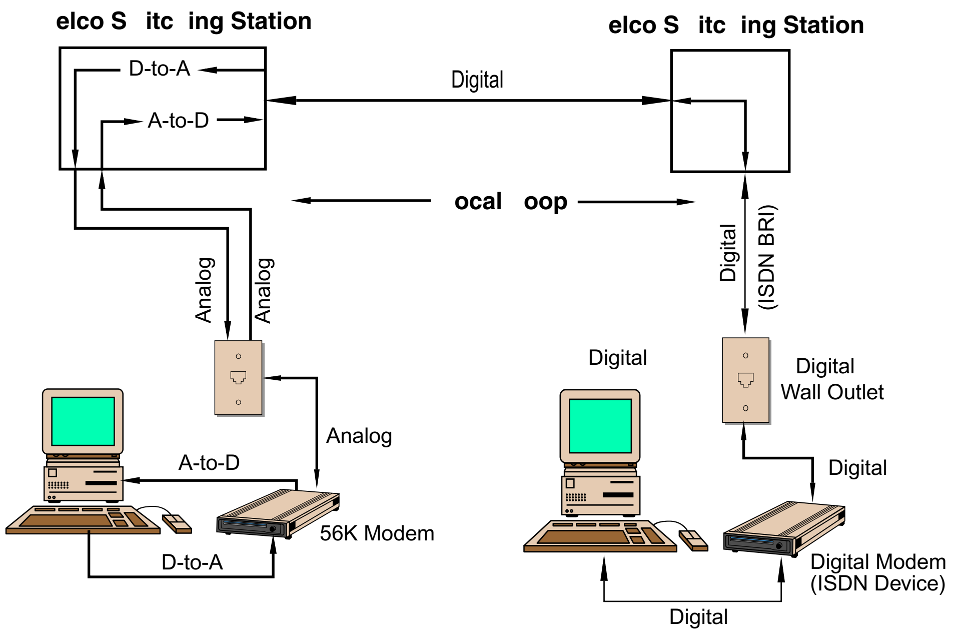


FIGURE 15.3 A 56K modem connection involves a hybrid analog–digital path. Unlike traditional analog modems, a 56K modem requires only two analog–digital conversions in each direction.

15. Let's move on to *x*DSL service. What is it?

DSL, which stands for *digital subscriber line*, is a technology that enables data, voice, and video to be mixed and carried over standard analog, (copper) telephone lines. This is accomplished by using the unused frequencies that are available on a telephone line. DSL can deliver data services without interfering with voice transmissions. Since DSL supports digital data transmissions, there is no need for any analog-to-digital or digital-to-analog conversions, as is the case with conventional dialup lines that employ analog modems. Furthermore, a DSL signal can be partitioned so that part of the channel's capacity is used for analog voice transmissions concurrently with higher-frequency data transmissions.

DSL is an "always-connected" or "always-on" technology. This means that unlike ISDN and conventional analog modem service, which are circuit-switched, you do not have to dial up a service provider to establish a connection. Instead, a DSL connection apes a leased line connection, and hence, the service is always available for transmitting or receiving data. Furthermore, DSL connections are dedicated point-to-point links. This is in contrast to cable modem service, which is bus-based and involves multiple subscribers contending for a channel's bandwidth. Since the local loop's bandwidth is not shared, DSL offers better security and dedicated bandwidth between the telco's switching station and a customer's premises.

Another feature of DSL is that it keeps data traffic off the voice network. With an analog modem service, dialup users use the public switched telephone network (PSTN) to call

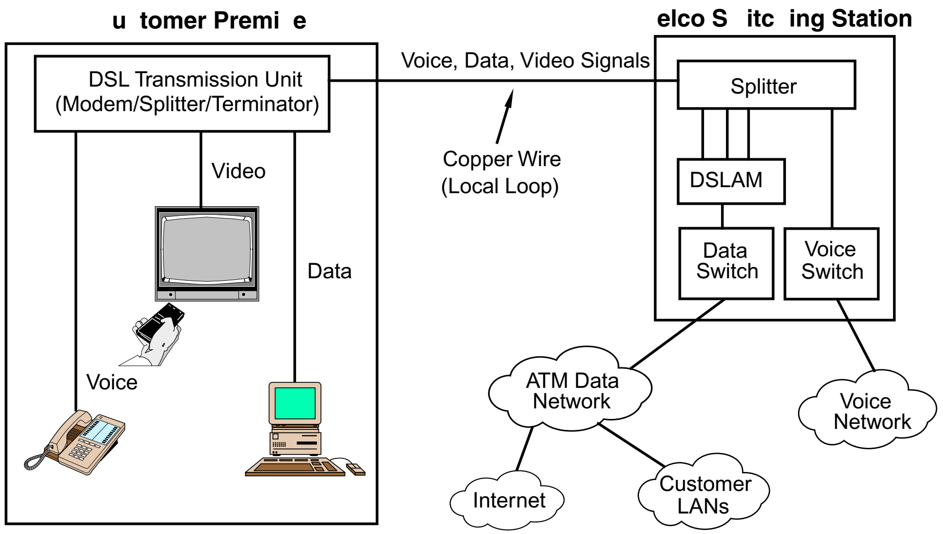


FIGURE 15.4 Generic illustration of a DSL-based connection. Voice, data, and video signals from the customer's premises are split into separate signals and transmitted across the local loop to the provider's site. Voice signals are directed to the telco's voice switch for transmission through the public switched telephone network (PSTN). Data and video signals are aggregated by the DSL access multiplexer (DSLAM) and directed to the telco's data switch for transmission through the telco's data network. Source: Adapted from Aber, 1997.

up their Internet service provider to gain access to the Internet. This ties up a port on the telco's voice switch, which reduces the number of voice ports available for voice calls. With DSL service, voice calls (i.e., signals) are segregated from data traffic via a line splitter and directed to the telco's voice switch for transmission across the PSTN. Data signals, however, are aggregated via a DSL access multiplexer (DSLAM), which feeds directly into a data switch for transmission across the telco's data network backbone. Thus, data signals bypass the voice switch, thereby freeing up ports for voice calls (Figure 15.4).

DSL is commonly written with a leading *x* (e.g., *x*DSL) to collectively represent the family of digital subscriber line technologies. The leading *x* is a variable that specifies a descriptor, which, when combined with DSL, defines the different DSL variants. Presently, there are nine variants, which range alphabetically from asymmetric DSL (ADSL) to very-high data rate DSL (VDSL). A summary of these variants is provided in Table 15.3. Many regard DSL technology as the telephone companies' answer to cable modems in competing for "the last mile."

16. I know you provide a summary of the DSL variants in Table 15.3, but I was hoping you could discuss them in a little more detail.

OK. A brief description of each follows.

TABLE 15.3 The Digital Subscriber Line (DSL) Variants

Type	Name	Description
ADSL	Asymmetric DSL	ADSL is a high-bandwidth local loop technology that provides higher data transmission rates from the service provider to the customer (downstream) than from the customer to the service provider (upstream). Downstream rates range from 1.5 to 6 Mbps, and are provided in T1 increments. Upstream transmissions are provided over a bidirectional channel with rates ranging from 64 to 640 kbps, which are provisioned in 64-kbps increments. A separate 4-kHz channel is also partitioned from the link to provide basic analog telephone service. Data transmission rates depend on line quality and local loop distance. Maximum local loop distance is approximately 10,000 feet (2 miles or 3 km). ADSL is suitable for Internet or intranet access, video-on-demand, 3-D images, database access, and remote LAN access. ADSL is an ANSI standard (T1.413).
ADSL Lite	ADSL Lite, G.Lite, Universal ADSL, Splitterless ADSL	Known by many names, this DSL variant is a slower version of ADSL that does not require a line splitter at the customer site. Instead, the ADSL link is split for the customer remotely at the service provider's site. Downstream data rates "max out" at 1.5 Mbps, and upstream rates are 384 kbps or less. Maximum local loop length is 18,000 feet, or about 3.5 miles. ADSL Lite is an official ITU-T standard (G-992.2), and its use is intended primarily for residential homes.
HDSL	High-Bit-Rate DSL	HDSL is a symmetrical DSL service that delivers T1 rates over two pairs of UTP and E-1 rates over three pairs of UTP. HDSL provides a more cost-effective strategy for provisioning T1-based local loops over existing UTP cable by separating the cable pairs into two full-duplex channels with each pair operating at 784 kbps. (In contrast, T1 cable pairs are simplex; one pair is used for downstream and the other for upstream transmissions, and each operates at T1 speed.) This provides less signal loss and less electromagnetic radiation, which result in distances twice that of T1 circuits in which a signal can travel without the need for repeaters. Maximum local loop distance is 12,000 feet (3.5 km). Applications include connecting PBXs and serving as an alternative to T1/E-1. It is also suitable for campus networks and ISPs.
HDSL2	High-Bit-Rate DSL 2	HDSL2 is second-generation HDSL that supports T1 transmission rates over a single UTP cable pair but still maintains a maximum distance length of 12,000 feet.
IDSL	ISDN-like DSL	IDSL is a symmetrical DSL service that delivers a maximum data rate of 144 kbps each way. It is based on ISDN technologies and uses ISDN BRI service. Unlike ISDN, which is circuit-switched, IDSL is an always-on service. However, also unlike ISDN, which supports voice, data, and video, IDSL supports only data transmissions. The maximum local loop distance for IDSL is 18,000 feet (5.4 km).
RADSL	Rate-adaptive DSL	RADSL is a nonstandard ADSL technology that determines the rate at which a signal can be transmitted across the local loop and then automatically adjusts this rate accordingly. Various proprietary solutions support downstream rates ranging from 640 kbps to 2.2 Mbps and upstream rates ranging from 120 kbps to 1 Mbps. Maximum distances range from 15,000 feet to 18,000 feet depending if 24- or 26-gauge copper is used for the local loop.

TABLE 15.3 The Digital Subscriber Line (DSL) Variants (continued)

Type	Name	Description
SDSL	Symmetric DSL Single-Line HDSL	SDSL is an HDSL variant that uses a single wire-pair to support maximum transmission rates ranging from 384 to 768 kbps. Maximum local loop distances range from 12,000 feet to 18,000 feet. Telephone service is not supported.
UDSL	Universal DSL	UDSL is a proposed standard for the European community that will deliver symmetrical DSL service at 2 Mbps.
VDSL	Very High Data Rate DSL	VDSL is an asymmetric DSL service that is currently being defined. Transmission rates are expected to range from 13 to 52 Mbps downstream, with upstream rates ranging anywhere from 1.5 Mbps to an upstream rate equivalent. Maximum distances will range from 1000 feet (300 m) to 4000 feet (1000 m). VDSL will most likely be deployed as part of a fiber-copper hybrid local loop in which fiber is used for the service provider-neighborhood link and VDSL over UTP is used for the neighborhood-residential home link. Applications include Internet or intranet access, video-on-demand, database access, remote LAN access, and high-definition television (HDTV).

Asymmetric DSL (ADSL) ADSL is an ANSI standard that was initially developed in 1987 by Bell Communications Research (Bellcore, which changed its name to Telcordia Technologies in 1999) to deliver video-on-demand and interactive TV over UTP-based local loops. The term *asymmetric* means that more bandwidth is reserved for downstream transmissions (i.e., from service provider to customer) than for upstream connections (i.e., from customer to provider). ADSL connections require an ADSL modem at the local end and at the telco switching office. Note that the term *modem* used in this context is really misapplied because ADSL is a digital technology. Hence, there is no modulation or demodulation of analog and digital signals. Nevertheless, modem is accepted and understood by the general public as any device that provides a dialup connection. ADSL service also needs to be purchased from a local phone service provider. Service might be available directly from an Internet service provider as well. Since ADSL operates over the same copper wire currently used for phone service, a second line or a new termination unit (which is required for ISDN) does not have to be installed. To use the service, you essentially connect the ADSL modem to your computer (or network) and telephone line (Figure 15.5). It is quite similar to using an analog modem except you can now transmit different types of data (voice, data, and video) simultaneously over the same circuit. ADSL is also powered like conventional telephone service so during a power outage voice service will still be available. Other ADSL-connected devices such as a computer, however, still require power to run.

The biggest problem with an ADSL connection is local loop distance restrictions. To get an idea of what kind of data rate to expect, we need an accurate measurement of the local loop distance. A second potential issue is the quality of copper on which the local loop circuit is based. Both issues are discussed later. A third potential problem is interoperability of ADSL equipment from different manufacturers. However, this is not as much of a problem as it was before ADSL became an ANSI standard. A fourth potential

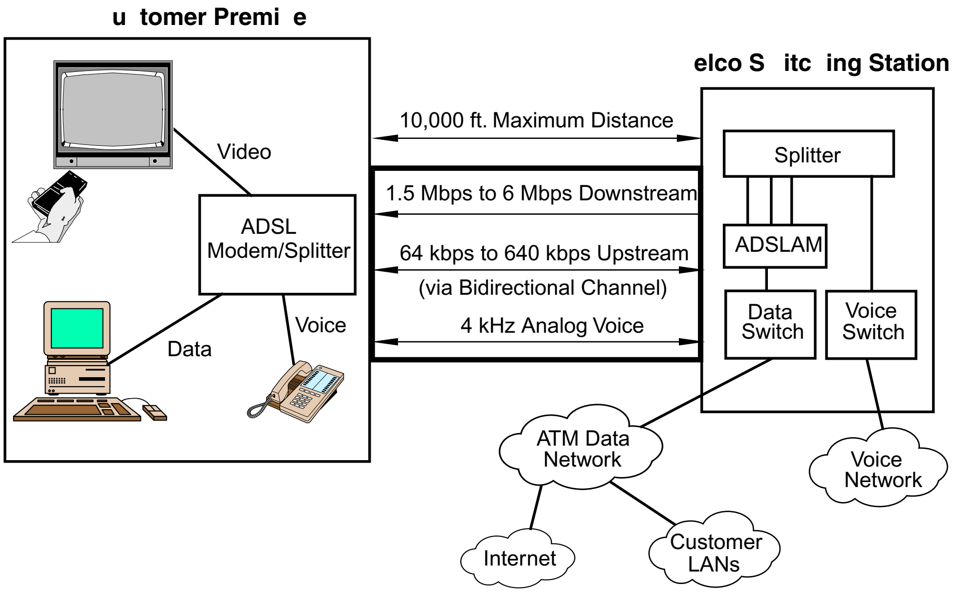


FIGURE 15.5 An ADSL connection involves ADSL modems at the local site and a "splitter" and ADSL access multiplexer (ADSLAM), which incorporates an ADSL modem, at the telco switching station. (The splitter can also be incorporated within the ADSLAM device along with the ADSL modem.) The splitter is used to separate voice and data signals. ADSL's maximum local loop distance is approximately 2 miles, its downstream rates range from 1.5 to 6 Mbps, and its upstream rates range from 64 to 640 kbps, which is provided via a bidirectional channel. A 4-kHz channel is also partitioned from the DSL link for conventional analog voice service. Source: Adapted from Aber, 1997.

issue is service availability. Deployment of ADSL service is not widespread yet. Consequently, many telco carriers might not offer it or might not have knowledgeable and skilled people onboard yet to help design or implement an ADSL-based connection. One final potential problem is relieving the headache you are sure to get trying to sort through all the rhetoric and hyperbole about DSL-service, cable modem service, ISDN, and 56K modem service.

ADSL Lite Commonly known as G.Lite, ADSL Lite is an ITU-T standard that is similar to ADSL except it does not require a line splitter at the customer site. Instead, the link is split for the customer at the service provider's site (Figure 15.6). The maximum downstream rate ranges up to 1.5 Mbps, and upstream rates are 384 kbps or less. Maximum local loop distance is 18,000 feet. Of all the DSL variants, G.Lite is the most popular among residential home subscribers.

High-Bit-Rate DSL (HDSL) HDSL is the most established of the DSL technologies. Unlike ADSL or ADSL Lite, HDSL service is symmetrical, providing T1 transmission rates over two pairs of UTP and E-1 rates over three pairs of UTP. HDSL was developed as an alternative to T1/E-1 service, and it provides a more cost-effective strategy for provisioning T1-based local loops over existing UTP cable. Unlike T1 service, which provides

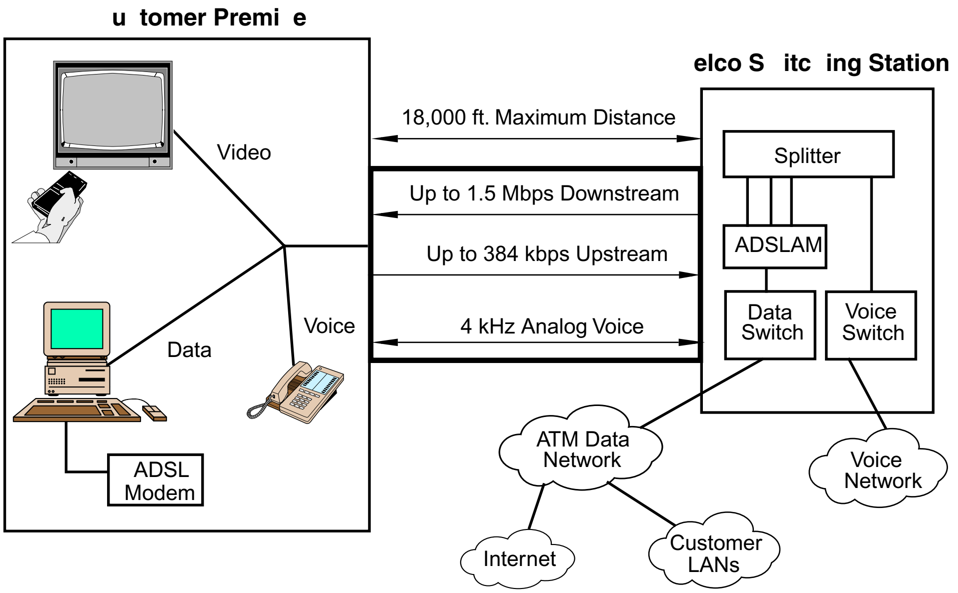


FIGURE 15.6 A G.Lite DSL connection is similar to that of ADSL except it is "splitterless." Here, high-speed data communications are provided by an ADSL modem connected to a PC, and the "splitting" of signals is performed at the service provider's end and not at the customer site. The maximum local loop length is 18,000 ft. (3.5 miles), upstream rates range up to 384 kbps, and downstream rates range up to T1 speeds.

T1 data rates over simplex UTP cable pairs, HDSL separates the cable pairs into two full-duplex channels with each pair operating at 784 kbps. HDSL's maximum local loop distance is 12,000 feet, which is twice that of T1. The increased loop distance is a direct result of HDSL's technology, which provides less signal loss and less electromagnetic radiation.

HDSL2 This is second-generation HDSL, which provides the same performance as HDSL, but it does so over a single UTP cable pair.

ISDN DSL (IDSL) IDSL represents a hybrid ISDN/DSL technology. It is a symmetrical DSL service that delivers a maximum data rate of 144 kbps each way that uses ISDN BRI service. Unlike ISDN, which is circuit-switched, IDSL is an always-on service. However, also unlike ISDN, which supports voice, data, and video, IDSL supports only data transmissions. The maximum local loop distance for IDSL is 18,000 feet.

Rate-Adaptive DSL (RADSL) RADSL is a nonstandard ADSL technology that determines the rate at which a signal can be transmitted across the local loop and then automatically adjusts this rate accordingly. Various proprietary solutions support downstream rates ranging from 640 kbps to 2.2 Mbps and upstream rates ranging from 120 kbps to 1 Mbps. Maximum distances range from 15,000 to 18,000 feet, depending on whether 24- or 26-gauge copper is used for the local loop.

Symmetric DSL (SDSL) Also called single-line HDSL, SDSL provides symmetric data transmission rates over a single wire pair to support maximum transmission rates ranging from 384 to 768 kbps. Maximum local loop distances range from 12,000 to 18,000 feet. Telephone service is not supported.

Universal DSL (UDSL) UDSL is a proposed standard for the European community that will deliver symmetrical DSL service at 2 Mbps.

Very-High-Data-Rate DSL (VDSL) VDSL is an asymmetric DSL service that is expected to provide downstream data transmission rates ranging from 13 to 52 Mbps and upstream rates ranging from 1.5 Mbps to an upstream rate equivalent. Maximum distances will range from 1000 to 4000 feet (300 to 1000 m). VDSL will most likely be deployed as part of a fiber-copper hybrid local loop in which fiber is used for the service provider–neighborhood link, and VDSL over UTP is used for the neighborhood–residential home link.

17. I see that you keep mentioning local loop distance restrictions. Please elaborate.

OK. DSL transmission rates are a function of line quality and the distance a customer's site is from the local switching station. The longer the local loop distance, the lower the rate; likewise, the poorer the quality of the local loop circuit, the lower the rate. DSL should be implemented with quality copper circuits at the local loop because of possible line attenuation and EMI problems (see Chapter 4), both of which can degrade data transmission. Also keep in mind that DSL rates represent the speed at which data are delivered across the local loop. If there is congestion at the switching station, or if there is congestion at some point within the network cloud, it doesn't matter how fast your connection is because a bottleneck exists somewhere beyond your loop. Something else to be cognizant of is the DTE-to-DCE rate, which we discussed earlier. Most current model PCs or Macintoshes are not equipped with serial ports that have the capacity to keep up with a DSL connection. The bottom line is, DSL data rates are certainly impressive. However, these are the *theoretical maximum rates possible*, which do not reflect real world constraints.

18. OK. We've discussed analog modem, 56K modem, and *x*DSL network connections. This leaves cable modems and cable-based network connections.

Don't forget that we also discussed satellite-based connections in Chapter 4 and ISDN connections in Chapter 11. You might want to review these two connectivity methods so that you can compare them to the methods we describe in this chapter.

Cable modem service represents the cable companies' infiltration into the telecommunications business courtesy of the Telecommunications Act of 1996. Cable modems use cable television lines for data communications. These lines use broadband coaxial cable (see Chapter 4), which has a multitude of frequencies available and significantly higher bandwidth than the UTP cable used by the telcos. Cable operators specify a frequency range within the cable and run data over it. Cable modems provide the interface (i.e., connection point) between a computer and the cable (Figure 15.7). Specifically, this interface is an Ethernet/802.3 network interface card. Thus, a PC must have an Ethernet/802.3 NIC installed to be connected to a cable modem. Once connected, it is as if the PC were connected to an Ethernet/802.3 LAN. The connection is always "up," and multimegabit data rates are possible.

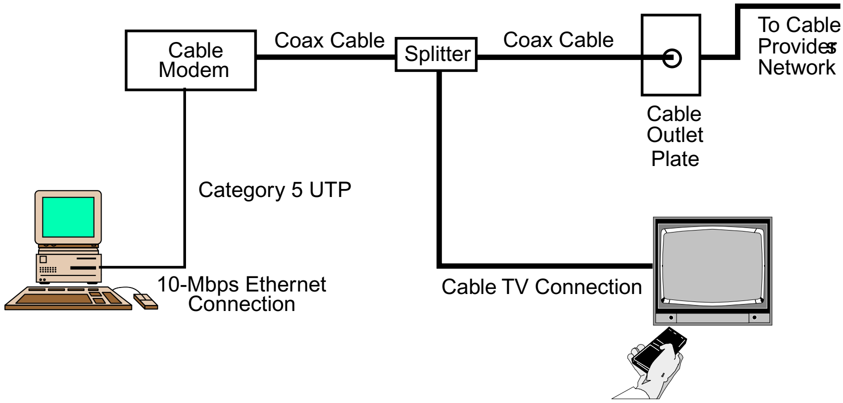


FIGURE 15.7 The cable television industry is now able to provide homes with conventional television viewing and two-way data communications (i.e., Internet access) concurrently via a standard cable TV connection and a cable modem. Internet data are carried by the cable on an unused frequency just as if it were a television channel. A splitter is used to partition the data channel from the television channels. The modem serves as an interface between the computer and the coaxial cable. The computer is connected to the modem via Category 5 UTP cable. This link provides a dedicated 10 Mbps Ethernet connection between the computer and the modem.

19. How do cable operators provide two-way data communications?

To answer this question, we first need to examine a typical cable TV network. Cable networks originally were designed specifically for downstream transmission only—they serve as a one-way broadcast facility for transmitting television signals. They were never designed for two-way transmissions. (*Note:* CATV originally stood for Community Antenna Television, which was designed to provide television broadcasts in areas where broadcast reception was poor.) The original network design is a tree topology that consists of a signal source (called the head end), a local neighborhood distribution site, and a collection of amplifiers. The head end aggregates signals from different locations and modulates them to the frequencies that are assigned to the destination sites. (The head end usually receives a master signal from a satellite dish or a fiber backbone.) These modulated signals are then piped along the cable to a neighborhood distribution site, where they are delivered directly to a customer's home. En route from the head end to the neighborhood distribution facility, and then again to the customer's site, the signal is regenerated via one-way amplifiers. (This is similar to the use of repeaters in an Ethernet/802.3 network.) This is necessary to maintain signal quality while increasing cable distances. A simple diagram illustrating this design is shown in Figure 15.8.

To provide for two-way transmission, cable networks had to be upgraded. Upgrade plans included replacing the one-way amplifiers with two-way amplifiers and replacing the coaxial cable with fiber-optic cable. The cable plant upgrade is occurring in two phases. The first phase involves replacing the coaxial cable with fiber at the head end. This eliminates the need for amplifiers but still allows the network diameter to increase without signal degradation. The fiber-optic cable also will carry more data. This resulting network

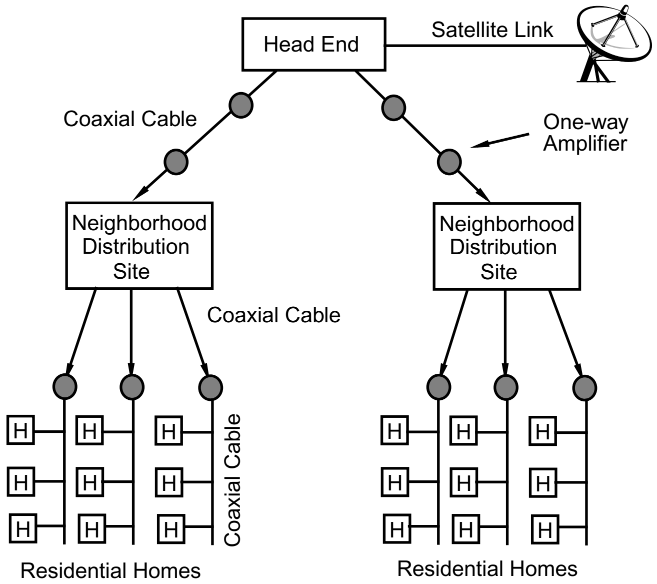


FIGURE 15.8 A coaxial-based cable television network is treelike in nature and designed for one-way transmission. Amplifiers are used to maintain signal quality while increasing distances. Source: Adapted from Fitzgerald, 1996.

is called a *hybrid fiber-coaxial* (HFC) network. The second phase involves replacing the coaxial cable at the neighborhood distribution site. Once fiber is deployed to the neighborhood, optical-to-electrical converters can replace the amplifiers, resulting in less amplification and a more robust cable plant (Figures 15.9 and 15.10). Note from Figure 15.8, 15.9, and 15.10 that connections to the home are still coaxial. The reason for this is because it is cost-prohibitive to retrofit every home with fiber-optic cable.

20. OK. So the cable operator's network has been retrofitted for two-way communications. This makes sense. What I don't understand, though, is how the cable itself supports two-way data communications.

Cable TV systems were initially designed to deliver conventional analog broadcast television signals directly to standard television sets. To do this, cable operators replicate that part of the RF spectrum used for television broadcasting. Thus, instead of broadcasting radio signals over the air, cable television operators deliver these signals via coaxial cable. To support two-way data communications, cable TV operators allocate two unused television channels. One channel is used for downstream traffic and a second channel is used for upstream traffic. These channels have exactly the same bandwidth as the television channels, namely 6 MHz. Downstream channels typically operate within the 50–750 MHz range; upstream channels usually operate in the 5–42 MHz range.

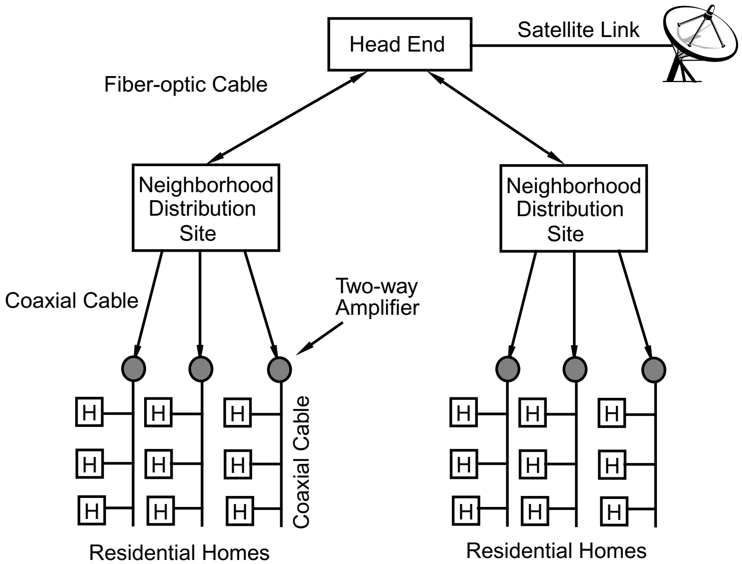


FIGURE 15.9 A hybrid fiber-coaxial (HFC) cable network consists of fiber-optic cable between the head end and neighborhood distribution sites. Coaxial cable is still deployed to the neighborhood, however. Source: Adapted from Fitzgerald, 1996.

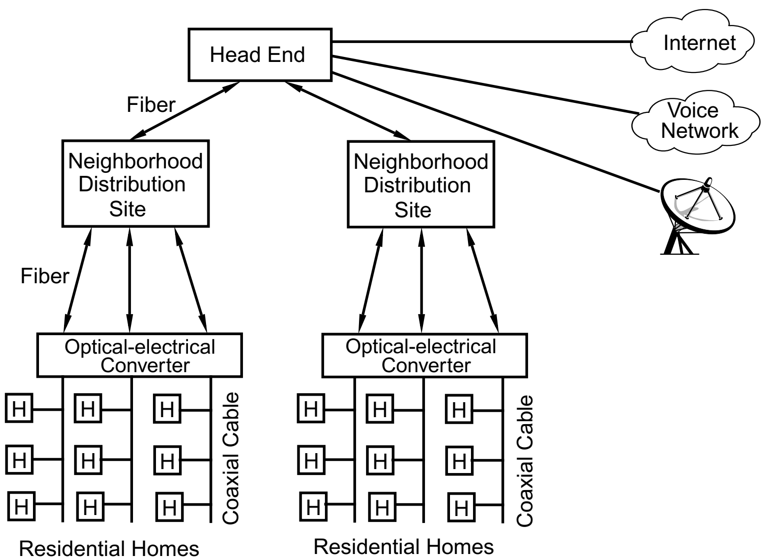


FIGURE 15.10 When completed, the upgraded cable network will be able to provide high-speed two-way transmissions for data, voice, and video service. Source: Adapted from Fitzgerald, 1996.

21. I am still looking for a response that describes how all of this works.

OK. Let's see if we can put this all together for you. Our perspective will be downstream, from the head end to the home. Thus, the following description is applicable to receiving data from the Internet. As stated earlier, cable operators have emplaced a hybrid fiber/coax (HFC) plant as shown in Figure 15.9. This plant consists of regional head ends, neighborhood distribution sites, optical-electrical converters, and coaxial cables. Regional head ends generally support 200,000 to 400,000 homes and have high-speed data connections to the Internet as well as to the public switched telephone network (PSTN). Head ends are linked to neighborhood distribution sites via fiber-optic cable; links are typically 622-Mbps (OC-12) channels. At neighborhood distribution sites (Figure 15.10), digital signals are modulated by the *cable modem termination system* (CMTS) to analog form where they are then transmitted via fiber-optic cable to optical-electrical converter nodes. Neighborhood distribution sites generally serve 20,000 to 40,000 homes and optical-electrical converter nodes serve between 500 and 1000 homes. The CMTS provides separate upstream and downstream channels. Upstream channel bandwidth ranges between 2 and 10 Mbps; downstream channel bandwidth is 27 Mbps. Finally, a splitter at the side of a home segregates the broadcast television channels from the cable modem channels. The cable modem demodulates the signal back to digital form where it is then processed by the computer. The link between the computer and the cable modem is Category 5 UTP cable. The cable connects to the computer via a RJ-45 connector, which is part of a standard 10-Mbps Ethernet/802.3 network interface card that resides in the computer (Figure 15.7). Newer cable modems support Universal Serial Bus (USB) connections and provide data encryption.

22. What are the limitations of the hybrid fiber-coax network?

Until two-way amplifiers are installed, cable operators can only offer high-speed downstream (from head end to customer site) data transmissions. Upstream transmissions are restricted to 33.6 kbps via an analog modem dialup connection through the telephone network. Even those areas where two-way amplifiers are in place, because coax is still deployed to the neighborhood, upstream transmission rates are considerably less than their downstream counterparts. Depending on the cable operator and service, current upstream rates are somewhere between 500 kbps to 3 Mbps; downstream rates range between 10 to 30 Mbps.

23. What's wrong with those numbers? It's a lot better than 56K modem service.

True. However, remember that the cable network is still a shared system. It is very similar to 10-Mbps Ethernet/802.3, which is a broadcast technology. Thus, as more subscribers come on-line, there will be more contention for the available bandwidth. (Contrast this with ADSL, which is a point-to-point topology. See also Chapter 8 for more information about Ethernet/802.3 LANs.)

24. Which is the best method for establishing a home-based Internet connection?

Let's first review the various connection methods from the perspective of establishing an Internet connection. Following is a brief summary and comparison of some of the most common ones used to connect a PC to the Internet from home.

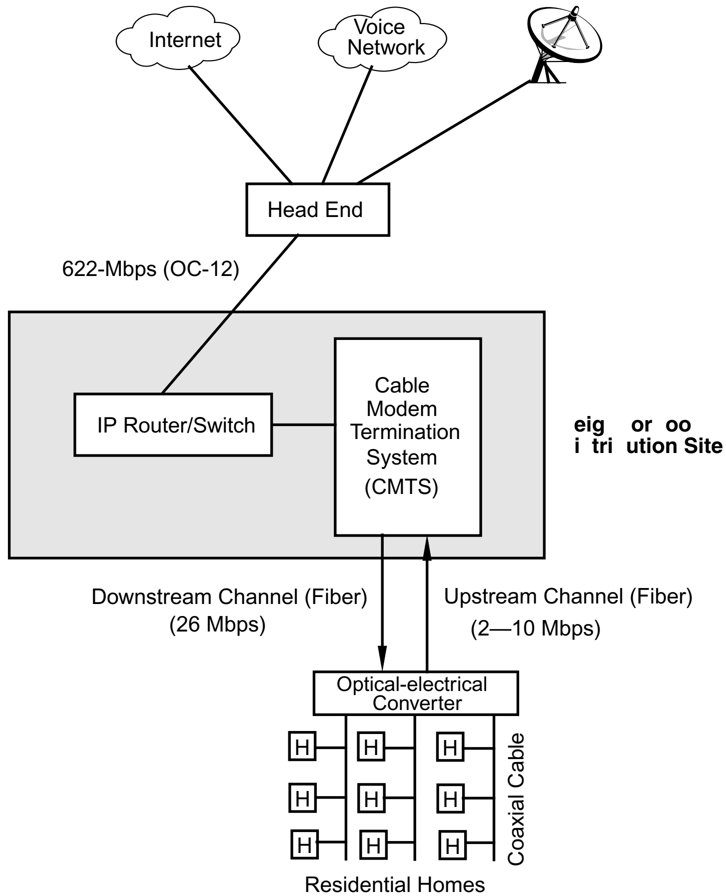


FIGURE 15.11 Neighborhood distribution sites serve as interchange points between the cable provider's regional fiber network and its coaxial cable plant.

Analog Modem This is by far the most common and most popular method. A modem is connected via a phone line to the ISP, and you can get access to the Internet at whatever speeds the modem and ISP support. At this writing, the most popular modem for sale is the ITU-TSS standardized V.90 56-kbps modem or the dual-56-kbps load-balanced modem. This second type of modem provides a dialup via two phone lines and load balances between them, yielding a throughput of about 112 kbps if everything goes as planned. In any case, a modem, phone line, access software (e.g., PPP), and ISP account are required for connectivity. Analog modem connections are both relatively easy and inexpensive to implement. There are also nearly no equipment compatibility issues to be concerned with either. However, analog modems have reached their technical upper limit at 56K and hence offer the slowest Internet access method when compared to the other services available.

ISDN ISDN provides a faster connection method for Internet access than analog modems. Of course, you pay more for it. There are two connection modes: dialup and permanent ISDN. To make an ISDN connection work, an ISDN-compatible phone line and an ISDN terminating device (commonly, but incorrectly, called an ISDN modem) are required with software and an ISP connection account. (See Chapter 11 for more information about ISDN.) ISDN is a viable alternative for relatively high-speed Internet connections, particularly if no other high-speed access is available within your area. ISDN works over copper phone wires, provides multiple channels that can be combined for higher transmission rates, supports voice, video, and data transmissions, and can be implemented in an always-on mode with always-on/dynamic ISDN.

Cable Modem Cable modems offer high-bandwidth Internet access over standard coaxial cable TV lines. In this scenario, the local cable company installs a cable set-top box that has the ability to connect to your PC. Your PC will "think" it's connected to an Ethernet/802.3 LAN. An Ethernet/802.3 NIC is required. The set-top box provides a broadband access capability to the cable company, and they in turn have a router connection somewhere to the Internet. A cable modem provides an always-on connection; thus, you will be "live" on the Internet 24 hours a day, 7 days a week, just like a corporate Internet connection. Except for the PC, the cable vendor will usually provide everything needed to establish a cable modem connection (e.g., cable modem, software, Ethernet/802.3 card, and cables). Although cable modems and DSL service both provide high-speed Internet access, the method in which they provide access is quite different. For example, cable modem subscribers share bandwidth and hence must contend for access to the channel. DSL subscribers, on the other hand, have a dedicated point-to-point local loop link. Thus, during periods of peak activity, cable modem subscribers will see a degradation in overall throughput. Shared access is also not as secure as a dedicated point-to-point link. Some cable modem providers also restrict video access via a cable modem connection and instead require their subscribers to view video via their cable TV service. Finally, cable modem connections do not provide dialtone service (yet); hence, a separate telephone circuit is still needed for conventional voice communications.

Asymmetric Digital Subscriber Loop (ADSL) ADSL in general, and ADSL Lite in particular, are two DSL variants that are applicable for residential homes. ADSL is similar to a leased line connection and, as with cable modem service, provides an always-on connection. ADSL service is asymmetric, meaning that upstream and downstream data transmission rates are different. (See Table 15.3 for more information about ADSL service.) Normally, the ADSL service provider will provide all the necessary hardware and software (except the PC) for an ADSL connection. Advantages ADSL has over other high-speed Internet connectivity strategies include high data transmission rates, its relatively lower cost, and the ability to use the existing UTP-based local loop for voice, video, and data transmissions across a single line.

Satellite-Based A satellite-based Internet strategy involves installing special base stations in a local neighborhood. These stations have a wireless relay to a small dish or flat data collector unit attached to a home. A wire from the dish goes to a relay system in the home, and the PC is connected to the network similar to an Ethernet/802.3 LAN (a NIC in the PC is required). Some commercial systems provide up to 10-Mbps symmetric connectivity to your home and do not depend on wire from the telephone company or a cable provider. The vendor provides the connection hardware and software. Requirements include service in the area, a dish on the home, a decoder system, cables, an Ethernet NIC in the PC, software, and an ISP account. The most widely available service is currently Direct Broadcast Satellite (DBS), which is also known as Digital Satellite Service (DSS). Download speeds are approximately 350 kbps. Don't forget, though, that satellite communication services have a pronounced propagation delay (see Chapter 4).

25. So which one is best?

Often, it's not so much a problem of "which is best" as much as "what is available." Telephone lines and modems are ubiquitous, and thus easy to install and configure. The other four described methods vary in availability depending on where you live. Thus, some research will need to be done. We have used all five methods and find that they work okay once they are mature (the technology, not the authors—we never claimed to be mature). The biggest differentials are the cost per month for service, reliability issues, and one-time capital equipment costs to implement the service. You will need to shop around for what works best for you.

26. What security measures do I need to take to protect my Internet connection?

A home-based Internet connection is just as vulnerable to attack as a corporate connection. This is especially the case with "always on" Internet connections such as cable modem, residential regional wireless links, ISDN permanent connections, and most ADSL connections. All the protocols work the same whether you are a large company or a one-bedroom apartment location, and all locations can be compromised by an unauthorized user. Nothing can stop an attack from happening.

Major threats to home computers that are connected to the Internet include viruses, worms, and Trojan horses. Viruses and worms usually attach themselves to e-mail messages and often destroy files residing on your hard drive. An example is the "ILOVEYOU" e-mail worm that began appearing in mid-2000 and affected approximately 45 million computers worldwide. Trojan horses consist of a set of instructions that are placed into an application program and instruct the program to deviate from its original intention. Thus, a Trojan horse masquerades as a "regular" program that triggers a malevolent activity.

The best you can do is put in safeguards to keep someone from getting to your systems and your files. Protecting home computers from online threats involve using common sense with heightened awareness and installing protective software or hardware. Commonsense measures include not opening suspicious or unexpected e-mail attachments, turning file sharing and Web sharing functions off before going online, and being careful about giving out sensitive information such as passwords. Software and hardware protection involves installing and maintaining a current copy of an antivirus checking program, using encryption software for files and disk drives, installing a home firewall product, and using virtual private network (VPN) software to safely connect to corporate resources. Additional information about network security in general is provided in Chapter 16.

27. Do attacks on home-based Internet connections happen that often?

Yes and they are expected to increase as more home owners subscribe to "always on" Internet connections such as cable modem or *x*DSL service. One of the authors had a friend who was fortunate enough to be one of the first in his neighborhood to get a cable modem connection. He installed the proper hardware and software on the system, and life was good. In the first week of operation, he left his PC on at all times and was, effectively, hooked into the Internet 24 hours a day. In that timeframe, an unauthorized user got onto his system and set up an IRC chat redirection program that allowed the hacker to get onto chat groups, be as rude and obnoxious as desired, and seriously disturb the chat groups that the hacker was attacking. In the process of this, the various larger chat bulletin board systems traced the IP address back to the friend's PC; then a lawyer got involved. The friend first found out that something was amiss when he and his ISP received a "cease and desist" letter that threatened a lawsuit. Since the friend had been out of town that whole time, and since no one else had touched the system, it was investigation time. Examination of the system showed that the hacker really had been there and how he gained access. The software was deleted, the ISP helped explain things to the lawyer, and all went back to normal. So, in a home networking environment, you have to take security seriously and use the proper precautions just like the company environment unless you want to lose your data, allow someone to set up camp on your system, or possibly have your privacy compromised in an ugly way.

28. OK. I now understand the various ways I can get connected to the Internet from home and some of the risks involved in doing so. What I want to know now is how do I establish a network within my home so that my computers can communicate with one another?

What you are describing is the essence of a home-based LAN, which, as we mentioned in Chapter 1, is sometimes referred to as a personal area network (PAN). As is the case for any networking endeavor, a home-based LAN requires some advanced planning on what you want the network to do and how flexible it needs to be. For example: Will the infrastructure be wired or wireless (or both)? What protocols do you want to support? Do you want to share a printer? Do you want the computers to share the same Internet connection? What security measures will you implement? These are only some of the many questions you should consider. Furthermore, once you answer one of these questions, there usually are several more nested questions that need answers.

29. Let's start with the infrastructure.

This is a good place to begin—layer 1 of the OSI model. One approach to establishing a home-based LAN is to use a wired topology. This involves connecting all of your hosts via Category 5 UTP cable. As noted in previous chapters, Category 5 cable will support 10-Mbps and 100-Mbps Ethernet/802.3, which presumably will be your LAN protocol of choice. If your LAN will be restricted to one room, then this is probably the best way to set up your network. However, if your LAN will extend into other rooms within your house, then you will have to retrofit these rooms with new cable. This can be costly if you have someone else do it or it can be time-consuming and possibly messy if you do it yourself. If you opt to do this yourself, we suggest that you first read Appendix C.

Given the potential marketplace for home-based LANs, there has been a move within the industry to develop standards for home-based network infrastructures that do not involve retrofitting your home with new wires. The three primary strategies include wireless networking, telephone line networking, and residential powerline networking. A brief description of each follows.

Wireless Networking As discussed in Chapter 4, there are two primary and competing wireless networking solutions for home-based networks: HomeRF and IEEE 802.11b. The latest specification for HomeRF supports a 10-Mbps data rate and 802.11b supports an 11-Mbps data rate. There is also IEEE 802.11a, which supports a 54-Mbps data rate, but products were not widely available at the time this book was published. Regardless of the wireless technology method, a home-based WLAN requires a wireless adapter (i.e., a network interface card usually in the form of a PC card, see Chapter 6), software, and (optionally) an access point. If you are using IEEE 802.11b, be sure to select equipment that has the Wi-Fi seal (see Chapter 4). Two configurations are possible: peer-to-peer and semi-wired. In a peer-to-peer WLAN, all wireless stations are capable of communicating with each other directly. The key is that the stations must be within a few hundred feet of each other. In a semiwired approach, wireless stations operate as part of a wired LAN. This type of setup requires access stations, which determines whether a wireless node is permitted to access the network (see Figure 4.19). Access authentication is generally based on MAC sublayer addresses such the Ethernet address of the wireless adapter, but can also involve other parameters such as passwords. Once you establish your WLAN, you will need to experiment to determine optimal distances and possible physical barriers that can negatively impact data transmission rates.

Phone Line Networking The Home Phoneline Networking Alliance (HPNA) has developed a standard for linking multiple computers within a home via existing telephone line. The current specification, HPNA 2.0, operates at 10 Mbps. A phone line network requires an adapter card (either internal or external), a special cable, and software. You simply install adapter cards into your computers and then connect each computer into a phone jack using the special cable. Network configuration is then done using the included software. Phone line network kits consisting of two adapter cards, cables, and software retail for approximately $100. Phone line networks are very sensitive to noise (see Chapter 4) from telephones and fax machines, and can also be negatively impacted by attenuation if the phone wiring is extraordinarily lengthy. Additional information about phone line networking is available at http://www.homePNA.org.

Residential Powerline Networking The HomePlug Powerline Alliance (known more simply as the HomePlug Alliance) is a nonprofit industry consortium that developed an open, high-speed residential powerline networking specification that uses a home's electrical outlets. Completed in mid-2001, with products available in Fall 2001, the HomePlug 1.0 specification enables a home's AC power outlets to serve as a network connection similar to the way RJ-45 connectors provide a UTP Ethernet connection. A HomePlug network requires a special network adapter that is inserted into your computer's PCI slot. Attached to this adapter is an electrical plug that is inserted directly into a standard 110-

volt electrical wall outlet. The current HomePlug specification supports a data rate of 14 Mbps. When compared to the present state of wireless LANs, including HomeRF and IEEE 802.11b, HomePlug is the fastest of the "no new wires needed" home networking solutions. HomePlug also provides ubiquitous network connectivity because every room in a house has at least one electrical outlet. The specification also enables devices to be produced that support simultaneous electrical power and network connectivity through the same outlet. HomePlug using the same orthogonal frequency division multiplexing (OFDM) scheme as IEEE 802.11a (see Chapter 4) and is expected to eventually support data transmission rates in excess of 100 Mbps. Further information about HomePlug is available at http://homeplug.com.

30. What about software?

This depends largely on the operating system you are using. If you are using Windows 95/98/2000/NT or MacOS, then you already have the software needed on your system as part of the operating system. The same applies to many UNIX environments, including freeware versions such as Linux and FreeBSD. To make the two systems talk to each other, you must set up a peer-to-peer connection (see Chapter 1). This is when one system connects directly to the other with no intermediate router or server. On most systems, you must configure a protocol that is not dependent upon a router or naming service to discover route paths or name translation of addresses to get things to work. For instance, on a PC running Windows, you enable the NetBEUI protocol and adapter services in the network control panel in the Windows operating system version you are using. You must also select File and Print Sharing and allow the system to install the software (have your system CD-ROM nearby). Once you do this and reboot the systems, all you have to do is enable disk sharing on your systems. This is usually via the disk's Properties facilities; select the Sharing tab at the top. You will also need to decide on what security features you want to implement. By enabling "sharing" mode, you can now access the hard disks of the remote system. In Windows, you can double click on Network Neighborhood to find out what is on the network. Connectivity to a particular system is effected by simply clicking on the system icon. You are now sharing disks from your systems over your little Ethernet in your home just like you would at the office.

31. What's a good resource that I can consult to help me establish a home network?

An excellent site is the "How Stuff Works" site at http:www.howstuffworks.com. Specific information about how to establish a home-based LAN is available at http://howstuffworks.com/home-network.htm, which details the necessary configurations for networking PCs running Windows 98 and Windows Millennium. There is also information for connecting Macintoshes as well as links to other home network related topics.

32. OK. Now the big question: How do I share my Internet connection from one system with another on my little home-based LAN?

This takes a bit more work, but nothing major. There are several ways to do it If you are using Windows 95, you can use a variety of packages such as WinGate. These act as routers and allow multiple systems to be connected to a single dialout over a network. If you are using Windows NT or UNIX, these operating systems are equipped with routing kernels that can be configured to allow your Internet-attached node to route packets from the Internet connection to your other systems on the mini-Ethernet. If you read the system manager's documentation, there are instructions on how to configure the routing kernel on both operating system environments. If you are using Windows 98, Microsoft includes the RIP routing protocol for IP on the distribution kit CD-ROM. By installing RIP, you can configure the system as a router for other IP nodes on your network and share your connection with them. Other operating systems will or will not have similar capabilities. Of course, another option is to purchase a small dial-up router. This will cost between $200 and $1,000, depending upon connection type, speed, and router configuration required. So, as you can see, getting your systems configured to route to and from your Internet connection is not very hard or expensive.

33. What kinds of problems can I expect when setting up my home network?

There are several things that seem to trip people up when they configure a home network. Most are pretty straightforward to solve if you think ahead. The most common ones we have seen are summarized here.

Phone Line In this situation, the phone company does not properly install a phone line in the residence to the proper noise level allowances as posted by the FCC in the United States and other organizations that set phone standards in other countries. When the line voltages, attenuation, crosstalk, and other settings are out of whack on the line, you get noise, signal degradation, and a myriad of other problems. Sometimes, especially when there is a lack of phone lines in a community, the phone company will put an electronic box on a single pair of lines called an "active multiplexer," which allows the phone company to put the equivalent of more phone signals on the same pair of lines. These boxes are notorious for introducing serious noise and reduced signal strength on telephone connections, which degrades a modem's capabilities and the speed of the link. Phone line noise is a very common problem with home networks and can be very irritating. Also, if you are installing a phone line network, there is a 1000-foot wiring limit between devices.

Cable This is a problem area even for seasoned network professionals. Because there are so many ways to hook up devices and so many standards for different devices, there are a plethora of cable connectors, pinouts, pair-matches, and so forth that plague the well-connected network. It is therefore extremely easy to get the wrong cable in the wrong configuration between two systems when trying to configure a network. Other cable problems include trying to get the right connector type for a connection port, proper cable length, proper cable type (e.g., shielded, unshielded, coaxial), and connector types. Cables can be a real hassle when configuring home networks. (See Chapter 4 for additional information about cables.)

NIC Settings In some NICs there are various jumpers (small wire pins on the board) or dual in-line package (DIP) switch blocks. You must configure the jumpers or DIP switches to match the type of network connectivity options required for the network type you are using. You might also have to specify DIPs or jumpers for bus speeds, memory utilization, and other parameters. Many network cards now are software configurable, so more and more equivalent DIP switch settings are now done in software, but the aggravation factor is about the same. If you get DIPs, jumpers, or internal software settings wrong, the card will not work as planned or desired.

Network Protocols Problems involving network protocols will always emerge when setting up any network connection. Although recent versions of popular operating systems are making it easier to identify and correctly configure the required protocols, the problems that emerge can still be related to complex issues that will need to be resolved before things can work correctly. The bottom line is: If you are not comfortable working with network protocols, then get help from someone who is.

Power In most homes, the circuit where the PC is plugged into has all kinds of other stuff connected to it as well. Most homes are not configured for “clean” power. Furthermore, the circuit breaker serving a set of outlets is usually not of sufficient capacity to support all the computer-related devices (e.g., PC, monitor, scanner, printer, hublet) plus all the other electrical appliances. If the power output of the circuit drops below what a device requires, a “brown out” occurs and the system will reboot as if a power failure had happened. It’s usually a good idea to invest about $75.00 and purchase a battery backup power conditioning unit (sometimes called an Uninterruptible Power Supply or UPS) to ensure that the system and components will continue to function properly during power “hiccups.” Also note that if you are installing a power line network, you cannot plug the adapter plug into a surge suppressor plug because the system will not operate properly.

34. Is there anything else I need to know about dialup or home networking?

There is always something else. That’s one of the curses of networking—you can never know enough, and trying to keep current is a full-time job in itself. The material presented here should give you a good understanding of the concepts and issues related to dialup and home networking.

END-OF-CHAPTER COMMENTARY

On this note, we conclude our discussion of dialup and home networking. Additional information about some of the material presented here is available in other chapters. For example, Chapter 1 provides a discussion of network standards that is quite relevant to the discussion of modem and *x*DSL standards. Chapter 4 contains detailed information about hardware and cables. It also provides a discussion of Shannon’s Limit and how it is applied in determining the maximum speed of analog modems. The concept of error control is explained in Chapter 5, and Chapter 7 has specific information about the analog-to-digital conversion process. The discussion on Ethernet/802.3 networks might also be beneficial in further understanding some of the inherent restrictions of cable modem service. Chapter 11 provides information about ISDN, which is considered one of the solutions to the local loop bottleneck problem. Finally, the next chapter, Chapter 16, presents a discussion on network security issues that you might find enlightening relative to home network security.

Chapter 16

Network Security Issues

In this chapter we discuss various concepts of network security, including techniques, issues, and problems involved in implementing a safe and secure network environment. The material in this chapter represents an overview of network security, not a detailed treatise. An outline of the terms and concepts we define and discuss follows:

- Network Security Overview (Question 1)
- Threat Assessment and Risk Analysis (Questions 2–11)
- Social Engineering, Denial of Service, and Applications (Questions 12–21)
- Network Security Preparations and Measures (Questions 22–27)
- Firewalls (Questions 28–35)
- Cryptography and Encryption: DES, AES, RSA, PGP (Questions 36–52)
- Authentication: Digital Certificates, Smart Cards, and Kerberos (Question 53)
- Internet Security and Virtual Private Networks (VPNs) (Questions 54–57)

1. What is network security?

Network security is defined as the proper safeguarding of all components associated with a network including data, media, and infrastructure. A comprehensive approach to network security involves three essential elements, namely, accurate threat assessment, use of the best cryptographic tools available, and deployment of effective network access control products (e.g., firewalls). Perhaps most importantly, network security may only be achieved by ensuring that all network resources are used in compliance with a prescribed corporate policy and only by authorized personnel.

In light of ongoing cracker attacks and the proliferation of viruses, most people agree that network security is one of the most pressing issues today. Every organization requires it, but few have a firm grasp on how best to achieve it. There are many ways to achieve varying levels of network security. However, these methods can be extremely expensive or might not completely protect users from the many hazards that emerge on a daily basis. Proper implementation of network security is neither trivial nor inexpensive, and it requires expertise that encompasses most areas of network science. To help give you an example of what it involves, Box 16.1 provides information about network security and relates it to the concept of network *ethics*.

BOX 16.1: Network Security and Ethics

Network security involves the various measures to protect a network's components and resources from various threats including physical (e.g., fire, natural disasters, environmental, and sabotage), and illegitimate uses by personnel. There are no simple solutions to the issue of network security. It is prudent, however, for organizations to have established policies in place to protect themselves from security threats. Examples of such policies include: a disaster-recovery plan, a data back-up policy, giving users access to only those areas or levels of a system warranted by their job requirements, setting file permissions to reflect authorized access, using personnel to monitor or test a network's security, and encrypting data. Of all these measures, however, none compares to making users cognizant of their role in security. It is critical that employees be made aware of security policies and the reason for these policies.

Network ethics refers to specific standards of moral conduct by network users for the responsible use of network devices and resources. Access to these devices and resources is a privilege and should be treated as such by all users. Responsible, ethical behavior should be the rule rather than the exception and is perhaps the ultimate security measure that an organization can establish. The issue of ethics is important because people are usually the weakest link in any network security scheme. People are susceptible to threats or bribes, they can make mistakes (e.g., writing down a password and leaving it on top of a desk), and they can suddenly subscribe to some new ideology that contradicts their company's policies. When placed in a questionable situation, users must be cognizant of what constitutes right or wrong behavior if they are to be held accountable for their actions. The Division Advisory Panel (DAP) of the National Science Foundation (NSF) Division of Networking and Communications Research and Information (DNCRI), defines as unethical "any activity which purposefully or through negligence: disrupts the intended use of the networks; wastes resources through such actions (people, bandwidth or computer); destroys the integrity of computer-based information; compromises the privacy of users, [or] consumes unplanned resources for control and eradication" (*Communications of the ACM*, June, 1989, p. 688).

2. OK, I'll agree that security is important. Where do I start?

A challenge inherent to network security is determining the right level of security required for proper control of system and network assets. This concept, known as *threat assessment*, identifies the assets you have and who may attempt to access them. Organizations can best assess corporate network threats by using a structured approach. To illustrate, there are many *threats to hardware* (e.g., theft of computers or related equipment, destruction of computer equipment), *threats to software* (e.g., viruses or software bugs, software deletions from a hard disk, theft of CD-ROMs), *threats to information* (e.g., database corruption or deletion, theft of key database files), *threats to system* or network operations (e.g., network congestion, electrical interferences, power outages), and *threats to security measures*. (e.g., theft of passwords or IDs). Once you have assessed your network's vulnerabilities you are in the best position to protect your organization's assets. (See Box 16.2 for information about creating strong passwords.)

BOX 16.2: Creating Strong Passwords[a]

The single most important component of any network security scheme is the password assigned to networked devices. It is important that strong passwords (i.e., difficult to guess) be created and maintained. Failure to do so can compromise the security of the entire network.

When selecting a password avoid using: (a) any type of name, including yours or any family members or relatives (spouse, children, parents, aunts, uncles, cousins, in-laws, pets); the name of your company or colleagues; the name of an operating system such as "windows," "unix," or "macintosh"; the host name of your computer or your e-mail address; (b) anyone's home, work, cell, or fax telephone number; (c) any part of a social security or student number; (d) anybody's birthdate; (e) any dictionary word (English or foreign); (f) a geographical area (e.g., city, state, county, park); (g) any string of characters comprising the same letters, numbers, or pattern of numbers or letters (e.g., xxxxx, 12345, wysiwyg); (h) any of the above that begins or ends with a digit (e.g., unix1 or 5xxxxx).

Passwords that are difficult to guess or crack include a mix of uppercase and lowercase letters, digits, punctuation symbols, and special characters (e.g., =, *, ^, @), and are usually seven or eight characters in length. Three suggestions for creating strong passwords are as follows:

- Intermix the first letters of an easy to remember (short) phrase with digits, punctuation symbols, or special characters. For example, *It was twenty years ago today* is represented as Iw$ty^aT.
- Combine two relatively short words with a special character, digit, or punctuation mark. For example, *buzz* and *off* combined with the tilde character and upper- and lowercase letters produces BuzZ~OfF.
- Use letters, special characters, and punctuation marks to represent an English (or foreign) sentence. For example, *You are so lazy!* is represented as UrSoLaz!

Also refrain from writing down passwords. A password committed to memory is more secure than one that is written down. If it is necessary to write down a password then: (a) do not label the written text as a password; (b) keep the corresponding username separate from the password; (c) never tape the paper on which the password is written to any part of a computer or its peripheral units; (d) do not maintain an electronic version of the password; and (e) make the written version different, yet still discernible to you, from the real password. For example, if your real password is *IMover4T* (I am over 40), then you might write down *IM>4T!* (I am greater than 40), or vice versa.

Remember: *A single user with a weak password can compromise the security of an entire network.*

[a] Source: Adapted from Garfinkel & Spafford, 1991.

3. So what do I do first?

The initial challenge of threat assessment is distinguishing between critical and noncritical assets. This process is absolutely essential to protecting an e-business. For instance, pharmaceutical companies maintain an enormous amount of sensitive information, including sales figures, customer order information, clinical trial data, FDA documentation, agreements with partners and suppliers, and just about any other standard office information system facilities common among any other company. In this example, the most critical business asset is the compound database, which stores information about products and current research. This is the data source that ultimately generates the company's deliverables and, thus, the profits. Compromise this database and the business is critically injured.

As another illustration, let's take a look at the airline industry. In this example, the keys to managing the industry's vast operations are scheduling and load management, which implies computing. Scheduling is an enormously complex task involving the management of crews, equipment, supplies, cargo, passengers and many other tasks. While scheduling is extremely important and essential to business, a very critical problem for an airline is the computation of weights and balances. Pilots need to know the aircraft's takeoff weight based upon its load (e.g., fuel, passengers, cargo) to set the aircraft's flight systems for the calculated takeoff speed in various weather conditions. If this operation isn't handled properly, an aircraft becomes too heavy and is unable to fly very far. In this case, it's clear that the scheduling system is a mission-critical asset that must be protected at all costs.

It's easy to understand the critical nature of network security in our airline example. However, in many companies, people are much more concerned about sales figures and cash flow than network security. Isn't it just as important to safeguard strategic product plans, the exposure of which might result in a lawsuit? What if your Web server goes down resulting in a major public relations disaster? High-visibility companies are particularly vulnerable to bad press, and they have a great deal to lose if they experience a major security breach. Imagine that a bank gets cracked and this becomes public knowledge? Who would want to save their money at a bank that cannot safeguard its own information? Every business has items that are crucial to safeguard from external parties. Identifying what they are and how damaging they could be to the company is the first step toward effective threat assessment.

4. Fine. I think I understand why threat assessment is important. Are there other steps to it?

Yes. Now that you have an idea of which resources are crucial to a company, the next challenge is determining who would want them and how might they use these resources against you. In the case of the pharmaceutical company, the loss of a patent or patent-worthy research might, in the long term, result in millions of dollars of lost revenue. Determining who might benefit from the assets you control helps you identify potential threat locations and who might attempt to steal what a company deems vital to its operations.

Sometimes the threat to a company is not a competitor but someone who harbors a vendetta against that company. Companies that perform animal testing for product safety verification (e.g., pharmaceutical companies, cosmetic companies, food-related companies, defense contractors) are frequently under attack by activist groups. Some of these groups have paramilitary factions that break into labs destroying equipment and data. Some groups have even targeted networks and computing infrastructure with specific viruses designed to infiltrate a company's systems and subsequently attack research data. In short, the threat to your data may not always be your competition and sometimes the threats come from sources you'd least suspect.

5. What do you mean by that? Who would threaten my network more than my competition?

In 1991, an FBI study found that over 80% of network crimes were committed by insiders within a company. In 1997, the number was adjusted to about 75% (direct internal and "known" dial-in users) through a joint study by the FBI and the Computer Security Institute. The change was mostly due to the fact that Internet and external break-ins were more accurately documented. In 2000, the percentage of internal attacks was refined again, this time to 50%. This refinement is primarily the result of better statistics on external attacks being maintained. However, any network security expert will probably tell you privately that they generally see more internal breaches than external attacks even today.

To root out potential threats from insiders, we must determine which employees might want to (a) threaten the company, (b) cause asset loss, (c) abuse fiscal fidelity, (d) sell to a competitor, (e) cause public relations problems for the company, (f) hurt customer relations or compromise customer information integrity, (g) cause internal outage of resources that would hurt corporate operations, or (h) do anything that might damage a company's ability to do business and generate revenue. Most companies do not take special precautions to protect internal network assets. This is a big mistake. Although most companies understand that security arrangements must protect corporate networks from external attacks, nearly nothing is done to prevent internal attacks, which is where most threats originate. In short, key assets must be protected from both external and internal threats. We must first understand the source of internal threats so that we can keep the honest folks honest and catch the criminals before they create problems.

6. OK, so now I can't trust anyone, right?

Not exactly, but you do have to be very careful about who has access to what and why. That includes internal personnel.

7. Now that I'm pretty paranoid, what else do I have to worry about to implement network security properly?

To properly understand the threat to a company, we have to assess how much the loss of assets will cost. This concept is called *risk analysis* and can be quite complex in larger companies that control many assets. In most cases, performing some simple calculations on the cost of replacing, upgrading, repairing, or managing a threat situation will produce some startling numbers. You should compare the costs to managing a threat situation to the price of defending the situation. Specially designed risk analysis software is available to assist you in this endeavor. These tools range in cost from $100 to as much as $20,000 per license. The National Institute of Standards and Technology (NIST, formerly the National

Bureau of Standards) publishes a risk analysis software guide, *Guide for Selecting Automatic Risk Analysis Tools*, Publication Number 500-174.

8. Can you give me an example of how poor risk analysis can impact a company?

Yes. Following is an example of the impact poor risk analysis can have on an organization: One day a large public company with a name we all know discovered that its Web pages was modified and included several obscenities. Prior to this event, management did not feel the need to secure the corporate Web site and placed the Web server on the outside of a fire-wall system. As it turned out, while no corporate assets were lost, the security breach resulted in two major problems for the company. First, there was a great deal of negative publicity in the local and national news, which caused a great deal of embarrassment and compromised the company's standing in the industry. To manage the negative media attention, a professional "crisis management" PR firm was called in for a fee of over $300K. Next, the repairs to the Web server and addition of the security technologies had to be implemented very quickly—and at great expense. Fixing this disaster cost the company substantially more than setting up an effective security solution would have cost at the outset. The second problem was that the Web pages served as a buyer's catalog and this hurt business in two ways. First, search engines that used blocking software based on URLs placed the company's Web pages in a blocking state in their databases. This effectively locked out potential customers who might use these search engines to find the company's home page because they were denied access to the company's site. Second, confidence in the integrity of the Web site was badly compromised. Many companies that were considering using the Web site for e-commerce prior to this incident backed out of the deal for over six months until they were satisfied that the company had taken measures to prevent a recurrence of the problem. In short, something as simple as an unauthorized Web change caused a ripple effect, resulting in a multimillion dollar revenue loss and a serious public relations problem that took several months to cure. As you can see, a "minor" security breach can cause major problems that are expensive to correct, especially if the problem was never properly planned for in the first place.

9. What else should I do to understand all the potential threats to my network environment?

Another threat assessment method is simulated *war games*. The Pentagon does it. Large companies do it. Kids play them on the weekends and in school. Role-playing games or "war games" in security terms are essential to truly understanding the threat environment and making sound decisions on what can and cannot be protected.

10. Sounds interesting. What kinds of things am I trying to simulate?

In a network attack, the idea is to examine all the potential points of attack and then try some out to see if they work, how they can be detected and defeated, and how to handle the repercussions (e.g., business, political, public relations, technological) of the attack. War games, simulations, role playing, or whatever you wish to call it is extremely useful in properly identifying threats and counterthreats. It also allows proper identification of actions that any warrior must understand to launch a proper defense. These include (a) setting up information defenses, (b) monitoring for information attack, (c) delaying the attack until assessment and reinforcements are made available, (d) counterattacking and capture or destroy, and (e) cleaning up any problems discovered. By exercising the foregoing concepts, all threats can be identified and proper defenses planned.

11. So far, so good. Any final items to consider for the threat assessment part of network security?

Yes. Documentation and updates. All of the previous items are for naught if documentation does not follow the work. The legacy should be recorded for others and updated as the threats change. This process can take a substantial amount of time and effort but it is an essential part of any well thought-out plan for threat management.

12. What's a common method used to attack a network resource?

"Social engineering."

13. Really? What is that?

Social engineering involves deception; it is, basically, lying your way into a facility. It takes a certain amount of self-confidence and the ability to talk your way out of situations where you might get caught. It also takes a certain amount of thinking "on your feet" so that you do not get into a situation where you will be discovered too soon. The idea is to test the facilities in your purview without getting caught too soon, if at all. From this exercise, you'll discover which improvements should be made.

The main purpose behind social engineering is to place the human element in the "network breaching" loop and use it as a weapon. For instance, having someone show up at the computer room with network hardware in hand and an appropriate vendor ID usually results in a staff person escorting you into the communications closet and, perhaps, even helping you to install hardware on the network. This is not good. By appealing to the victim's natural instinct to help a customer, it can be startlingly easy to breach the physical perimeter of a company and gain access to the network and all its treasures.

14. Any ideas on how someone might socially engineer themselves into my company?

One favored technique is to walk into a customer's branch office and tell a tale of woe to those employees. A popular story is that you work for the corporate office and have to kill a few hours before a flight. You might say: "Do you think that there might be an empty cubicle around where I can work for a while? I don't need anything except a place to park and work." Usually, someone finds an empty space for you, the "corporate visitor," to work and almost invariably, there is a live network connection in that space. Since most laptops have either a built-in Ethernet/802.3 controller chip or PC card—perhaps also a token ring PC card, you're good to go. With such a network-capable laptop loaded with various network analysis and protocol analysis tools, you, the intruder, can collect data from the network and gain valuable network attack information (e.g., passwords or user IDs). A further enhancement to the deception tactic is using a business card of someone else in the corporation to give the false impression that (a) the intruder is indeed that person, (b) the intruder does indeed work for the company, and (c) the title of the person on the card is “important” enough to impress employees into providing access to a work area.

15. That sounds like you have to have some technical skills. Are there easier ways?

Yes. Here’s another example. One company created a rather elaborate security clearance procedure for users prior to gaining access to its network resources. An intruder, however, completely circumvented the company’s established security procedures for accessing new accounts and acquired an invaluable list of accounts by simply contacting network personnel and posing as a vendor offering to demonstrate software. All this was accomplished with very little effort and without the consent of upper management, which was required by corporate policy. As you can see, there are many ways to thwart established security procedures by exploiting human nature.

16. Is it hard to find information on how to breach or crack a site?

It is not very difficult to find information about how to initiate a network attack. On the Internet, for example, there are, quite literally, thousands of locations that include very specific instructions on how to attack almost any type of protocol, operating system, or hardware environment. Much of the information available promulgates known operating system or application protocol weaknesses. For example, the simple mail transfer protocol (SMTP), which is the backbone of Internet e-mail service, has been the source of several exploitations over the past 15 years. In a study that assessed corporate network security of 46 customer networks over a six-month period, Cisco Systems identified SMTP as the second most vulnerable service. According to the report, “If an SMTP server is accepting Internet connections, there’s approximately a 61 percent chance that it will be misconfigured or that it’s running outdated software that contains security bugs” (Cisco Systems, 2001, p. 7). A listing of the most vulnerable Internet services reported from this assessment is given in Table 16.1. A copy of the complete report, called the *Cisco Secure Encyclopedia*, is available to authorized users at http://cisco.com/cgi-bin/front.x/csec/csecHome.pl.

17. I have heard of a network attack called a denial of service. What is that?

Several years ago, some crackers began writing small programs that would “flood” a network address by sending as many packets as possible to the address using network utilities such as *ping* (see Chapter 3). Continuously “pinging” a machine causes it to crash or to stop accepting connections because of traffic overload. This particular type of attack, in fact, was called “Ping of Death.” Variants on this idea were written to take advantage of protocol weaknesses, system bugs, or holes or application problems with high traffic volumes. These types of attacks collectively became known as *denial of service* (DoS) attacks. They were so named because users were denied access to the services of the system that was attacked. DoS attacks do not actually compromise or damage a host. They simply shut down the host or service by blocking out legitimate traffic. DoS attacks have become quite commonplace on the Internet.

A variant of a DoS attack is a “stateful” attack, which exploits the handshaking process that various protocols implement when establishing connections. An example of this

TABLE 16.1 Internet Services Most Vulnerable to Security Breaches[a]

Service	Percentage of Time Security Problem Was Found
Remote Procedure Call (RPC)	93.4
Simple Mail Transfer Protocol (SMTP)	61.1
Finger	59.6
Trivial File Transfer Protocol (TFTP)	57.4
Domain Name Service (DNS)	35.0
File Transfer Protocol (FTP)	33.0
Network File System (NFS)	30.2
Simple Network Management Protocol (SNMP)	27.1
Hypertext Transfer Protocol (HTTP)	26.7
X-Window System	23.0
Authentication or Logins	19.9

[a] Based on a six-month review of 46 corporate network systems.
Source: Vulnerability Statistics Report (11/5/2000); Adapted from Cisco Systems (2001)

handshaking process is TCP's three-way handshake, which we discussed in Chapter 3, and an example of a stateful attack is the TCP SYN attack. Recall from Chapter 3 that a TCP SYN message is required during a session activation as part of a normal connection setup. In a SYN attack, random source addresses are connected to a receiving system but never respond to the receiver. In this manner, the attacker can create thousands of "half" sessions on the destination system until all the memory or network connection slots are expended and the system stops accepting new connections. Thus, system services are denied by "ghost" sources that never acknowledge the next state of the connection handshake. So instead of denying services by overwhelming a system continuously with packets as a ping attack does, a stateful attack such as TCP SYN causes a system to wait until the protocol times out for a response, effectively rendering the system and its resources inaccessible. In fact, if an attacker is persistent, the destination system can time out all incoming connections after a short time so the system is unusable to anyone attempting a connection.

18. Are DOS attacks the same thing as a zombie attack?

What you are referring to is a highly malevolent form of a DoS attack called *distributed denial of service* (DDoS). A DDoS attack involves placing hundreds or thousands of small programs on different computers connected to a network. These programs are called agents or, in common terms, Trojan horses or "zombies." From a master console somewhere on the network, a cracker activates them by sending a command to the zombies and forcing them into attack mode. The zombies then begin sending messages to the specified destination, completely overwhelming the destination system as well as the network pathways to get to the system. In a DoS attack, single systems are usually targeted, and the network resources in the path are usually not adversely affected except for slow links. A DDoS attack, however, results in a "scorched earth" from a network perspective. Because of the distributed nature of the attack, the many networks used by the zombies to get to the destination system are totally overwhelmed by traffic and are effectively disabled.

19. How do you combat a DDoS attack?

Although some DDoS attacks can be stopped, it depends on the type of attack. For example, some DDoS attacks have specific profiles and use specific protocols, which can be detected by firewalls and, in some cases, router filters. Some types actually generate what appears as legitimate traffic to the network. These "legit" traffic profiles are extremely difficult to detect and defeat. As a result, the only option for some types of attacks is placing packet rate limiters on routers. A packet rate limiter is a type of router filter that detects certain elevated traffic levels for specific conditions and then slows the packet rate down to keep from overwhelming the network connections. In this manner, the attack is throttled back to a condition where it is more of an annoyance and not destructive. Another method for stopping DDoS attacks is by reverse engineering the zombie command set so that the equivalent of an OFF command can be issued to the zombies just as the master console would do to stop the attack. These are not always successful, though, and are not appropriate for detecting the attack.

20. Where does something like the ILOVEYOU e-mail worm fit into all of this?

The "ILOVEYOU" e-mail worm, which began appearing in mid-2000, is yet another variant of a DoS attack called *application-level denial of service* (ADoS). Launching the ILOVEYOU program eventually caused loss of e-mail services (and hence a denial of application service). These types of worms can be filtered by e-mail server virus killers and e-mail string filters, so they are not as insidious as DDoS. On the negative side, though, they can take up a great deal of e-mail server space with totally useless data. Furthermore, some can download a secondary component that can be extremely destructive if activated. For example, some of the Visual BASIC scripts that are downloaded can do some interesting things such as delete a hard drive's content or copy it to some public location for all to see. There are also many copycat variants of these types of attacks that usually flood networks within a very short time of the initial program's release.

21. I never realized there were so many different types of attacks. Is there anything that can be done to prevent them from occurring on my network?

Yes and no, depending on the type of attack. For example, an ADoS attack can be prevented by filtering e-mail attachments and by educating users not to open unexpected or suspicious e-mail attachments. DDoS attacks, however, usually cannot be prevented. Table 16.2 provides a summary of the primary attack types, along with various preventive measures.

22. I am beginning to think that a network attack is inevitable. Given this premise, what can I do to be prepared?

You're right. All networks get attacked from time to time. Larger networks get attacked more often because, statistically, they are more active and visible to the networking world and are therefore an easier target to find and attack. Given the constant threat and inevitability of a network attack, you should be prepared for the day your network is compromised. We call this state of preparedness *incident response*, or IR. Incident response is analogous to calling the fire department when you detect your house is on fire. It consists of preparation, prepositioning of technology, and then real-time response. Preparation

TABLE 16.2 Summary of Primary Internet or Network Attacks

Type of Attack	Description	Preventive Action
Reconnaissance	Gathers information from services such as SMTP and Finger that could be used later to compromise a system.	Disable all unneeded services on routers and servers. (See Table 16.1 for most vulnerable services.)
Access	Exploits known vulnerabilities in Internet and network authentication services (see Table 16.1).	Use strong passwords (see Box 16.2); protect e-mail servers using firewalls or proxy servers; restrict incoming TCP requests to only those needed.
Denial of Service (DoS)	Sends large volume of traffic to a specific host or port that effectively shuts down access to the host or service.	Difficult to prevent because attack can mimic legitimate traffic. Best to have measures in place to address the attack once detected. Also: read RFC 2267; install an intrusion detection system; disable unnecessary network services.
Distributed Denial of Service (DDoS)	Compromises many hosts (called zombies), which then are directed in concert to attack a specific host or group of hosts.	Same as DoS.
Application-level Denial of Service (ADoS)	Usually an e-mail virus or worm that compromises a specific user application such as e-mail.	Filter out e-mail attachments; educate users not to open unexpected or suspicious attachments.

involves surveying the network and thoroughly understanding the topology and components that make up the network and its vulnerabilities. This is similar to a fire department conducting a building inspection—it provides firefighters with critical information about the building, including where everything is located and any special hazards or problems the building presents in the event of a fire. Prepositioning means getting equipment and software in place to collect real-time information and to allow the IR team quick access to information that is critical to determining the attack profile and specific information about the attack. Real-time response is the ability to put technology and expertise on the attack problem in a very short amount of time, preferably while the attack is in progress, so that the IR team can isolate the attack, stop it, and assist law enforcement agencies in identifying and prosecuting the intruder.

23. This all sounds good, but there is no way my company has resources to do this. And even if we did, I doubt if management would authorize it.

You just identified two principle obstacles to implementing network security: lack of resources, including personnel, money, and tools, and upper management apathy. In a recent survey about network security, nearly 60 percent of respondents blamed weak security on a lack of staff to handle the issue. Another 55 percent said their budgets are insufficient to the task. More than 40 percent attributed the problem to nonexistent software tools, and 45 percent blamed upper management. Management claims security is extremely important because of the myriad potential threats, including current and former employees, suppliers, customers, and competitors. However, there usually is no management support or funding to protect a company. Unfortunately, line items for personnel support (e.g., training) and security are absent from most network budgets.

24. Can you give me some additional examples of how lax security measures can impact a company? We seem to have a problem not only with network security but also with overall system security. I would like to use this information when I discuss network security with my supervisor.

Good for you. Network and system security should not be treated independently because they are indeed linked to each other. Good network security begins with the stability of each system on the network, and in many cases, network vulnerabilities can increase a company's risk of data manipulation and destruction. Here are some examples of how data manipulations can harm an organization. On a research site, jealous coworkers might "taint" extensive research data, causing scientists to reach improper conclusions. Financial analysts might find that their carefully crafted spreadsheets produce improper computations on critical budgeting line items due to static, dynamic, or formulaic information being altered. Word processing documents containing standard contracts and procedures might be modified to weaken the document's meaning and undermine the intent of a contract. Database demographic information may be erased or modified to produce false market data that is critical to a rollout of service or business offerings, causing much time and money to be wasted in the creation of market products and services. The list of opportunities for malicious manipulation goes on and on. Without good system security, the network can increase the opportunity for security breaches.

25. Are there companies or special targets for security breaches?

Yes. The nature of some businesses makes them prime targets for security breaches. For example, companies that engage in animal testing might be the targets of groups that engage in the willful destruction and infiltration of corporate assets to bring attention to the issue of animal testing and to disrupt corporate entities involved in such testing. Medical practices, such as family planning centers and abortion services facilities, are often the targets of groups formed for the express purpose of putting such operations out of business. Other companies may contain specific technologies that are of interest to industrial espionage specialists who wish to sell information about the technologies to competitors. Companies with extensive telecommunications systems are often the subject of PBX attacks that allow the intruder access to free long-distance services for their use without the corporate entity knowing about the attack for a very long time—if at all.

26. How do I properly implement network security if the process is so complicated and replete with obstacles?

You do so by not thinking of network security from the perspective of a single control point. Instead, think of network security as a series of security barriers. The reality is that any single control point can be breached by an expert. However, by placing multiple security barriers in the path of a critical asset or resource, the chances of someone getting through without detection are greatly diminished. The concept is much like that of the strong castle protected by a series of moats. As the storming horde nears the castle, it must traverse the moats. It is possible to traverse some moats by pole-vaulting, but eventually the invader is likely to fall into one of the moats and be caught. If there is only one moat and the invader is a good jumper, odds are the invader will succeed. If there are moats, trip wires, razor wires, tall fences with broken glass on them, land mines, Doberman pinschers at the gate, and other such traps placed in the path between the intruder and the "jewels," one or more of the obstacles are likely to alert the keepers of the castle that someone is trying to infiltrate it and that something must be done to protect the assets and destroy the intruder.

The notion that a singular perimeter is an effective defensive measure for network asset protection is long gone. Since many network assets are attacked by internal personnel, the need to protect the assets from external and internal attack is rapidly becoming the norm. This notion is similar to the plan advanced by the Great Wall of Texas Society in the United States. The society's intent is to build a 30-foot-high wall around the northern sections of the Texas border with New Mexico, Oklahoma, Arkansas, and Louisiana. Their whole premise is that the concern over the border with Mexico is outdated; they assert that the real enemy is to the north, not the south, and that "we have been watching the wrong border for too long." Similarly, most companies are concerned about external intruders, but the enemy within is far more dangerous. The fact is that the bulk of all network break-ins happen from inside the company. Therefore, intrusion protection for a network should never be limited to external network connections; it should always start directly with the information repository itself as the first line of defense. Box 16.3 provides some basic commonsense network security measures. In addition to the security measures listed in Box 16.3, there are many different technologies that can be implemented to help make a network more secure. These include firewalls, encryption techniques, authentication systems, and access control measures. All are discussed next.

27. What technology can I implement to help me make my network more secure?

There are many. Three in particular are firewalls, encryption, and digital certificates. (Remember: These are only three technologies; there are many others.)

28. What's a firewall?

Historically, a firewall was defined as a "wall" placed between an automobile's floorboard and engine to keep an engine fire or explosion from entering the passenger compartment. In the context of network security, a firewall performs a similar function between the connections on a network. Specifically, firewalls are devices or products that allow the systems or network manager to restrict access to network components. There are various types of products that claim to be firewalls but which clearly are not firewalls. One sad aspect of firewalls is that the terminology has been overused, much like the word "virus." What can and cannot be accurately called a firewall is subject to interpretation by the vendor and the consumer.

At its most basic level, a firewall is a packet-filtering facility that can restrict the flow of packets to and from a network via a set of rules implemented in an interconnection

BOX 16.3: Network Security Measures

Following are several measures that can enhance the security of a network:

Electrical-Related

1. Use dual power supply modules for network critical devices.
2. Connect devices to uninterruptible power supplies (UPS).
3. Ensure that all network devices are connected to “clean” power.
4. Install surge protection directly to the main circuit panels that feed the electrical outlets to which devices are connected.

Environmental-Related

1. Ensure all rooms with network devices are properly ventilated or air conditioned.
2. Do not place devices in combustible areas, and restrict the use of volatile materials (e.g., cleaning supplies) in these areas.
3. Install smoke detectors and adequate fire-extinguishing equipment in all rooms with network critical devices.
4. Do not place network devices in areas that are susceptible to flooding or exposed to water (e.g., a utility closet with a sink).
5. Do not place network devices near windows.
6. Place critical devices such as servers in locked rooms with an alarm system, and restrict access to these rooms to authorized personnel only.

Media-Related

1. Minimize the physical exposure of copper-based cable by enclosing it in a conduit; any taps that penetrate the conduit will be noticeable.
2. Maintain a physical map of your network that includes a wiring diagram so you know if any cable has been tampered with.
3. Identify the location of any buried media before the ground is dug up.
4. Use a cable scanner to scan the cable and record values. Do this periodically and compare readings.
5. Maintain a secure wiring closet. This includes using a separate, enclosed space with a locked door for the wiring closet.
6. Enclose all cable buried underground in metal pipe and document its location.
7. Do not use copper cable to interconnect buildings. If you must use copper, enclose it in a metal conduit and place it high enough so that it is not easily accessible.
8. Use fiber-optic cable when possible, particularly in high-security locations.
9. Always encrypt data prior to transmission.
10. Use line-of-sight transmissions instead of broadcast transmissions.
11. Use optical-based links instead of RF-based links.
12. Use wired-based media instead of wireless media.

(continued)

device. Examples of this might be a filtering router unit that is capable of restricting which packets can be transmitted and which ones can be received from an Internet connection based upon packet addresses (source and destination) or a specific IP transport protocol

BOX 16.3: Network Security Measures (continued)

User-Related

1. Create and maintain strong passwords to all systems (see Box 16.2).
2. Do not give personal access privileges to unauthorized users.
3. Understand the potential security risks related to connecting your workstations to the Internet via a dialup line.
4. Understand the differences between client and server processes and the security implications related to enabling server versions of Internet utilities.
5. Virus-check all software downloaded from the Internet.
6. Disable TCP/IP routing and all other unneeded services if your workstation is connected to a LAN and if you intend to use it for Internet access.
7. Understand the potential hazards of e-mail attachments; be alert for unexpected or suspicious attachments and do not open them.

Network Administration-Related

1. Enable, maintain, and review system accounting and log information regularly.
2. Install virus protection software and update virus definition files regularly.
3. Keep virus protection subscriptions current.
4. Enforce idle time-outs for dialup connections.
5. Prevent users from uploading data to a system's hard disk. If this is not feasible, then restrict uploads to an area that is automatically virus-checked.
6. Maintain proper file permissions.
7. If possible, place all critical data on a centralized server and protect the server.
8. Establish secure, centrally managed servers for remote workers.
9. Edit configuration files carefully and always save a copy of the current file prior to making modifications.
10. Do regular and frequent backups of all data onto tape or disk and store backups in a different location than the original data.
11. To minimize the effect of unauthorized packet-sniffing programs, design your network using switches so that each workstation has its own dedicated network segment. This reduces the overall amount of network traffic, and it limits the type of data that can be collected to broadcast or multicast messages. Alternatively, encrypt all messages prior to placing them on the network.
12. Disable all unneeded network services from routers and servers.
13. Establish and enforce acceptable use policies for all users.
14. Disable a user's access immediately after he or she leaves the company.
15. Develop, implement, and review on a regular basis a disaster recovery plan.
16. Keep current with Computer Emergency Response Team (CERT) publications and regularly install all authorized security updates and patches.
17. Subscribe to various network security listservs.
18. Attain network security tools and learn how to use them and interpret their output.
19. Periodically check your security by attacking your own system and network.
20. Document *all* activities.
21. Understand your network and know what's running on it.

type. Other types of firewalls might include intelligent port and socket (application) filters, session-level (user) filters, and a variety of other types of filtering tools that restrict traffic flow. In short, a firewall is frequently a sum of many different components that work together to block transmission and reception of traffic.

29. What types of firewalls are there?

There are five generally accepted types of firewalls used on Internet connections: frame-filtering, packet-filtering, circuit gateways, stateful and application gateways, and proxy servers. There are other less-known proxy firewall implementations and variants, but they all function fairly similarly, with varying degrees of performance and ease of configuration. Following is a brief description of each type:

Frame-Filtering Firewalls A *frame-filtering firewall* has the ability to filter to the bit level the layout and contents of a LAN frame (e.g., Ethernet/802.3, token ring/802.5, FDDI, and others). By providing filtering at this level, frames that do not belong on the trusted network are rejected before they reach anything valuable, even on the firewall itself.

Packet-Filter Firewalls A *packet-filtering firewall* is either a router with packet-filtering capabilities or a dedicated device that does packet-filtering. A packet-filtering firewall is best used as a dedicated unit in conjunction with a router. This way, the router does not have to perform a dual (and contradictory) function—it can facilitate communication as it is designed to do, and the packet-filtering firewall can provide the network security. The performance of a packet-filtering firewall will degrade considerably as more filters and conditional filter handling are set up. Packet-filtering also does not handle certain types of transactions on a network that are context sensitive. That is, many packets are required to do something, which, when taken as a whole, implies a certain (and usually ominous) condition has occurred.

Circuit Gateway Firewalls A *circuit gateway firewall* typically involves the session setup between a system and the user security options relative to that system for a particular user. For instance, a circuit gateway might check user IDs and passwords for a connection request. Other types of circuit firewalls might implement proxy connection authorization or other types of authentication services. Circuit firewalls are also responsible for logging who came from where and went to what, which is not trivial.

"Stateful" Firewalls Following the establishment of proxy firewalls (described next), the need to examine the transaction condition between two interoperating applications becomes essential to defeating certain sophisticated types of network attacks. IP address spoofing, session hijacking, piggyback session acquisition, and many other technical attacks were allowing crackers access to applications and eventually entire systems. To stop this type of attack profile, the firewall must be intelligent enough to watch all transactions between two systems and understand enough of the details of how a protocol works to identify a specific condition in the transaction between two applications, be able to predict what should transpire next in the transaction, and be able to detect when normal operational "states" of the connection are being violated. This type of firewall is called a *stateful inspection facility* and allows the network security manager to specify rules and filters for specific technical transactions between the systems and applications and what to do if they are violated by anyone. Many vendors of stateful firewall facilities also include detailed filtering capabilities similar to proxy filtering. In some cases, however, proxies do a better security job (depending on the application being secured), so stateful firewalls, for the most part, are capable of providing a full security rule-based range of services but sometimes are not as complete for specific applications as a proxy might be.

Application Gateways or Proxy Firewalls An *application gateway firewall* provides protection at the application level. If viewed from a functionality perspective, an application gateway firewall is the opposite of a packet-filtering firewall—the former is application or program specific and the latter is general purpose. For example, consider a typical file transfer session. Suppose you want users to be able to download files from the Internet using the file transfer protocol (FTP) but you do not want anyone from outside your organization placing files on any of your networked hosts. More specifically, you want to permit "get" FTP sessions but reject "put" FTP sessions. With a packet-filtering firewall, you have an all or none case—either it allows file transfers (get and put) to occur or it does not. An application gateway firewall, however, can be configured to permit "get" sessions and reject "put" sessions because it can examine the details of the application. Another example might be a Telnet firewall facility that provides security facilities, full packet content scanning, session management, session capturing, and other facilities. This type of firewall is specific to a particular IP application, Telnet, and is usually much more secure than packet and address filtering in a router—it might not only consider user IDs, passwords, and proxies, but it also might consider application-specific access methods and security issues.

An application gateway firewall uses custom programs for each protected application. If a new application that requires protection is added to the network, a new program has to be written and added to the set of other programs that reside on the firewall. These customized application programs act as both a client and server and effectively serve as proxies to the actual applications. For example, if e-mail is to be protected, a custom e-mail application is written that includes specific security rules (e.g., what type of e-mail is permitted). When users want to use e-mail, they must either log into the application gateway and use this special application or use a client application on a host that supports this secured e-mail service. Since these specially written applications act as proxies to their "real" counterparts, the collective set of these programs is commonly referred to as *proxy services*, and application gateway firewalls are often called *proxy server* or *proxy gateways*. There are two types of proxy gateways. In the first type, an incoming connection for a destination would be intercepted by the proxy, and a "new" connection from the proxy to the destination would be created. In this manner, a connection originating from outside the firewall is not able to "touch" the destination directly, and full filtering of the application is accomplished. The second variant of the proxy gateway allows the firewall to appear as the only destination for all applications to a trusted network from an untrusted network. Through this facility, the internal network is completely "hidden" from network view to any outside connections. This has the by-product of allowing an internal network to use unregistered address ranges for IP users to access the Internet and other external networks expecting valid address ranges. It also increases security by not allowing an external network direct access to an internal address or to any knowledge of what the internal address is for a specific node on a trusted network.

30. Is there a simple way to select a firewall? Isn't there one vendor that offers all the types of firewalls in one package?

Since firewall requirements vary dramatically from company to company, there are many situations in which more than one product from more than one vendor is required to provide proper firewall facilities. A router with packet filters would be almost a necessity for each site. A user terminal security facility for Telnet users is also necessary, but no routers manufactured today can provide both the sophisticated security facilities for terminal traffic as well as swift routing facilities. Consequently, these two functions alone will result in the need for separate systems for control and access.

31. How hard is it to set up a firewall, anyway?

One of the difficulties of firewall implementation is the fairly high degree of technical expertise required to configure them. If one does not understand TCP/IP reasonably well, there is little hope of properly setting up a packet-filtering facility. Exactly how much technical knowledge is required depends on the model, but frankly, no firewall is easy to set up. Almost without exception, no matter how mature and well-written the software is, there are a myriad of administrative tasks involved in setting up a firewall.

32. Can a firewall be defeated by an attacker?

Even with fairly effective firewalls for Internet access, there are situations that, left unchecked, will cause defeat of the firewalls. For instance, end-to-end protocol encapsulation (see Chapter 7), which involves tunneling a protocol within another protocol, can be very difficult to filter and control. Some sites, for purely political reasons, will not permit restrictions on certain applications that allow remote Internet users to gain access to critical data about a site that may be used to exploit the network. It is important to note that, even in the best environments, firewalls don't work forever and may be defeated.

33. When implementing a firewall, do I have worry about other network protocols or just TCP/IP?

Yes, you do. In 1995 and 1996, the concept of generally accepted Internet applications (e.g., Web servers, POP- and IMAP-based e-mail) being deployed on internal networks for employee access began a steady and dramatic increase in both scope and size. Mission critical applications are now deployed on these intranets while existing network technologies continue to coexist on the same network environment. With "real" networks on corporate facilities, the existence and use of protocol suites besides the IP facilities found on the Internet represent the bulk of networking protocol use. In most corporate networks, especially those established in the 1980s and early 1990s, the common use of Novell's IPX, Digital Equipment Corporation's Pathworks environment (DECnet, LAT, LAST), IBM's System Network Architecture (SNA), Apple Computer's AppleTalk, and Banyan's VINES environment (StreetTalk) are still very active and very much a part of the culture at many companies. Other corporate environments support specialized protocols (e.g., ALC) for terminal applications that are very popular in a particular vertical industry segment (e.g., the airline industry) but are not seen in mainstream computing. Still other network types, such as industrial networks, warehouse networks, process control networks, and many other types of specialized networks, utilize custom protocols for maximum efficiency or for specialized applications where commercially accepted protocol suites are much too large or too general to solve a network problem.

The bottom line is that multiple protocols, not just TCP/IP, are the norm on intranet and extranet networks. Unfortunately, most firewalls available today are designed to operate only within a TCP/IP environment. To protect only one protocol, such as IP, is ludicrous in a standard corporate network environment where the norm is to deploy between 3 and 18 protocol suites to satisfy legacy and computational requirements. Popular operating systems such as Windows NT/2000 come configurable with multiple protocols to satisfy corporate clients (e.g., NT comes with IP, IPX, NetBEUI, and AppleTalk standard on each system) and provide connectivity to the variety of systems on a corporate network. Protection of an IP application environment is fine if that is all a system is using, but many systems utilize other protocols. Protection of a single protocol on intranet or extranet environments without consideration for the other operating protocol's security needs is much like locking the front door of your house while leaving the back door open. Proper network security requires all entities to receive equal protective treatment to truly address the network security threat.

34. Yeah, but we made a decision to only use TCP/IP for our protocols. We aren't multiprotocol, right?

This is a common misconception. Even in locations where IP is the only or predominant protocol, there is the problem of a new version of the IP protocol stack, namely, IPv6. (See Chapter 3 for additional information about IPv6.) Sometimes referred to as IPng (for "next generation"), IPv6 is not only different in its address structure (e.g., it uses a 16-octet address compared to IPv4's 4-octet address) but it also differs in the mechanics of how a system receives its address (it is done dynamically) and what the remainder of the protocol does to address routing issues, security layer issues, and other new features of the protocol. To say that the current IPv4 is "compatible" with IPv6 is a gross mistake, particularly when the related RFCs call for IPv4 and IPv6 "coexistence" and not "compatibility." Coexistence, in technospeak, means that a machine may run both protocols at the same time to achieve the ability to use both at the same time, as opposed to running the new one and continuing to converse with the old one (compatibility). Therefore, for environments running only IPv4 at this time, they are about to become multiprotocol, like it or not, even if all that is running is the current version of IP and the upcoming upgrade to the protocol. No network manager who is cautious about the network environment will totally cut over to IPv6 without a parallel phase in which both protocol suites are running. In some cases, this will be a lengthy time period for sites where applications may require IPv4 to remain for application "survival" indefinitely. Therefore, to protect the internal network from attack by internal personnel, and to support multiple versions of a protocol such as IP with external networks such as the Internet, networks will need multiprotocol protection because, as you now know, the need to protect the internal network is as great or perhaps greater than the need to protect the external network.

35. So which type of firewall is better than the others?

The fact is, each firewall architecture has its merits and drawbacks. Let's review the features of all major firewall types. Router screening is fast and allows rejection of common errors, cracker attacks, and user strangeness which is part of any network connection. Application filtering firewalls provide extensive application control and monitoring of application behavior. Proxy facilities provide application control and session control between sources and destinations as well as address translation facilities. Stateful firewalls allow technical attacks from breaching a network and provide sophisticated filtering techniques that rival almost any proxy or application gateway. In short, almost all firewall approaches have strengths and weaknesses. In fact, this is a good thing for customers. What is best for the customer is security, implemented for the right reasons and in the most productive manner. This means that the optimal firewall configuration is one that can perform all the various types of rule-based filtering previously described depending on the type of application being used and the best security methodology approach to solve the security problem for the application environment. No single security rule-based approach that has been described can properly address all security issues in a networked environment. Most network security experts will tell you that it is impractical to expect one approach to be useful in all environments. A list of other security features that you might want to consider when evaluating routers or firewalls is given in Table 16.3

36. OK, I think I understand firewalls. You also mentioned encryption. What is that?

Encryption is a process that converts sensitive data into a coded form. When retrieved by authorized users, this coded form is then reconverted (i.e., decoded) into meaningful text. Encryption essentially hides or disguises information from unintended recipients, but enables authorized users retrieve it. The study of secret communication is called *cryptology,* and the practice or art of encoding messages is called *cryptography.* Unencrypted data is referred to as *plaintext* and its encrypted output is called *ciphertext.*

A simple encryption technique is a letter-substitution cipher. For example, let's agree that we will use the following "key" for coding and decoding messages:

A → O	B → D	C → C	D → I	E → S	F → V	G → A	H → P	I → Y
J → M	K → F	L → R	M → Z	N → X	O → E	P → B	Q → G	R → N
S → K	T → W	U → H	V → Q	W → U	X → J	Y → T	Z → L	, → #

Now, the message,

DEAR JANE, NOT GETTING ANY BETTER, HURRY HOME

is coded by substituting the plaintext characters with those of our cipher. Thus, the encrypted message is,

ISON MOXS# XEW ASWWYXA OXT DSWWSN# PHNNT PEZS

This message can now be sent via public channels and decoded by an authorized person who knows the key. If the message is intercepted or finds its way into the hands of an unauthorized person, it will most likely appear as meaningless gibberish. Of course, it is always possible that an unintended recipient could crack the code.

TABLE 16.3 Additional Router-Firewall Security Features to Consider

Feature	Description
Audit Trail	Records all session activities, including source and destination hosts, ports, duration, time stamp, and bytes transmitted.
Authentication Proxy	Authenticates users as being authorized to access a LAN.
Denial of Service (DoS) Detection and Prevention	Examines packet headers and filters those deemed suspicious.
Intrusion Detection	Monitors network traffic for known security breaches, filters out suspect packets, and sends alert notice to management console.
Java Applet Blocking	Filters Java applets from unknown or untrusted sources.
Traffic Filtering	Enables network administrator to specify traffic that is permitted to access the network.

Source: Adapted from Wexler, 2001.

37. In what ways can cryptography be implemented?

Cryptography can be implemented either symmetrically or asymmetrically. In *symmetric cryptography*, a secure communication between two parties is effected by exchanging a secret key (also called a private key), which is used both to code (i.e., encrypt) and decode (i.e., decrypt) a message. Ideally, key exchanges should not be done electronically, though, and are best conducted in person. A feature of symmetric cryptography, which is commonly called *private key encryption*, is the speed in which messages are encrypted and decrypted. A disadvantage, though, is the number of keys required when more than a few people are involved. For example, only one key is needed for two people to exchange secret messages. However, if ten people are involved, then 45 private keys are needed. In general, n people require $(n)(n-1)/2$ private keys. An example of a private key encryption strategy is the data encryption standard (DES), which is described later.

In *asymmetric cryptography*, each person maintains two keys—one private and one public. The public key, which is published and available to anyone, is used to encrypt a message; the corresponding private key, which remains secret and is not given to anyone, is used to decrypt the message. For example, if Randee wants to send Ron a secret message, Randee encodes her message using Ron's public key, and Ron decodes the message using his private key. A message can also be encoded using the private key and decoded using the public key. Thus, the two keys represent a key pair that must be used in tandem. Although asymmetric cryptography significantly reduces the number of private keys that need to be exchanged, it runs much more slowly than symmetric cryptography. Two examples of asymmetric cryptography, which is commonly referred to as *public key encryption*, are Pretty Good Privacy (PGP) and RSA. The concepts of private key encryption, PGP, and RSA are described more fully later.

38. Is any of this related to public key infrastructure, otherwise known as PKI?

Yes. Cryptography is a critical component of a relatively new technology called *public key infrastructure* (PKI). PKI involves the process of issuing, delivering (i.e., publishing), managing, and revoking public keys. For example, secure Web transactions are built on a PKI. Web browsers (i.e., clients) and Web servers use what is known as a *secure sockets layer* (SSL) session for engaging in secure communications via the Internet. Prior to establishing a secure connection to a Web server, a browser first requests a copy of the server's security certificate, which contains the server's public key. After verifying the certificate's authenticity, the browser generates a symmetric key, which it will use to encrypt its data. This key is also encrypted using the server's public key. Both the encrypted data and the encrypted symmetric key are transmitted to the server. For the server to decrypt the coded message, it must first decrypt the encrypted key. Since the browser's symmetric key was encrypted using the server's public key, the key is decrypted by the server using the server's private key. (Remember that a private and public key pair work in tandem.) Once the server has the browser's symmetric key, it then decodes the message. It will also use this symmetric key to encode any data that it transmits to the browser. Note how secure Web transactions actually employ a hybrid cryptography strategy. A browser's symmetric key is used to encode data, and a server's public–private key pair is used to securely exchange the symmetric key between the browser and server across the network. Most SSL-enabled Web browsers have preinstalled certificates, which can be viewed by examining the security option.

39. How can I see a certificate? I'd like to know what one looks like.

In Netscape 4.7, go to Communicator/Tools/Security Info and then click on Signers. Any of the listed certificates can then be viewed by selecting one and clicking on Edit. In Internet Explorer 5.0, go to Edit/Preferences and under Web Browser click Security. Any of the listed certificates can then be viewed by selecting one and clicking on View. A Web server's certificates, which originate from a root certificate authority (CA), all have corresponding expiration dates and hence are only valid until this date. Once a certificate expires, it is considered revoked. Unfortunately, as of this writing, current Web browsers are not capable of detecting revoked certificates.

40. This is really interesting. Tell me about DES.

Developed by IBM and NIST in the 1970s, the *data encryption standard* (DES) is a mathematical model or algorithm that is used to encode data. It is also the most widely used commercial encryption algorithm. As with letter-substitution cipher, a key is used to determine the transformation from plaintext to ciphertext. The key for a DES user consists of any one of 2^{56} possible keys, each one of which is a list of 56 zeros and ones (plus eight parity bits). This translates to 7.20575×10^{16} or 72,057,500,000,000,000 possible permutations (i.e., distinct arrangements) of 56 zeros and ones.

DES, which can be implemented in hardware or software, has been the single, most thoroughly tested encryption algorithm. In over 20 years of testing, DES was never cracked. However, in 1998, John Gilmore and Paul Kocher broke the code in 56 hours using a homemade supercomputer they built for $250,000. Funding for the supercomputer, which was a configuration of hundreds of Intel processors, was provided by the Electronic Freedom Foundation (EFF). The project was sponsored by the U.S. government and carried a $10,000 prize. The time to crack DES has since been reduced to 22 hours.

41. Yikes! It sounds like we need to either improve DES or replace it.

You're right, and the government boys know this as well. To enhance its cryptographic capabilities, alternatives to DES were implemented. One increases the number of keys from 56 zeros and ones, to 1024 zeros and ones. This increased key size makes the algorithm more complex and hence more difficult to crack. A second replaces DES with a completely different algorithm called Skipjack, which uses an 80-bit key space, and a third uses a variant of DES called Triple DES, which involves three DES operations instead of one. With Triple DES, a data message is first encrypted by the first key. This encrypted message is then encrypted a second time by a second key. Finally, this twice-encrypted message is encrypted yet one more time by a third key. Most people today who require high security use Triple DES for two reasons: it is compatible with DES, but, more importantly, unlike DES, it is unbreakable using current code-breaking techniques and computing power.

42. Does this mean, then, that Triple DES has replaced DES?

Not quite. One of the problems with Triple DES is that it is highly computer intensive because you are applying the DES algorithm three times before the message is transmitted. This not only taxes computing resources, but it also takes time to do, which delays the transmission of data. Although this is not a problem presently, Triple DES will not be sufficiently robust to satisfy the increased demand for data encryption as more sensitive and private information is transmitted across the network. As a result, in 1997, NIST announced the first public and worldwide competition for the design of an *advanced encryption standard* (AES) algorithm to replace DES. The final AES criteria mandated that the new algorithm implement a private key (i.e., symmetric) cryptography, be a block cipher, and operate on 128-bit blocks with the three key sizes of 128, 192, and 256 bits. Twenty-one proposals were received by the competition's June 15, 1998, deadline, with 15 of the 21 satisfying NIST's criteria. In August 1999, five finalists were selected after careful review by the global cryptographic community. The names of the proposed AES algorithms (and their corresponding proposers/country) were: MARS (IBM Corporation/U.S.), RC6™ (RSA Laboratories/U.S.), Rijndael (Joan Daemen and Vincent Rijmen/Belgium), Serpent (Ross Anderson/United Kingdom, Eli Biham/Israel, and Lars Knudsen/Norway), and Twofish (Bruce Schneier, John Kelsey, Doug Whiting, David Wagner, Chris Hall, and Niels Ferguson/U.S.). In October 2000, Rijndael (pronounced "Reign Dahl," "Rain Doll," or "Rhine Doll") was selected as the AES because of its combination of security, performance, efficiency, ease of implementation, and flexibility. Rijndael was ultimately approved as an official United States standard by the U.S. Department of Commerce in summer 2001.

43. What will become of DES and Triple DES?

DES is being phased out of use, although it is still permitted in some legacy systems. Triple DES will, for now, remain as an approved algorithm for U.S. government use.

44. How many possible arrangements of 0s and 1s does AES support and how long will it take to eventually crack AES?

As indicated earlier, AES specifies three different key sizes: 128, 192, and 256 bits. A 128-bit key equates to 3.40282×10^{38} permutations of 128 zeros and ones; a 192-bit key equates to 6.2771×10^{57} permutations of 192 zeros and ones; and a 256-bit key equates to 1.15792×10^{77} permutations of 256 zeros and ones. NIST estimated that if a machine is capable of cracking one DES key (i.e., one unique arrangement of 56 zeros and ones) per second, then it will take approximately 149 trillion years to crack a 128-bit AES key. Given that DES was unbreakable for 20 years, NIST believes that AES will remain secure for at least 20 years as well. As an official U.S. government standard, Rijndael will be formally reevaluated every five years.

45. Who is permitted to use this algorithm?

AES will be used domestically and internationally. Within the United States, government organizations will use it to protect sensitive information. Commercial and other non-U.S. government organizations are not required to adopt and implement the AES, but it is expected that they will do so. Although none was available as of this writing, products implementing the AES are expected by the end of 2001. Thus, AES will soon be used for e-mail, financial transactions, medical information, and a host of other application-specific purposes. Furthermore, given its international roots, AES implementations will also be exportable. This is in stark contrast to current U.S. Department of Commerce policy, which prohibits the exportation of encryption devices and algorithms with key sizes greater than 64 bits. (There is no limit on key sizes within the United States.) For additional information about the AES, see http://www.nist.gov/encryption/aes.

46. DES and AES are private key algorithms. Can you give me an example of a public key algorithm?

Sure. A good example is RSA, which is a widely accepted and implemented method of public-key encryption. RSA, which stands for the first initials of the last names of its designers—Ronald Rivest, Adi Shamir, and Len Adleman—requires two keys for each user, one public and the other private. Each user's public key is available to anyone, whereas the private key is secret and known only to the user. A message coded with the public key is decoded with the private key, and vice versa. This system provides three distinct methods of sending coded messages. The first method involves sending a coded message to a receiver from a sender whose identify is verifiable. To illustrate:

> If we wish to send you a coded message, we code our message using your public key. This way you can decode it using your private key.

The problem with this method is that the receiver of the message can never be absolutely certain who the sender is because it is the receiver's public key that is being used to code the message. The second method involves sending a coded message from a sender whose identity is verifiable, but the message can be decoded by anyone. To illustrate:

> If we wish to send you a coded message and we want you to be absolutely certain that it is from us, then we will code our message using our private key. You in turn decode our message using our public key. Since it is our public key that actually decodes our message, you are certain that the message came from us.

Authentication of a sender's identity, as demonstrated by this second method, is very important in certain transactions such as in electronic funds transfers. Now, to be absolutely clandestine, if we want to send you a coded message and we want you to know it is from us, a third method is used:

> We first code our message using your public key. (This makes the message secret.) We then encode this code once more using our private key. (This guarantees that the message is from us.) Upon receiving the message, you must decode it twice, first using our public key and then using your private key. Now, only you can read the message and it could only have been sent by us.

47. OK. This makes sense. What would really be helpful at this stage is to see a real encryption algorithm. Is it possible for you to demonstrate how one works?

Yes. A description of the RSA algorithm with an example of its implementation is provided in Box 16.4. Be forewarned, though, that as is the case for any encryption algorithm, RSA is mathematical in nature., As a result, we also provide some background mathematics to refresh your memory.

48. Can you give me an example of an application that uses RSA?

Yes. One that you might have heard of is *Pretty Good Privacy* (PGP), an e-mail encryption package written by Philip Zimmerman. PGP combines three separate algorithms: RSA, International Data Encryption Algorithm (IDEA); and the Message-Digest algorithm, Version 5 (MD5). IDEA is a conventional encryption algorithm similar to DES that uses a 64-bit block cipher with a 128-bit key space. MD5 is a hashing algorithm—developed in 1991 by Ron Rivest and described in RFC 1321—that takes a message of arbitrary length and generates a 128-bit message digest. A message digest is used for digital signature applications.

PGP provides both encryption and digital signature services. Encryption enables a user to encode files; digital signature service enables a user to “sign” a document so that the document’s authenticity can be confirmed by checking the signature. Thus, encryption provides confidentiality; a digital signature proves a message was not modified. Encryption service is provided via RSA and IDEA. PGP first encrypts messages with IDEA, and then uses RSA to encrypt the IDEA key that was used to encrypt message initially. Intended recipients then use RSA to retrieve the IDEA key, which in turn is used to decode the message. PGP also uses MD5 to create a digital signature, which is then encrypted using RSA.

PGP was once the source of much attention by many U.S. legislatures. When PGP was first released, encryption algorithms with key sizes of greater than 40 bits in length were prohibited from being exported outside the United States. Since PGP incorporates IDEA, which uses a 64-bit block cipher with a 128-bit key space, it was illegal to export it. Unfortunately, someone made PGP available on the Internet without Phil Zimmerman’s knowledge and Zimmerman was indicted for exporting PGP outside the United States. As noted earlier, the Department of Commerce has since removed the restriction on exportation of encryption devices and algorithms with keys of 64 bits or less. Another incident involved patent infringement. Earlier versions of PGP (v2.3 or earlier) contained the RSA algorithm, which, at the time, was patented in the United States. Consequently, users of these versions risked patent infringement if they used PGP in the United States without a license. In May, 1994, however, an agreement with the RSA patent holder was reached

BOX 16.4: The RSA Public Key Cryptosystem

The RSA public key cryptosystem is a mathematical algorithm with roots in a branch of pure mathematics known as number theory. The algorithm's basic design involves elementary number theory concepts such as prime numbers, greatest common divisor (GCD), and modular systems. Before we present the RSA algorithm, we provide some brief mathematical background related to these fundamental number theory concepts.

MATHEMATICAL BACKGROUND

Prime Numbers

The field of mathematics consists of several different classifications of numbers. One classification frequently used is the set of *integers*. The set of integers is an infinite set of numbers which is represented as $I = \{\ldots, -3, -2, -1, 0, 1, 2, 3, \ldots\}$, where the ellipsis symbol, …, denotes that the observed number pattern continues indefinitely. The set of integers can be partitioned into three distinct sets: the set of *whole numbers*, $\{0, 1, 2, 3, \ldots\}$; the set of *natural* or *counting numbers*, $\{1, 2, 3, \ldots\}$, which is also called the set of *positive integers*; and the set of *negative integers*, $\{\ldots, -3, -2, -1\}$. If an integer p is greater than 1 (denoted $p > 1$) and is divisible by only itself and 1, then p is called a *prime number* or, more simply, is referred to as being *prime*. For example, given the set of the first 100 positive integers, $\{1, 2, 3, \ldots, 98, 99, 100\}$, the set of prime numbers contained within this set is

$$\{2, 3, 5, 7, 11, 13, 17, 19, 23, 29, 31, 37, 41, 43, 47, 53, 59, 61, 67, 71, 73, 79, 83, 89, 97\}$$

Note that we exclude 1 because 1 is not prime by definition and that 2 is the only even prime number. An integer $p > 1$ that is not prime is called *composite*. Given these two definitions, we observe that the set of negative integers, as well as the integers 0 and 1, are neither prime nor composite. Furthermore, around 300 B.C., a mathematician named Euclid proved that there are an infinite number of primes or, expressed another way, that there is no single largest prime number.

Greatest Common Divisor (GCD) and the Concept of Relatively Prime

When two numbers are multiplied together, the result is called a product, and the numbers being multiplied are called factors of the product. For example, given $p = 2$ and $q = 4$, the product of pq is 8, and 2 and 4 are the factors of the product, 8. To determine if a number n is a factor of a given product, we divide the product by n. If the division yields a 0 remainder, then n is a factor of the product. Factors are also thought of as *divisors* of a product because they evenly divide (i.e., yield a 0 remainder) a product. Thus, 2 and 4 are divisors of 8. The *greatest common divisor* (GCD) of two natural numbers is the greatest (i.e., largest) natural number that evenly divides the given pair of natural numbers. Note that the GCD may also be thought of as the greatest common factor (GCF). For example, given the two natural numbers 12 and 16, the divisors of 12 are 1, 2, 3, 4, 6, 12, the divisors of 16 are 1, 2, 4, 8, 16, and the GCD of 12 and 16, denoted GCD (12, 16), is 4. Now consider the two numbers, 24 and 25. Note that the GCD (24, 25) = 1. If the GCD of two natural numbers p and q is 1, then p and q are said to be *relatively prime.* Thus, 24 and 25 are relatively prime. Similarly, 8 and 15 are relatively prime because the GCD (8, 15) = 1, but 15 and 36 are not relatively prime because the GCD (15, 36) = 3. The GCD (p, q) may be found readily for small values of p and q by factoring them into their prime factors.

(continued)

BOX 16.4: The RSA Public Key Cryptosystem (continued)

For larger values of p and q, this process becomes tedious and hence is done by applying a rule known as Euclid's algorithm. (For more information about Euclid's Algorithm, consult a book on number theory. Also, note that the fundamental theorem of arithmetic states that any natural number greater than 1 is either prime or can be expressed as a product of prime factors.)

Modular Systems

A modular system is a mathematical system that cyclically repeats itself. For example, the set, $\{0, 1, 2, 3, 4, 5, 6, 7\}$, contains only the given eight elements. It is a finite set, and only the elements within this set may be used. This means that if we were to work with these set elements in sequence, after using the number 7, we must cycle back and start with 0. A modular system with set elements $\{0, 1, 2, 3, \ldots, m-1\}$ is called a *modulo m system*, which is abbreviated mod m. Thus, our given illustration represents a modulo 8, or mod 8, system. A number that is not an element of a given modulo system set is found by dividing the number by the mod, m. This is formally denoted as $a \equiv b \pmod{m}$, where a is the given number, b is the remainder that results from dividing a by m, and m is the mod. An interpretation of this notation is that a and b have the same remainder when divided by m. For example, $23 \equiv 7 \pmod{8}$ and $916 \equiv 20 \pmod{32}$.

THE RSA ALGORITHM

The strength of the RSA algorithm is based on prime numbers. Specifically, its design takes advantage of being able to readily generate very large prime numbers, but at the same time, the algorithm's security feature relies on the difficulty in trying to factor the product of very large prime numbers. The algorithm itself is described in three stages. The first stage corresponds to the steps involved in generating two keys—one public and one private. Public keys are usually published; private keys are kept secret. The second stage encrypts a given message, and the third stage decrypts the message. A running example is provided for each stage. If a given message is encrypted via a public key, then it is decrypted using the associated private key. Similarly, if a message is encrypted using a private key, then it is decrypted using the associated public key. Hence, the two keys are inverses of each other, and manipulating a message with the two keys successively, in either order, results in the original message.

Stage 1: Key Generation

1. Randomly generate two large prime numbers, *p* and *q*

We will select $p = 5$ and $q = 11$. In practice, though, p and q should be very large (e.g., 300 decimal digits each) because small values are not secure. At one time numbers consisting of around 100 decimal digits each were considered sufficiently large. However, during the mid-1990s, a message that was encrypted using values of p and q such that their product yielded a 129-digit number was cracked in less than a year. Thus, larger values for p and q should be used to preserve the basic premise on which RSA was designed.

2. Find *n* by letting *n* = *pq*

Given $p = 5$ and $q = 11$, $n = (5)(11) = 55$.

(continued)

BOX 16.4: The RSA Public Key Cryptosystem (continued)

3. Let $m = (p - 1)(q - 1)$

Given $p = 5$ and $q = 11$, $m = (5 - 1)(11 - 1) = (4)(10) = 40$.
This step is considered an intermediate step and represents a special mathematical function known as *Euler's phi*, which is denoted as $ø(n)$. See Cormen, Leiserson, & Rivest, 1990, p. 817, for additional information.

4. Find a small, odd integer, *e*, that is relatively prime to *m*

Recall from the earlier mathematical background information that e and m are relatively prime if GCD $(e, m) = 1$. Since the numbers we are working with are small, we find e by trial and error. Generally, though, Euclid's Algorithm is used.

- If $e = 2$, then GCD $(2, 40) = 2$
- If $e = 3$, then GCD $(3, 40) = 1$

Since GCD $(3, 40) = 1$, we let $e = 3$.

5. Find an integer, *d*, so that $de = 1 \pmod{m}$ and $d < m$

Finding an integer d so that $de = 1 \pmod{m}$ is equivalent to the solving the equation, $de = 1 + am$ where $a \geq 0$, for d. Thus, $d = (1 + am) / e$. This step formally computes d as the multiplicative inverse of e mod $[ø(n)]$. (See Cormen, Leiserson, & Rivest, 1990, p. 834, for additional information.) Since the numbers we are working with are small, we find d by trial and error.

- If $a = 0$, then $d = [1 + (0)(40)] / 3 = 1/3 = 0.33\overline{3}$ (This is not an integer)
- If $a = 1$, then $d = [1 + (1)(40)] / 3 = 41/3 = 13.66\overline{6}$ (This is not an integer)
- If $a = 2$, then $d = [1 + (2)(40)] / 3 = 81/3 = 27$ (This is an integer)

Thus, $d = 27$.

6. Let the public key = (*e*, *n*)

Given $e = 3$ and $n = 55$, the public key = (3, 55). This key gets published.

7. Let the private key = (*d*, *n*)

Given $d = 27$ and $n = 55$, the private key = (27, 55). This key remains secret.

Stage 2: Message Encryption Stage: **$E(s) = s^e$ (*mod* n)**

To encrypt a message, we apply the public key to the function, $E(s) = s^e \pmod{n}$, where s is a given message, and e and n represent the public key integer pair. One example of an electronic message might be the number of sequential bits a data frame comprises. For our running example, we select the number 4 as our message, s. Thus,

$$\begin{aligned} E(s) &= s^e \pmod{n} \\ &= 4^3 \pmod{55} \\ &= 64 \pmod{55} \\ &= 9 \pmod{55} \end{aligned}$$

As a result, our encrypted message, $E(s) = 9$. This is what gets transmitted.

(continued)

BOX 16.4: The RSA Public Key Cryptosystem (continued)

Stage 3: Message Decryption Stage: $S = [E(s)]^d \pmod n$

To decrypt a message, we apply the private key to the function, $s = [E(s)]^d \pmod n$, where *E(s)* is the encrypted message, and d and n represent the private key integer pair. The result should equal the original message, 4. Note the various mathematical manipulations we employ so that we can work with reasonably sized numbers. These manipulations are based on laws of exponents and by continually expressing a given result in mod 55. For example, in the third step below, $9^3 = 729$, which is equal to 14 (mod 55). The process of raising a number a, to a power, b, mod another number, m, that is, $a^b \pmod m$, is formally called *modular exponentiation*. An efficient method used to compute this power electronically is known as *repeated squaring*, which uses the binary representation of b. See Cormen, Leiserson, & Rivest, 1990, p. 829, for additional information.

$$
\begin{aligned}
s &= [E(s)]^d &&\pmod{55}\\
&= (9)^{27} &&\pmod{55}\\
&= (9^3)^9 &&\pmod{55}\\
&= 14^9 &&\pmod{55}\\
&= (14^3)^3 &&\pmod{55}\\
&= 49^3 &&\pmod{55}\\
&= 4 &&\pmod{55}
\end{aligned}
$$

Thus, we successfully decrypted the encrypted message.

and subsequent versions (v2.6 or later) of PGP can now be used legally, for noncommercial purposes, in the United States without a license. Today, though, this is a moot point because the U.S. patent for the RSA algorithm expired September 20, 2000, and is now in the public domain. However, PGP is still subject to the rules of the International Traffic in Arms Regulations and cannot be exported without an export license. Additional information about PGP can be found at http://www.pgp.com.

49. What if I put a cryptographic product like PGP on my notebook computer so that I can communicate with the office securely? What legal trouble will I get into?

You can use the product in the United States without any trouble. Use outside the United States, though, is another matter altogether. Therefore, if you reside outside the United States and plan to put a cryptographic product like PGP on your notebook computer and use it for secure remote communications, it might be prudent to consult your government's regulations first. If information is to leave your country of origin, there are usually export laws regarding the use of cryptography either in or out of your country. In the United States, the cryptotechnologies that can and cannot be used is regulated by the Department of Commerce. DOC also regulates which countries can do business with U.S. companies. The Department of Trade and Industry (DTI) helps regulate security issues in the United Kingdom and determines what can and cannot be exported or used outside the country. In France, rules on cryptographic use in communications are extremely strict and in many cases cryptography is not allowed at all unless your company is an approved financial institution. It gets pretty complex country to country.

50. I've always heard that if something is encrypted, it's secure. Right?

No. Nothing lasts forever, and that includes cryptography. There is always a way to exploit a weakness in an algorithm or key structure in cryptography, but it may be very difficult to do so. The real issue with the use of cryptography is economic: Is your data valuable enough to justify the computer power and people power required to decrypt whatever you are transmitting over a network? If so, you may still be a target anyway. If not, then less complex cryptography facilities may be more than sufficient for your security requirements.

51. Wait a minute. Are you saying that cryptography comes in different strengths?

Absolutely. The degree of cryptography "strength" is a function of several components. Undoubtedly, the algorithm and key lengths are two of the more important components that help determine how difficult it is to break the cryptographic facilities being used. There are, however, other issues that affect cryptography strength. One is the manner in which keys are exchanged. Key exchange is the same idea as how to get new passwords in a secure way to the users. It is basically a computerized method to exchange cryptographic keys. Other factors include standardized implementation of the cryptographic methods, and export laws in various countries. The sensitivity of the data you're trying to protect will dictate how "strong" the cryptographic facility should be. For instance, current sales figures are really only valid for one quarter of the year and, in a public company, they will be published the following quarter. Therefore, the cryptographic "strength" required to protect these figures is only going to safeguard the information from general viewing for a brief period of time (assuming that the data cannot be viewed via another system on the network or easily compromised by individuals who know about the data). If you are storing patent information or data that have a long-term economic viability, more care should be exercised in the selection of the cryptographic methods used to safeguard the information.

52. Which algorithm is best to use for encrypting network traffic?

This is a tough question to answer. There are really two aspects to encrypting network traffic that have to be addressed to determine which algorithm(s) and key length(s) you need. They are (a) identifying what you are trying to protect and assessing its value and (b) taking into account any plans you might have for encrypting traffic destined for outside your county's borders and the laws governing this action. As we've said before, the algorithm and key length are important for safeguarding the information. In the United States, DES implements a 56-bit key length and Triple DES uses a 168-bit key. AES is based on key sizes of 128, 192, and 256. So single DES may be plenty strong enough for the bulk of the traffic you are going to pass around, Triple DES may be necessary for really sensitive data such as 5-year plans and the like, and AES may be essential for safeguarding the keys of the cryptographic method, which is critical. If someone knows the keys for the method, then it does not matter how secure the method is—it's broken! Thus, the answer to the question, "Which is best?" depends on what you are trying to protect and where.

Speed may also be an issue. This was one of the considerations in selecting the AES. Some algorithms are much slower than others, and this may factor into the decision as to which one to use. For example, DES is faster than RSA in real-time applications. RSA, as with all public key encryption schemes, is time-consuming. Time is needed to "sign" a message and verify it at the receiving end. DES, on the other hand, requires more keys than RSA. Public key systems can do authentication and encryption, but DES can only do encryption. As you can see, there are trade-offs. One alternative is to use a combination of conventional encryption and public key methods.

A comprehensive reference guide that contains the specifications for a wide range of common public key strategies, including key agreement, digital signatures, and public key encryption is the *IEEE P1363 Standard Specifications for Public Key Cryptography.* This document provides detailed descriptions of the primary encryption algorithms, including RSA and Diffie-Hellman. IEEE P1363 represents a different type of IEEE standard in that it provides a set of tools from which implementations and other security standards can be designed. Thus, instead of defining interface specifications, P1363 defines functional specifications relative to cryptographic parameters and public key strategies. Corresponding to P1363 is IEEE P1363a, which is a supplement to the base standard and contains information that did not get published in the original document. The two documents will eventually be combined in a future revision. IEEE also is planning a second supplement, P1363b. For additional information about the P1363 series of reference guides, see http://grouper.ieee.org/groups/1363.

53. Two words are frequently here in the context of network security are authentication and access control. What do these terms mean?

Authentication is a form of identification verification; it is a process of verifying a claimed identity. This can include not only the sender's identity, but also the sender's message as well as the receiver's identity. The concept of authentication is similar to providing someone access to a secure area. A security guard, for example, needs to verify the identity of the person who seeks admittance. There are several ways in which this can be done. For instance, access may be granted on the basis of the person possessing a badge, by a known individual vouching for the person, or by the person reciting a secret code or phrase.

Network authentication systems are not much different from what we just described. For example, access to a network or computing resources might be granted by a user ID and password, through the use of a magnetic-strip card or badge, or via some unique human physical characteristic such as fingerprints or retinal patterns. All of these access methods, known formally as *access control measures*, are an important component of the authentication process. This is because authentication implies that the remote node believes you are who you say you are; that is, it proves the sender of a message knows the "key." Just because you have a key to the front door of the house does not mean that you have permission to go through every dresser drawer in the house. The combination of authentication and access control ensures that you are who you say you are and then allows you access to only what you are supposed to access. All of this activity might also be encrypted. Once again, there are many pieces to making sure network security works well, and no single security measure works well alone. One example of an authentication system is RSA, which we discussed earlier. RSA verifies to a remote node the sender's identity. Three authentication measures are digital certificates, smart cards, and Kerberos. A brief description of each follows.

Digital Certificates *Digital certificates* are kind of an electronic passport. People use a variety of techniques to identify each other: looks, sounds, smell, and feel. Fingerprints, retinal scans, facial thermography, and other biometrics also help identify someone as being who they claim to be. In most cases, though, implementation of biometric security devices is in its infancy at this writing and expensive. A simpler method is to have someone identify certain attributes about themselves to a separate entity in which a sender and receiver both have a level of trust. This third party issues the user a numerical value, pattern, or key called a digital certificate. The certificate, in conjunction with cryptographic tools, identifies a specific user on the network, regardless of where the user is located or what application the user is using, in a reliable method. Digital certificates are available from a wide variety of trusted parties such as VeriSign. Unfortunately, there are as many ways to provide digital certificate authentication as there are vendors, and this means that their use tends to be spotty and somewhat chaotic at this writing.

Smart Cards Smart cards are similar to digital certificates in that they represent another form of authentication. Similar to a credit card, a smart card has integrated circuits embedded in it that store information in electronic form. Smart cards use personal identification numbers, biometrics (e.g., fingerprints, voice, signature), and encryption methods to authenticate a user. Smart cards communicate with an external "reader," which can be a computer system, a cash register, or any other type of input device. The method of communication is either by direct contact or by radio signals. In either case, once contact is established, the "reader" provides the required voltage to power the card.

Smart cards have several applications besides providing authentication service for network security. For example, the medical profession uses smart cards to store patients' personal medical histories. This technology provides privacy and protection of patient records, enables the tracking of medication and medical information, and enables a patient's insurance coverage to be verified almost immediately. Smart cards can also store patient's x-rays and other graphical data. Another application is education. Many universities provide students with smart cards for meal plan authorization. These cards also can be used as credit or debit cards for purchasing products, as a library card for checking out books, as a "vending" card for photocopying or vending machines, and for accessing secured buildings or dormitories.

Kerberos In Greek mythology there is a three-headed dog called Cerberus that guarded the gates to Hades. Its namesake is used to describe a client/server network security authentication system. Originally developed at MIT, Kerberos, which is based on DES encryption, is an Internet standard that uses a three-pronged approach for authentication: a database that contains users' rights, an authentication server, and a ticket-granting server.

To illustrate how Kerberos works, let's assume we want to access a data file stored on one of our company's primary servers. When we first log on to our workstation and request access to this file, an authentication server searches its database for our access rights. Once the server confirms that these rights include the requested service (i.e., we have permission to access the file), it generates an encrypted "ticket," which enables our workstation to access the ticket-granting server. The authentication server also returns the "key" that was used to encrypt something called an "authenticator," which contains our name, network address, and the current time. Our workstation then sends the ticket and authenticator to the ticket-granting server, which decrypts both pieces of data. If they match, the ticket-granting server generates a ticket for the requested service to be used only by us. This ticket is then returned to our workstation, which we then present to the company's server on which the file is stored. Once this server receives our ticket, it gives us access to the file.

A Kerberos-generated ticket is programmed to have a short life cycle (e.g., 1 hour or 1 day). This way, if an unauthorized person acquires a session ticket, it will only be valid for that time period. To use Kerberos, every network application has to be rewritten to support it. Additional information about Kerberos and other network security information can be obtained from http://www.securityserver.com. The Internet RFC that describes Kerberos is RFC 1510, which is available from http://info.internet.isi.edu:80/in-notes/rfc/files/rfc1510.txt. See also http:// www.rfc-editor. org/rfc.html for information about RFCs.

54. What security mechanisms other than firewalls are there to secure transmissions across the Internet?

One of the most frequently used strategies for protecting data transmitted across the Internet is *virtual private networks* (VPNs), which is discussed in Chapter 7. A VPN is an IP connection between two sites over a public IP network that has its payload traffic encrypted so that only the source and destination can decrypt the traffic packets. A VPN enables a publicly accessible network to be used for highly confidential, dynamic, and secure data transmissions. Of course, this type of security can be mostly implemented by encrypting files and other user data before transmission, but it is not quite as secure as a VPN. VPNs provide further security because they are capable of encrypting not only the actual user data but also many of the protocol stack informational items that may be used to compromise a customer site in a technical session attack profile. The current VPN exploitation that has emerged in the industry is centered mainly on IP-based networks such as the Internet. One of the major problems of VPN technologies is that there is a great variety of implementation styles and methods, which cause much confusion when trying to develop a strategy for their use in a company (see Chapter 7). Another problem with VPNs is their use of the Internet. The underlying technology of the Internet (see Chapter 3), namely, the Internet Protocol (IP), was never designed with security in mind. As a result, several VPN protocols have been developed to help secure VPNs. These include the *Point-to-Point Tunneling Protocol* (PPTP), *Layer 2 Forwarding* (L2F), *Layer 2 Tunneling Protocol* (L2TP), and *IP Security* (IPSec). Chapter 7 provides a description of these protocols.

55. How do I protect my connection from home to the office with VPNs?

Multimegabit technologies such as cable modems and *x*DSL (see Chapter 15) enable high-speed access from homes, remote offices, hotels, airports, and other locations. As such, the need for VPNs is greater than ever to allow remote computing and distributed connectivity. Attendant to this is the need to provide VPN and full firewall security. With "push" technologies, personal computers at the residence and on the road will be connected more hours of the day and, in many cases, continuously connected to access information from providers around the world. This also means that while the system that occasionally dials in to a public IP network is more difficult to crack, this is not always the case. Today, working with corporate documents at home entails being continuously connected to public IP networks and VPNs. Desktop firewall products will be instrumental in securing these environments. Chapter 18 contains information about securing the mobile enterprise.

56. Do I need cryptography for VPNs to work?

Not really. In fact, many implementations do not implement cryptography facilities because they slow down the connection between the systems using the VPN. Obviously, an encrypted link is more secure than one that is not encrypted. However, VPN, especially the tunneling varieties, don't need cryptography to provide the needed connectivity. Cryptography helps make the link more secure.

57. If network security is this complex today, what does the future hold?

As network capacity continues to increase and technology continues its relentless pursuit into more segments of our lives, network security is going to get even more complex, just as network protocols and operating systems must. New technologies and new techniques to provide authentication and access control will be coming out over the next few years, and this will cause methods of implementation of network security to change. Firewall facilities will be available on desktop, workstation, and server systems just as they are provided on network-to-network connections today. More electronic access to financial facilities will be forthcoming, and this means that the entire area of electronic commerce (e-commerce) is exploding and will continue to grow. Crackers and the threats they provide will increase as technology becomes more available to everyone. This means that the methods to provide network security must change to adapt to new threats. Consequently, the network security business will be a growing segment of the industry for a long time.

END-OF-CHAPTER COMMENTARY

This brings us to the end our discussion of network security. Please note that there is much more to network security than what is presented here. We do not have the space to address every aspect of network security in this book. This is best left for books specifically dedicated to this topic. You are encouraged to consult the Bibliography for additional information as well as the Web sites listed throughout the chapter. We also suggest you review Chapter 3, which discusses the Internet, Chapter 7, which contains specific information about VPN security protocols, and Chapter 15, which provides information about dialup and home networking issues. Finally, the next chapter, Chapter 17, contains information related to mobile network security, including various initiatives and protocols.

Chapter 17

Network Convergence

Network convergence involves the integration of different technologies and applications, which, when combined, create powerful new entities and products. One area of convergence that receives much emphasis today is the merging of data, voice, and video on the network. As we will learn later, network convergence is much more than simply combining these three data types. In this chapter, we introduce the concept of network convergence and address it from several different perspectives. An outline of the major topics we discuss follows:

- Network Convergence Overview (Questions 1–5)
- Impact of Convergence on Network Media (Questions 6–19)
- Network Convergence and Multimedia (Questions 20–26)
- Impact of Convergence on Businesses (Questions 27–41)
- Network Convergence at Home (Question 42)
- Network Convergence and Voice over IP (VoIP) (Questions 43–54)

1. What is network convergence?

Network convergence is a powerful concept and buzzword in networking circles today. Basically, it involves the melding of dissimilar network media, protocols, and applications to provide a network environment dramatically different than what was previously available.

2. Can you give me a practical example?

Sure. Did you ever experiment with food combinations when you were growing up and mom wasn't around to control your actions? For example, did you ever put vanilla extract in a cola or mix peanut butter with vanilla ice cream? Even Elvis had a penchant for peanut butter-and-banana sandwiches (or "PB and 'nanner sandwich" as he was said to have called it). If you were like most of us, you were working on your own personal convergence experience at an early age. By taking different entities and combining them in a new way, an entirely new product is produced. This is the essence of convergence. Sometimes the results are powerful; sometimes they just taste bad and need to be trashed at the first opportunity (e.g., tuna and peanut butter didn't work for us).

3. Actually, I was hoping for a practical example in the context of networking.

Oh! Sorry about that. In the context of networking, convergence represents combining different technologies and applications. For example, facsimile (i.e., "fax") technology represents the convergence of several technologies and applications, including optical scanning, computer printing, and data and voice communications. A major networking convergence issue receiving much attention today is *IP telephony*, which involves integrating voice and data. In IP telephony, digitized voice signals are converted to IP packets, which are then transmitted over the Internet. In this scenario, packet technology and the Internet are uniting voice and data such that traditional voice communication services are being delivered over an IP data network.

Consider for a moment the various systems and technologies that are common among many companies today. Most organizations or their employees rely on a(n)

- mainframe system with a very large corporate database, which contains enterprise information useful for a wide variety of applications;
- departmental servers that provide database sharing and access to the mainframe;
- corporate e-mail server(s);
- LAN-based fax server(s);
- desktop PCs;
- company PBX system for in-house telephony;
- personal cell phones with digital messaging and paging facilities;
- Web servers;
- Internet connection(s); and
- personal digital assistants (PDAs) such as a Palm Pilot™ system.

Without a doubt, each component on this list is useful in its own right. However, by combining different network types (e.g., LAN, WAN, wireless voice, and wireless messaging) along with the various components listed and uniting them with various applications and settings, some very powerful computational environments are possible.

4. This is twice now that you've said convergence can lead to something "powerful." Please give me an example of something you consider a "powerful computational environment."

OK. Consider the concept of "follow me anywhere" e-mail. By converging a cell phone's capabilities with a PDA and wireless messaging, and then using the corporate e-mail server in conjunction with message forwarding and a "where am I" phone server at an Internet site, an individual could receive e-mail on the PDA, cell phone, or other devices anywhere and at any time. This is possible by converging the available technologies into a cooperative network of functionality that produces an entirely new product. This type of service is not possible without the convergence of network technologies, applications, and data resources into a single delivery stream. Extending this concept further, a new cell phone introduced in Japan contains a built-in video camera with TCP/IP and modem technologies. This phone allows a user to connect to another phone via the Internet and conduct a slow video frame conversation between wireless phones. It also is in color and has a built-in mini-Web browser. These illustrations might seem very futuristic, but they are not. Such technologies are available at this writing and have been around for some time. As convergence continues in networking and applications, the melding of techniques and facilities will continue to produce some fascinating and exciting results.

Another current illustration of network convergence is the Bluetooth wireless technology, which interconnects various devices including computers, mobile phones, mobile computers, and handheld or portable terminals using short-range (approximately 33 feet, or 10 m) radio links. The Bluetooth technology is designed to facilitate wireless LANs (see Chapters 4, 15, and 18) so that networks of different hand-held and mobile devices can communicate and exchange data. One usage example is the three-in-one phone concept where a telephone functions as a portable phone at home, as a mobile phone when you are on the move, and as a walkie-talkie when the phone is within range of another Bluetooth-enabled telephone. Another usage example is the automatic synchronizer in which Bluetooth technology automatically synchronizes a user's desktop PC, mobile computer, notebook (e.g., PDA), and mobile telephone. Thus, mobile office workers, for example, upon returning to their offices, can have their current calendar, which is stored on their notebook device, update the calendar stored on their desktop PC. This kind of synchronization, where data can be exchanged without reentering it, is only one of Bluetooth's major application goals. A second major application area is e-commerce, where users can electronically pay for parking meters, bus tickets, shopping, and movies automatically through the use of Bluetooth-enabled devices. In short, Bluetooth technology fosters personal area wireless connectivity. The primary members of the Bluetooth Alliance are 3COM, Ericsson, IBM, Intel, Lucent, Microsoft, Motorola, Nokia, and Toshiba and include thousands of manufacturers throughout the world. It is the alliance's objective to embed Bluetooth technology into hundreds of millions of electronic devices by 2002 and have them operate at a globally available frequency band so they are compatible worldwide. Additional information about Bluetooth is available at http://www.bluetooth.com.

5. Now that's pretty cool. In the introduction you claimed that convergence is more than combining voice, data, and video, yet all the examples you've given so far do exactly that. Can you give me some examples of convergence that involve merging something other than voice, data, and video?

You bet. Consider the following recent developments and think about how they might affect computing and the networking marketplace:

Molecular Electronics Hewlett-Packard (H-P) announced research results in early December 1999 revolving around the concept of *molecular electronics*. After several successes at H-P and other collaborating institutions of higher learning, they stated that future processor technologies would not be limited to the current physical boundaries of transistor electronics. Molecular computing is based on the idea that switches, transistors, and logic gates—the basic building blocks of computer chips—can be constructed out of individual molecules. Given a molecule that is designed to stop or start the flow of electricity and connect it to wires that are only a molecule or so wide, the creation of processors and memory chips millions of times smaller than the fingernail-sized silicon chips currently in production now becomes possible. This type of technology will not only lead to massively powerful processors the size of a grain of sand, but it will also revolutionize computing and networking as we currently know it. From a networking perspective, think about the technology's impact on the size and ubiquity of networking components, as well as the need for massive networks to interconnect all the potential processing systems of a molecular size. The potential uses and the networking requirements for such an implementation are many. Some illustrations follow.

- *Medicine*. Injected into the bloodstream, tiny computers could traverse through the human body, destroying bacteria, identifying harmful cells, and diagnosing disease. Devices the size of a wristwatch, powered only by body heat, could store vast libraries of data and keep a record of images, facts, and events from a person's entire life.
- *Space Science*. Space probes that today weigh a ton might be the size of a baseball, powered by a peanut-sized battery, and travel to other solar systems.
- *Foreign Language*. Molecular computers could be used to translate the spoken word in any language on the fly.
- *Automobiles*. Vehicles could drive themselves, avoiding traffic jams and accidents, homing in on a destination while passengers sit back and relax.

Artificial Intelligence Computers can calculate the square root of 73,324,969 in a fraction of a second. However, put your PC next to any 3-year-old child and ask the two to identify the penguin in a picture of a zoo and there's no contest: The kid wins every time. Researchers around the world are at work on sophisticated mathematical techniques that might even the odds. They're applying probability theory to those problems computers have always been deficient in solving such as evaluating large amounts of input data (e.g., photographs) and making qualitative judgments about the data. The result is that by 2010, we'll likely see increasingly smarter computers that finally achieve some of the long-discussed but never realized goals of *artificial intelligence*: Computers will be able to carry on a conversation with you, recognize your face, read your handwriting, and help you search for objects. What's different is that researchers are more or less giving up on teaching computers exactly how to perform each individual task. That was the near-fatal failing of earlier artificial-intelligence efforts; they required humans to generate rules for everything (e.g., "If it's round, then it's a face," with myriad other qualifications).

Today, researchers are letting mathematical algorithms do the heavy lifting, so to speak, so that computers teach themselves. Instead of trying to describe what a face looks like, they give computers hundreds of digital pictures of faces and train them with software to develop their own increasingly accurate and flexible mathematical definitions of a face. These techniques have their roots in what's known as probabilistic inference, and a more recent refinement called *probabilistic networks*, which involve assigning numbers to degrees of uncertainty about each bit of data or possible outcome and drawing up rules about how you can manipulate those numbers. We're familiar with this type of probabilistic thinking when it comes to the weather—meteorologists talk about a 70% chance of rain, for example. Furthermore, our brains are thought to be constantly making decisions in a very similar manner, effectively using shortcuts to find the most likely answers, rather than perfect answers. Optical illusions offer a tangible example of this process in action: You can almost feel your brain "click" to a likelier way of seeing things when it figures out what's going on. Work in this area is not limited to obscure statistics journals or academics' offices. Already, Microsoft, for one, has incorporated several probability-based features into its products. Help software running in the background of some programs can determine what part of the program a user is working on and whether or not the cursor has moved recently, which might mean the user is stalled, before suggesting instructions. More sophisticated probability-based features are likely to come. At its Cambridge labs, more than half of the Microsoft researchers are working on projects that include at least some elements of probabilistic inference. Even Mr. Gates has said that probabilistic techniques are part of Microsoft's competitive advantage in Internet ventures, making its search engines and customized techniques more robust. Needless to say, probabilistic technologies are a solution to helping users communicate better with systems and software. It is quite obvious that the demand for networking will be greater than ever before. Who has all the answers to their information needs on a single machine? Also, what are the statistical odds that a user will require text-only information? Or, is the world really moving to a multimedia experience on the desktop for all types of information?

Wearable Computing Dressing smart will soon take on new meaning. Do you want to make phone calls on your necktie and e-mail friends by your sleeve? How about a shirt that prescribes medication or a suit that moves in response to light, heat, and pressure? Although this might sound like something James Bond's "Q" might design, this is the kind of clothing we will most likely be wearing in the new millennium. *Wearable computing*, a term coined at the Massachusetts Institute of Technology's Media Lab in the 1970s, often conjures up images of clunky headpieces or belts. However, as chip technology advances and computers get smaller, engineers say that so-called *wearables* will stop being accessories and will be woven into the fabric of our clothes. Instead of creating fabric that contains a complex network of lumpy connectors, such as the keyboard of a computer, companies are creating fabrics that have a continuous field of sensitivity. The individual keys dissipate into the threads, making the cloth sensitive to touch no matter where pressure is applied. To date, a team has created a tie that doubles as a mobile telephone, with a tiny microphone lodged into the knot and a keypad woven into the silk. In the next 5 to 10 years, we can expect to see clothes that not only detect light, heat, sound, and pressure, but can also emit them. Combine this with voice recognition and fabrics that respond to gestures and you have clothing of an entirely new order.

What does all of this have to do with the concept of network convergence? Well, it's pretty straightforward to predict that convergence will include voice and data. What is not so straightforward is the further convergence of other newer technologies onto networks that either are very new or not currently in production. Some technologies being deployed at this writing, such as the Bluetooth wireless interoperability standard, will have their work cut out for them when technologies such as clothing become part of the converged networked world.

6. OK. What impact will convergence have on the network environment?

We can examine this issue from two perspectives: the impact of convergence on network media and the impact of network convergence on network applications.

7. Media should be easy. Everything will be fiber-based. Right?

You're partially correct. In the long-term, (i.e., five or more years from now) we expect *long-range terrestrial networking* to be predominantly fiber-based (glass and plastic). However, *short-range terrestrial networking* will be predominantly wireless.

8. Will data transmission rates be able to keep up with computing demands?

We think so, as long as the research continues to be funded to develop the new networking materials. For example, TRW is using indium phosphide, which is an alternative base chemistry to the popular gallium arsenide method of chip construction. Indium phosphide is expected to have a top-end frequency range of 200 GHz; gallium arsenide is expected to peak at about 50 GHz. As a result, TRW recently announced a 69-GHz communications processor that will easily allow 40-Gbps connectivity over fiber. Wireless LANs also, over time, should be as fast as land-based technologies.

9. What do you think the impact of convergence will have on media in the short-term, say 2 to 5 years?

In the short-term we expect copper, fiber, and wireless media to predominate the network convergence efforts in the world. The application of these media relative to network type is not too difficult to predict. For example, LANs will continue to support copper media, and MANs and WANs will support different media types including fiber, hybrid coaxial/fiber, and wireless. A brief description follows.

LANs The ever popular Category 5 unshielded twisted-pair (UTP) cable, as well as Enhanced Category 5 (Cat 5E), will continue to lead the pack in the under 100–m land-based networking realm. Under development are Category 6 and Category 7 cabling standards, primarily designed to enhance existing Category 5/5E cables for higher-speed LAN technologies such as Gigabit Ethernet. Phone systems, copper-based video, and almost any data network can use Category 5/5E cabling at this writing. Category 5 UTP does have its limitations, however. The most common problem is cable length. The longer the cable, the greater its ability to receive unwanted signals, and long cable lengths foster increased buildup in signal resistance. There is also the need to support faster networks such as high-speed ATM and Gigabit Ethernet. Both technologies have very short transmission wavelengths, which result in restrictive UTP cable lengths. In addition to wire-based LANs, we also expect a strong presence of wireless networking in the LAN environment so that docking station computing (e.g. laptops, notebooks, and PDAs) will interoperate via infrared wireless connectivity or microcell radio wave-oriented network facilities for short distances in an office environment.

MANs Metropolitan area networks are still mostly fiber-based, but new wireless technologies and the strong emergence of broadband will change the complexion of this space significantly for small businesses and residential networking consumers. Already in Canada, over 20% of all residential telephone connections are via broadband connection (also known in residential circles as "cable TV connectivity"). In the United States, the massive purchasing of cable television providers in 1999 by AT&T will result in a large and steady push to provide voice and data access via broadband as well as existing broadband video facilities. This will enable broadband companies such as AT&T to bypass the traditional telephone company's copper-based local loop and establish their own self-managed cable path to the consumer. Broadband networking has the ability to provide massive network infrastructure capabilities for a variety of needs in a metropolitan area (1–100 km range). This ability will allow small companies to gain services similar to large companies. It will also enable small companies to compete in global markets historically dominated by companies that are able to provide network infrastructures for interconnection to their customers and suppliers. Ubiquity of MAN media technologies will change the way that networks will affect small- and medium-sized businesses in a profound manner.

WANs Wide area networks currently employ several different media types, including leased lines, fractional T-carrier, T-carrier, SMDS (in Europe), and frame relay. Major growth over the next 2 to 5 years will include enhanced frame relay facilities, SONET, and synchronous digital hierarchy (SDH). In Holland, for example, a 40-Gbps SDH backbone is operational throughout the area known as Maasflacht, near Rotterdam. This very high-speed WAN provides connectivity throughout the Europort docking and terminal facilities within southern Holland at a very reduced price compared to previous connectivity options. In the United States, large companies such as AT&T and MCI WorldCom are installing or expanding existing SONET connections to provide national SONET facilities to replace or augment existing frame relay facilities. Through the use of transnational SONET, the convergence of high-speed networks becomes a reality for city-to-city and site-to-site connectivity. One interesting development is in the area of gigabit connectivity for WANs. The current 10-Gigabit Ethernet proposal (IEEE draft 802.3ae) provides for a 10-Gbps connection speed over WAN facilities. This is expected to become very popular for network service providers in lieu of using frame relay, SONET, and other high-speed network solutions due to its direct compatibility with Ethernet LANs (see Chapter 8).

GANs Global area networks are following the trend of WANs except that larger multinational network providers are taking over the predominant positions in the market to become a "one-stop supplier" of services to companies requiring multinational connections. Long-term and very long-term networking trends are all looking toward a fiber-based terrestrial backbone with interconnections to wireless methods for various networking needs. Copper is here and will be around for many years, but there are plenty of countries (including very large ones such as Russia and China) that have no wiring infrastructure and where it does not make any sense to create one when wireless facilities will do the job nicely and with fewer problems. Some areas of the world will migrate to almost pure wireless networking environments for all converged services very quickly. For the more "wired" countries in today's market, the migration to pure wireless overall will be slower.

10. OK. You basically confirmed what I suspected, namely, that wireless technology will be an integral component of convergence and that fiber will continue to prevail. The other issue is applications. What kinds of converged applications can I expect to see?

Before we respond to this question, let's review the current situation. In today's network environments, application technologies are usually independent of one another. For example, e-mail systems do not collaborate to a great extent with other applications used on networks. Accounting systems do not integrate with network management. Network performance and quality of service measurement facilities do not integrate with office information systems. Yet, there are useful components of all facilities, which, when abstracted among various application environments, become useful to all applications environments on a network.

In a converged environment, traditional networked applications will eventually merge to provide richer and more feature-filled applications that are limited only by the imagination. Applications will evolve into new systems of interactivity that are radically different from those seen today. E-mail, voicemail, videomail, and hyperlinked documents are obvious candidates for convergence because they all convey specific types of information in a manner that can be understood by the same destination entity, namely, a human recipient. Microsoft's Outlook 2000 program is a good illustration of the initial steps in this direction toward the concept of integrated messaging. Outlook 2000 can hyperlink with MS Word, the MS Internet Explorer Web browser, contact management software, videoconferencing (via NetMeeting add-ons), and fax facilities. Third-party offerings also enable Outlook 2000 to interact and transfer voicemail between systems and e-mail delivery facilities. While neither is totally cohesive nor seamless in user interface, it is a start that shows the direction of desktop application convergence.

Many other application systems, including enterprise resource planning (ERP) facilities, accounting, electronic data interchange (EDI), business modeling, and traffic management—all of which are designed to provide intra- and intercompany assets and inventory exchanges—are becoming "Webified." That is, access to these applications is being provided in an integrated manner via Web-like interfaces that facilitate a distributed access method throughout an organization, regardless of corporate personnel or location. For example, upper management executives are able to see what they need to see, while at the same time and via the same system hierarchies and subcomponents, the loading dock personnel can manage product inflow/outflow. Thus, application information converges to the user depending on the user's perspective of the information provided.

11. Can you give me a specific Web-based convergence application?

Sure. Consider the recent Internet technology called *Enum*, which is an IETF standard defined in RFC 2916. Enum is a protocol that maps E.164 telephone numbers (see Chapter 13) to a Uniform Resource Identifier (URI) (see Chapter 3). In more simple terms, Enum enables users to use telephone numbers to access Internet services, including Web pages, instant messaging, Internet-based faxes, and Internet-based phone calls referred to as voice over IP, or VoIP (discussed later). Enum-enabled applications demonstrate the convergence of the public switched telephone network and the Internet.

12. Tell me more about this technology. For example, when did this happen and why is it based on telephone numbers?

First, the service is not widely available as of this writing. There are some trials under way, but mostly these are private in nature. (See http://www.enumworld.com for more information about these trials.) Enum's genesis dates to 1993 with the development of Internet fax service (see RFCs 1528 and 1529), which enabled people to send faxes over the Internet via their telephones. Although Internet fax service was not widely implemented, it did bring to light the idea of using telephone numbers to access Internet services. Given that there are literally billions of telephones worldwide and that telephone numbers today enable people to access a host of services, including voicemail, fax, and name/address lookups, it is only natural that the use of phone numbers evolve to include accessing Internet services. This became apparent with the development of VoIP in which telephone calls are carried via an IP-based network such as the Internet. Enum enables people to collapse all of their separate contact information such as their home phone number, cell phone number, work phone number, fax number, e-mail address, and Web address into a single contact number.

13. How does this work and what types of applications does it support?

Enum is a transparent technology. In other words, you will neither do anything specific to invoke it nor will you see it work. Accessing an Enum service is analogous to accessing a Web page or sending an e-mail message. Instead of entering a specific Web or e-mail address, though, you simply enter a telephone number. Central to Enum is the Naming Authority Pointer (NAPTR) records, which are defined in RFC 2915 and maintained by your service provider. A list of the registered services that correspond to a specific phone number are saved in these records. As stated earlier, examples of the services a person can register include VoIP, fax, e-mail, instant messaging, and Web pages. Using a mechanism similar to that used by domain name service (DNS, see Chapter 3), the telephone number you enter is translated into a fully qualified domain name (FQDM). Enum then issues a DNS query on this domain to find an authoritative name server that contains the NAPTR record associated with the phone number. This record is then examined for the requested service and the uniform resource identifier (URI, see Chapter 3) associated with that service is then retrieved. Thus, you enter a phone number and in response to this entry a URI is returned based on the specific service requested.

14. I'm still confused. First, is the number I enter a regular telephone number?

Yes. The number is indeed a "regular" telephone number as you know it. More specifically, it is a standard E.164 number. The E.164 designation means that the number follows the format and structure of the international telephone numbering plan, which is administered by ITU-T and specified in a document labeled "E.164." Addresses are a maximum of 15 digits and have a geographically hierarchical structure, which makes them well-suited for worldwide application. The structure of a fully qualified E.164 number is

country code   area or city code   local phone number

- *Country codes* are up to three digits long and are only used internationally. The convention when writing a country code is to precede it with a + sign. Country codes consist of a zone code followed by one or two national identifiers. As an example, consider the country code for Italy, which is +39. The +3 designates the zone, in this case, Europe, and the 9 designates the country, namely, Italy. Europe actually has two zone codes: +3 and +4. As an exercise, consult the international calling pages of your telephone book and look at the country codes of different European countries such as France (+33), Greece (+30), Germany (+49), Ireland (+353), and Iceland (+354). Note that these respective country codes all have a leading digit of +3 or +4. The remaining digit or digits of the country code refer to the national identifier, which specifies the country itself. Further note that the United States and Canada do not have a country distinction (i.e., a national identifier). These countries only use the +1 zone code and hence, don't have a "real" country code. This is because they follow the North American Numbering Plan (NANP), which was in place before international country codes were formally assigned.

- *Area* or *city codes* are up to three digits long. The United States and Canada use area codes; other countries use city codes. Thus, in the United States and Canada, area codes effectively serve as "national identifiers."
- *Local phone numbers* consist of a three- or four-digit prefix or exchange code followed by a four-digit line code.

As an illustration, let's say that the local telephone number of a hotel in Florence, Italy is 234-3201. This means that its fully qualified E.164 number is +39-55-234-3201. As another example, let's assume that a U.S. business in Rochester, NY (area code 716) has the local phone number 442-2890. Then its fully qualified E.164 number is +1-716-442-2890. A feature of Enum is that you do not have to enter a fully qualified E.164 number. You simply enter the same number you would normally dial. Enum will then convert this number into its fully qualified form. Enum also enables you to enter 4-digit office extensions within a company.

15. What about 911?

Emergency services numbers such as 911 in the United States and 112 in Europe are considered special access codes and are not part of the E.164 numbering structure.

16. I never realized there was actually a well-defined structure to phone numbers. You've helped clear up part of my confusion. I am still confused about Enum services, though. Where do I enter a number and how does the system know which service I'm requesting if multiple services are registered to a single number?

The number is entered in an application program such as a Web browser or e-mail program, or an Enum-enabled device such as a telephone (Figure 17.1). For example, if you are using Internet Explorer or Netscape, you simply enter the phone number instead of the Web address (i.e., the URL). If you are using an e-mail program such as Outlook or Eudora, you enter the phone number on the "To:" line instead of the person's e-mail address. If you are using a telephone, you just dial the number. The service that is being requested is a function of the device or application you are using to enter the phone number. To illustrate this, let's assume that your Internet service provider, with your authorization, has registered your e-mail address, your fax number, your personal Web page address, and your business telephone number, which is VoIP-based. Now, if we want to send you an e-mail message, we simply invoke our Enum-enabled e-mail application and enter the single contact number that corresponds to your registered Enum services. Our e-mail application converts the

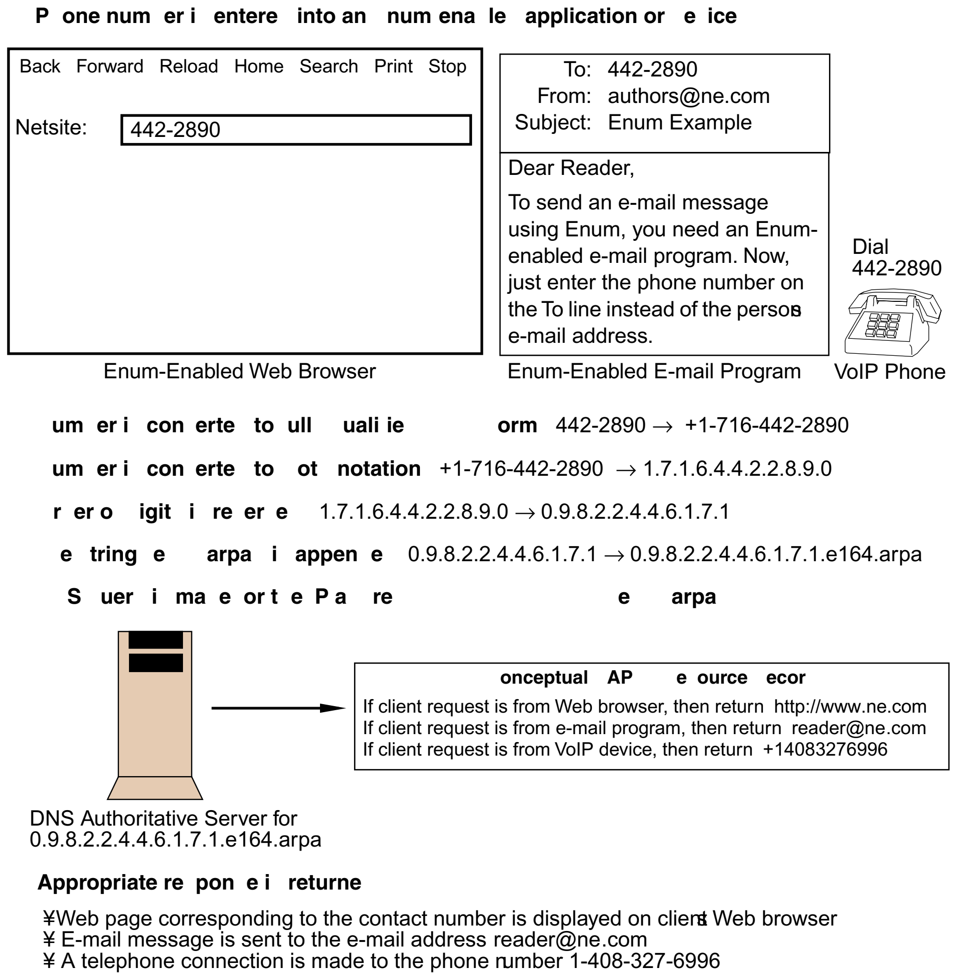


FIGURE 17.1 An example of how Enum works. Note that in steps 3–5, the phone number is resolved to a special IP address format that is used for the DNS query. For more information on how this is done, see RFC 2916. Further note that the NAPTR resource record shown in step 6 conceptually depicts the mapping between a phone number and the registered services. Specific information about the format and contents of these records can be found in RFC 2915.

phone number we entered into a fully qualified E.164 number, which is then translated into a fully qualified domain name (FQDM). The Enum protocol issues a DNS query on this domain to find an authoritative name server that contains the NAPTR record associated with the phone number. This record is then examined for the requested service. Because the originating Enum-enabled application was an e-mail program, the record that corresponds to your e-mail address is returned and our message is directed to your e-mail address. Similarly, if we entered your contact number in our Enum-enabled Web browser, then this number is mapped to your Web page address; if we dialed your contact number on our Internet phone or Enum-enabled standard telephone, then the number gets mapped to your VoIP phone number at your office. Note how Enum converges the Internet and PSTN.

17. Oh! I get it. The intelligence is in the application or device.

Exactly.

18. What happens if a specific service isn't registered? For example, suppose you try to access my Web page but I don't have one?

In this case, we would receive the standard 404 "dead link" error code. If it were an e-mail message, then we would receive an error notice (e.g., "unknown recipient" or "address not found"). In the case of a VoIP number, the protocol will attempt to make a conventional connection using the public switched telephone network (PSTN) as if it were a regular telephone call.

19. Well, that is pretty cool. So this makes it possible for me to list a single number on my business card and people can reach me via any medium (telephone, fax, e-mail, Web) just by entering my number.

That's right. For more information see http://www.enum.org, RFC 2915, and RFC 2916. One final note: As alluring as Enum is, it also is a form of centralized computing. Thus, if the authoritative DNS server for your contact number crashes, or some DNS administrator inadvertently removes your entry, then all of your contact information will be lost because the server represents a single point of failure.

20. OK. Let's move on. Tell me about another network convergence application.

Well, a very popular one is *multimedia networking*, which involves transmitting information as a combination of full-motion video, sound, graphics, text, and animation across a network. Nearly all desktop computer models sold today are multimedia capable. They are equipped with speakers, microphones, sound cards, video cards, high-resolution monitors, and DVD drives. As stand-alone devices, PCs deliver rich, full-feature multimedia output via interactive games, music CDs, and the like. The delivery of multimedia information across a network, however, is another issue. Multimedia applications generally can be classified as prerecorded, live, or real-time interactive.

- *Prerecorded multimedia* implies that the "action" was recorded earlier and a user is simply viewing or listening to it in a non-real-time manner. Two examples include prerecorded music CDs and prerecorded DVD movies, which conventionally involve acquiring the CD or DVD medium and playing it on your computer or dedicated device such as a CD or DVD player. Prerecorded multimedia can also be download from the Internet as prerecorded multimedia files. Internet downloads of multimedia files can be effected by a conventional download method or by a strategy known as *streaming*. A conventional download entails waiting until the entire file is received before playing it. In many cases, conventional downloads are time consuming and frustrating because the files are usually extremely large in size (e.g., 5 or 10 megabytes) and because you cannot play back the file until you receive it in its entirety. If the file is not what you wanted, then the download wasted both your time and bandwidth. Streaming downloads, however, enable you to view or listen to a multimedia presentation within a few seconds of receiving parts of a file from a particular download. Commonly referred to a *streaming audio* or *streaming video*, this download strategy enables a user to almost immediately engage in a multimedia session using one part of the file while the other parts of the file are still being received. Thus, it is not necessary to wait until the download process is completed. The concept of streaming is analogous to having a house party. For example, let's say your party is scheduled from 10 A.M. to 6 P.M. Around 10 A.M., a stream of guests arrive and you immediately interact with them. While you are greeting and entertaining this group, other guests are en route to the party and upon their arrival you greet and interact with them as well. Thus, throughout the 10 A.M. to 6 P.M. time period you are constantly greeting and entertaining a stream of guests as they arrive. Streaming requires using special software such as RealPlayer from RealNetworks and Microsoft's Windows Media player.

- *Live multimedia* refers to viewing or listening to a live broadcast of a radio or television program. The difference is that the broadcast is being carried by the Internet and not via a conventional radio or television transmission. Live multimedia content is not prerecorded, which implies that it is not stored on a server (although it can be recorded and made available at a later time). Live multimedia is received using streaming. Many radio stations, for example, now transmit their programs using streaming audio. A good place to sample this concept is at RealNetworks' Web site (http://www.realnetworks.com).
- *Real-time interactive multimedia* involves live and real-time interaction. Unlike the previous modes, this means that the recipient is able to interact, in real-time, with other participants. Videoconferencing is one example of real-time interactive multimedia. LAN- and WAN-based interactive multimedia networking has obvious applications in education because of its potential for delivering real-time educational content to the desktop. However, it also creates problems for the network.

21. I'm familiar with prerecorded and live multimedia but I have no experience with real-time interactive multimedia. What kinds of problems does interactive multimedia networking create?

There are two primary issues with interactive multimedia networking. First, data streams are extremely bandwidth intensive. These data streams must carry video information as well as audio information that is synchronized to the video. Second, multimedia is isochronous, which means that the data stream must be delivered as a continual and uninterrupted flow of data. Isochronous communications (see Chapter 1) are normally not a problem in a circuit-switched environment. However, in a packet-switched environment such as the Internet, delivering interactive multimedia across the network requires a strategy that will guarantee regular, continuous, and timely data delivery. That is, it requires a quality of service (QoS, see Chapter 5).

22. Aren't there standards that address these issues?

Yes. During the past few years several standards have been developed that facilitate Internet-based and LAN-based interactive multimedia networking. These standards developments, which address the bandwidth and isochronous communications issues, include the Moving Picture Experts Group (MPEG) series and the H.320 standards family.

23. What can you tell me about MPEG? I'm interested in learning something about it because I have an MP3 player and I download MP3 music all the time.

MPEG is the nickname of an ISO working group that was formed in 1988 to develop international standards for compression, decompression, processing, and coded representation of full-motion video and audio. The standards produced by this group are used in a wide variety of applications, including your MP3 player and MP3 music files. MPEG's official name is ISO/IEC JTC 1/SC 29/WG 11, which stands for "International Organization for Standardization/International Electrotechnical Commission Joint Technical Committee 1/Subcommittee 29/Work Group 11." MPEG members meet several times a year. Meetings are usually attended by more than 300 experts from over 200 companies with representation from an average of 20 countries. The compression techniques the group has developed, collectively known as the MPEG series, reduce multimedia data streams into smaller-sized entities for efficient transmission across the network. The series includes MPEG-1, MPEG-2, MPEG-4, MPEG-7, and MPEG-21. All versions except MPEG-21 are ISO/IEC standards. A brief description of each follows.

MPEG-1 MPEG-1 consists of five parts: systems, video, audio, conformance testing, and reference software. The first part specifies how audio and video data streams are to be multiplexed and synchronized. The second part addresses the video component and specifies how video data streams are to be encoded to support VHS-quality transmissions at T1 rates. The third part addresses the audio component and specifies how audio data streams are to be encoded. Collectively, parts 1, 2, and 3 provide for the efficient storage and retrieval of stereo audio and full-motion video on compact discs (CDs). The fourth part describes the methods vendors must use to ensure that their products conform to the standard. Sample bit streams are also provided for testing purposes. The last part contains the actual computer code of the standard. Officially known as ISO/IEC 11172, MPEG-1 was approved in piecemeal fashion: parts 1, 2, and 3 were approved in 1992, part 4 was approved in 1993, and part 5 was approved in 1994. MPEG-1 has been implemented in a variety of products and applications, including MP3 devices and related software programs. An MP3-based application takes a specific track from an audio CD and compresses it using MPEG-1's audio layer III (thus the name MP3). As you probably know, MP3 has spurred the creation and growth of an entirely new segment within the music industry. Thanks to MP3, consumers can now download specific audio tracks via the Internet and can compile custom-made CDs. Furthermore, unlike conventional audio CDs, which can store a little more than an hour's worth of prerecorded music, MP3-compressed music files can provide approximately 10 hours of music on a single CD.

MPEG-2 MPEG-2, known officially as ISO/IEC 13818, consists of nine parts. The first five are similar to those of MPEG-1 and hence address systems, video, audio, conformance testing, and reference software issues. Part 6 specifies the protocols for establishing sessions across disparate networks; part 7 provides a new multichannel audio coding that is different from and incompatible with MPEG-1 formats; part 8 is defunct; and part 9 specifies the interface between an MPEG-2 transport system and decoder. MPEG-2 provides compression levels as high as 200 to 1 and supports multimedia transmissions ranging from 4 to 100 Mbps. It is the standard for high definition television (HDTV) applications, is used in digital television set top boxes, and is the backbone of digital versatile disks (DVDs). MPEG-2 is the recommended standard for network convergence applications.

MPEG-4 MPEG-4, known officially as ISO/IEC 14496, is designed expressly for multimedia applications, especially in small multimedia devices such as videophones. MPEG-4 is being utilized for video on demand applications and mobile communications. Recent extensions to the standard include support for multiuser applications, 3D animation, and transmission rates of 1 Gbps for studio applications.

MPEG-7 Known officially as ISO/IEC 15938 and approved in 2001–2002, MPEG-7 is designed for searching, managing, and filtering multimedia information. MPEG-7 is different from previous MPEG standards because instead of specifying protocols for representing actual audio or video data, it provides information about how the audiovisual data are being represented. This enables fast and efficient searching for targeted material. The need for this standard is predicated on the increased use of multimedia in nearly all aspects of society and the desire to rapidly search for multimedia information. MPEG-7 enables users of digital libraries, for example, to search for a specific music file by entering a few notes of the song on a keyboard. We can search for all of Martin Luther King's speeches simply by entering a sample of his voice or a photograph of him. We can search film or radio archives for favorite movies, television programs, or radio broadcasts by simply entering some distinguishing characteristic of the desired entity regardless of the form (text, audio, video). In short, MPEG-7 facilitates rapid retrieval of multimedia information using multimedia data.

MPEG-21 MPEG-21 is a new standard currently being developed. Its purpose is to provide an overall multimedia framework that will ultimately describe how all the elements of a multimedia-based infrastructure relate to each other. The goal is to address all of the issues related to the design, delivery, receipt, and use of multimedia data. It attempts to view multimedia systemically, from content creators to end users, in the hope that the resulting multimedia environment will be seamless across all different types of networks and community of users. For more information about MPEG-21, as well as the other MPEG versions, see http://www.cselt.it/mpeg.

24. Is MPEG the same thing as JPEG?

Not really. One starts with an M and the other starts with a J. Seriously, though, although they might sound the same, JPEG and MPEG are not the same. JPEG, which stands for Joint Photographic Experts Group, is for still image compression, not multimedia or full-motion video, which is MPEG's focus. JPEG is designed for compressing full-color or grayscale images and works very well on photographs and artwork. It doesn't work that well, though, on lettering or line drawings.

25. How does H.320 compare to MPEG? In other words what does H.320 provide?

The H.320 family consists of several different standards that were developed for distinct networking environments. Individually, these standards—H.320, H.321, H.322, H.323, and H.324—represent a suite of standards that specifies audio/video compression and multiplexing methods, data framing, control information (e.g., call setup), the type of communications interface, and other related issues. Collectively, though, these standards represent the H.320 family of standards. A brief description of each follows.

H.320 H.320 is a 1990 ITU standards suite that was developed primarily to support videoconferencing and multimedia communications over circuit-switched networks such as ISDN (Chapter 11). H.320 includes a video compression algorithm called H.261, which is known as Px64 and designates multiple 64 kbps transmission rates. As a result, H.320 is sometimes referred to as Px64.

H.321 This is a 1995 ITU standard for adapting H.320's suite of standards to broadband ISDN networking environments, including ATM.

H.322 This is a 1995 ITU standard designed expressly for delivering QoS capabilities to LANs such as IsoEthernet (Chapter 8).

H.323 (Version 2) This version of H.323 is a 1998 ITU standard for packet-based multimedia networking. More specifically, H.323 establishes standards for videoconferencing and multimedia communications over local area networks (LANs). This standard is receiving the most attention currently in multimedia networking circles because its focus is on providing QoS capabilities to Ethernet/802.3 LANs as well as the Internet. H.323 is nondevice and non-OS specific. Thus, it can be implemented in a variety of devices that support different OS platforms. H.323 supports IETF's Real-Time Transfer Protocol (RTP) for delivering packet-based audio and video streams across the Internet, as well as the Real-Time Control Protocol (RTCP) for maintaining QoS information. H.323 also supports IP multicasting, which enables multiple destinations to receive the same multimedia stream simultaneously without having to retransmit the stream multiple times to each destination node. The first version of H.323, which was approved in 1996, did not support IETF's Resource Reservation Protocol (RSVP). However, with Version 2, end nodes can now set QoS via RSVP. The second version also provides support for URL-type addressing, as well as improved audio and video capabilities.

H.324 H.324 is a 1996 ITU standard that specifies low-bit rate multimedia transmissions over analog modem (V.34) dialup connections. In other words, it supports high quality audio and video compression so that data, voice, and video can transmitted over high speed modem connections using the conventional telephone network. It also addresses interoperability issues, which should pave the way for different vendor "videophone" products to interoperate with each another.

With the release of H.323/Version 2 in 1998, the convergence of data, voice, and video via multimedia streams is spurring the development multimedia-based applications and the deployment of interactive multimedia networking. As advances in this area continue, distributed real-time interactive multimedia will become more commonplace within the LAN/MAN/WAN environments. Furthermore, with broadband technologies such as *x*DSL and cable modems permeating the residential marketplace, real-time interactive multimedia networking's presence will be felt in this area as well. Work on H.323 continues at this writing, with Version 3 having just been completed and Version 4 currently being drafted.

26. Given the current body of multimedia standards, does this mean that any organization can now engage in multimedia networking?

Not necessarily. Although standards are now available that resolve the two major hurdles of true real-time interactive multimedia networking, this does not mean that organizations are prepared to deploy this technology onto their LANs. To support LAN-based multimedia networking, organizations need to examine their local network infrastructure carefully to ensure that it can support multimedia's impact. Specifically, a review of the overall infrastructure should consider capacity and performance issues (e.g., What are the multimedia application performance requirements and does the current infrastructure have sufficient bandwidth to support the applications?) and management resource issues (e.g., Are current network management tools sufficient?). Additionally, firewall configurations should be examined to ensure that they will pass multimedia data streams, and routers should be configured (if not already) to support QoS as well as IP multicasting.

27. There's a lot more to network convergence than I initially thought. Your last response triggered my next question. What effect will convergence have on businesses besides what you mentioned in your previous response?

As network convergence begins to establish its presence in today's business networking environments, its effects will be pronounced and impact nearly all aspects of networking. These include an organization's legacy networks and systems, the computing and networking styles currently being used by an organization, and the network performance and growth rate of an organization's overall network infrastructure.

28. How will convergence affect legacy networks and systems? I would think these would simply be replaced by newer technology that can support convergence applications.

In an ideal world in which money was of no concern, this would be true. However, not all organizations have a maintenance and replacement budget for computer or networking hardware and software. Thus, in most corporate networks that have existed for more than 10 years, there are legacy systems that continue to support key functions. These systems comprise special purpose operating system environments or employ minicomputers, super-minicomputers, mainframes, or supercomputers. Some of these systems are highly specialized or have very large-capacity capabilities that are simply not available from other vendors, and most use non-Windows and non-UNIX-based operating systems. For example, minicomputers that use Data General's AOS or Digital Equipment Corporation's (now Compaq) RSX-11M, IAS, RSTS/E, or RT-11 operating systems provide access to 8- and 16-bit computing hardware. Super-minicomputers such as the VAX and Alpha processors use UNIX variants and OpenVMS. Specialized middle-range systems such as Tandem have operating system environments such as VLX and NonStop. IBM mainframe systems have historically employed MVS and VM/CMS as their operating systems. Cray and Fujitsu supercomputers use COS, LTCS, and a UNIX variant called UNICOS. In some cases, server-based architectures such as Novell Netware provide specialized operating environments with special purpose protocols such as Novell's IPX and SFX. Even MS Windows environments still use a proprietary protocol for file and printer sharing called Network Basic Extended User Interface (NetBEUI). All of these systems have networking capability with vendor proprietary protocols. System protocols such as DECnet, Banyan VINES IP, Systems Network Architecture (SNA), Local Area Transport (LAT), and Local Area VAXCluster (LAVC) operate over specialized hardware or LANs such as Ethernet/802.3 and token ring. Some protocols such as DECnet and SNA have been used on hundreds of thousands of middle-ware systems for many years and are so entrenched in existing applications that only a complete replacement of the applications and system types will provide for removal of the protocols embedded in the applications. In other words, you would have to "sweep the floor clean." Like automobile engines, you cannot replace parts from different auto vendor's engines without a lot of compromises to the engine itself and without a lot of hassle. Over time, the only logical step is to replace the whole thing with a more holistic, unified vendor approach. Legacy networks will be profoundly affected by convergence because many components must be replaced to enable interconnectivity between new systems and legacy systems.

29. I see your point. We still have some old 68000-based Macs where I work. What about the second area you mentioned, namely, computing and networking styles? How will convergence impact this?

In the not so distant past, it was commonplace to have a computer system that was self-contained, that is, no network resources whatsoever, not even for printing. Now, it is inconceivable to provide computing resources without network resources. The two have truly become inseparable. As the melding of network technologies and computing technologies progresses, so has the downward scaling of computing into hand-held personal digital assistants (PDAs). These devices are often as powerful as desktop computing systems but are physically small enough to be placed in one's pocket. With high-density storage, a PDA can contain a great deal of information. With the convergence of terrestrial and wireless networking, these units truly become limitless in their ability to supply information to the user. In fact, most PDAs now contain more computing power than the original PCs released in 1984. It is not impractical to expect these devices to become the mainstay computing power for most individuals in the future.

As noted earlier, some companies such as IBM are working on technologies called "wearable computing." Wearable computers are expected to become extremely ubiquitous over the next few years. It will not be uncommon to find computing technology located in clothing, automotive products, household appliances, and building automation devices. Potentially, even something as trivial as a candy wrapper might sport a small microprocessor to provide date of purchase and storage information. Obviously, having a desktop or mobile PC is not enough for this type of computing style. The reality is that the "network is the computer," which implies that the integration and wealth of network resources make the network the most critical element in a converged environment.

At the same time that convergence is melding networking and computing systems, there is also a convergence among copper-based, fiber-based, and wireless systems in the networking environment. Personal digital assistants are being used in environments where traditional notebook paper was used in years past. It is also possible to purchase a wireless "always-on" Internet connection that allows a Palm Pilot with the proper hardware to communicate with an Internet service provider in certain North American cities. By integrating an Internet connection with a voice recognition system, the Palm Pilot becomes a multifunctional communication system that does not require a keyboard or even a pen to input data. Keyboards, mice, and other input devices become obsolete because the system is capable of communicating via voice recognition. By adding capabilities such as infrared transmission between the Palm Pilot and a cellular telephone, data stored on the Palm Pilot can now be exchanged among the Palm Pilot, the cellular telephone, and a PC. Although this is one of the application areas of the Bluetooth technology initiative (Chapter 18) using a wireless environment, the same effect can be achieved through a variety of network media including copper and fiber. In this brief example, current technology allows the convergence of computing and networking platforms as well as a variety of networking transmission media. It does not take a large leap of logic to expect that the Palm Pilot will eventually allow all the functionality that a cell phone would allow in addition to the range of items that the PDA allows now. The original architects of the Palm Pilot have started up another company called Handspring, which makes a competing and 100% compatible system called Visor that uses the existing products and software for a Palm Pilot and attachments such as video cameras. Extrapolation of taking the Palm Pilot technology and merging it with traditional computing capabilities, extended network facilities including wireless capabilities, and adding attachments such as video cameras all provide for the convergence of the technologies into something new that has not existed in the previous technical spectrum. This example of convergence clearly shows how data, voice, and video are on the marching path to becoming a singular technical solution in the future.

30. The third area you said convergence will impact businesses is network performance and growth rate. In what way do you foresee this happening?

Historically, data, voice, and video have been transmitted over separate and distinct networks within a corporate environment. With convergence, these separate networks are being merged into a single network as a means of simplifying network management, access, and cost. Undoubtedly, this convergence will also increase network traffic. Whenever a technology causes an increase in network traffic, there's going to be a corresponding network performance problem. The question then becomes which network is affected based upon the type of technology being used. As the convergence of data, voice, and video becomes more of a reality, a single network architecture will most likely be implemented at most companies, and that network will be affected by all three application types. For example, in data networking, performance is largely governed by packet size, packet arrival rate, connection speed, and network type. Variances of any of these parameters cause the network to perform in different and often unpredictable ways. Some applications in data networking are very predictable in performance, but others are not. Some applications use stateful protocols, some use stateless protocols, and others use ACK/NAK protocols, all of which have completely different performance variances and tolerances. Some applications require continuous-bit rate (CBR) transport capabilities, others require variable-bit rate (VBR), and still others a combination of both. This makes planning and implementing a data network a rather difficult exercise in performance management. Think about how much more difficult this task will be once voice and video are a part of this network.

31. How is this different from voice networks? Aren't they planned for in the same way as data networks?

The science of planning conventional voice networks and predicting their performance is much more structured than that for data networks. Conventional voice networks are usually very predictable because there are standards and other types of constructs that limit their speed and capabilities. So, no, voice networks really are not planned for in the same manner as data networks.

32. OK. But if you're going to incorporate voice onto a data network (e.g., voice over IP), then doesn't a VoIP network get planned the same way as a data network?

Many voice networks are indeed becoming voice over IP (VoIP) networks, and yes, VoIP networks are indeed planned for and configured similarly to data networks assuming that the data networks were created as standalone entities. The reality is, though, that most VoIP networks are configured to be partnered with a data network or inserted onto an existing data network. This means that many of the performance modeling and performance estimation problems that exist for data networks now exist for voice networks. It also means that many of the problems that exist for the IP protocol now exist for voice connections as well.

33. What about video networks?

Video networks to date have always been a separate network since they usually involve broadband technology or high-speed network connections to compensate for the video frame speed requirements. In the last several years, new video technology that reduces the frame rate from 30 frames per second to 12 to 15 frames per second has been making its way into the marketplace as an acceptable videoconferencing solution. The result is that connection speeds of approximately 384 kbps are more than adequate for the slower speed frame rate. As new compression technology begins to appear in the desktop videoconferencing area, the ability to achieve 30 frames per second over a standard data network becomes reality. This means that the days of dedicated networks for video are gone and replaced with the opportunity to operate a data network using video services. This is similar to what is happening in the voice area and brings with it the same performance and congestion issues.

34. Are some of the issues of combining data, voice, and video mitigated somewhat if they are all running IP?

To some degree, yes. By using IP, all three network application types can coexist on the same physical network structures. The key to coexistence of all three application types on the same network, however, is the need to supply a large enough data pipe to satisfy performance requirements of the applications. The deployment of 1 Gigabit Ethernet as well as 10 Gigabit Ethernet should address this issue. Speed, however, is only part of the picture. Not only must the network be fast enough, it must also support the burst rate required to keep all applications happy. Burst rate performance is the ability for the network devices in the path to deal with a constant burst rate of the highest performance rate of the network for a period of time in such a manner that the network does not lose the traffic as a congestive failure. This means that network hardware and software must have sufficient memory allocated and available to deal with the instantaneous traffic arrival rate in any particular location on the network. While doing this type of estimation is not terribly difficult, it is different from what most network managers actually do when designing and implementing a network to deal with any particular data type. Very few network managers actually use modeling software or any type of performance estimation software as part of their design criteria for the network. This means that addressing a converged network's performance requires a new set of skills to deal with the merging of the different network data types on the same physical network plant.

35. Speaking of a new set of skills, I would think that convergence would also have an impact on personnel. For example, if you're going to merge data and voice, then I would think that the data and voice people also must merge. Doesn't this imply, then, that network convergence will also require organizational changes?

Yes, it does. Organizational changes will indeed need to be made, or at least considered, to properly plan and prepare for converged network environments. As you noted in your question, in most corporate network environments, the voice, data, and video departments rarely are part of the same networking infrastructure, nor do they talk to each other very much. Even worse, the different networking departments may report to very different management structures, which can lead to political infighting as the converged networks are deployed throughout a company. The skills sets traditionally required to manage a voice or data network are substantially different enough that separate people with separate skills sets were required to manage each network type. If both network types adopt the common use of IP, the skills sets emerge rapidly into more of a data skill set than to a voice skill set. This means that the folks working on the voice side of the network need to understand IP and all the networking capabilities associated with it.

36. Isn't there some common ground between the two networking applications in their implementations?

Yes, there are some common areas across the two network types. For the most part, though, the two networks are radically different enough that neither side will fully understand some of the nuances required to make the new combined network function correctly. Thus, as part of organizational changes, networking departments must realize and embrace the concept of training networking personnel at a fairly substantial level. For example, if the voice network is converted to a VoIP network, there is a substantial amount of training that must happen over and above IP training. Training might include understanding the new PBX technology; system management training for working with operating systems such as Windows NT or Windows 2000; basic network training to understand standard network interfaces such as Ethernet NIC cards and cabling plant issues; application management training for VoIP applications running on the operating system of choice being used for the PBX; new telephone handset training because the telephone handsets will have an IP address and not just a cable interface to the network; and a variety of other miscellaneous training that is required to fully understand the new VoIP environment. The collective scope of this training can be quite difficult, and hence, management cannot expect its networking personnel to assimilate all of the related information in a short period of time. Oddly enough, it might be easier to train a system manager to understand the salient voice-related aspects of VoIP than to try and educate voice people to understand the system and network management requirements of a data network.

37. I tend to agree with you on that last point. What about layoffs, though? Doesn't combining different networks and their respective departments into a single network and single department provide management with an opportunity to reduce its workforce?

Yes it does. This is another consequence of network convergence. The management structures involved in an organization must also change to deal with the convergence of voice, data, and video networking into a single network. This usually involves eliminating management positions and the consolidation of some functions into other management positions. Viewed from the perspective of a corporation's bottom line, this is definitely a desirable effect. However, these actions cause tremendous hardship on personnel and on personnel relationships in any department affected by the changes. It also causes the remaining or newly formed management structure to understand many networking types that were previously not part of the job description.

38. Speaking of management, how will convergence impact network management?

Network management of a combined network changes dramatically as well. Most companies will use their existing data network as the base network for combining voice and video functions in a converged networking environment. Well-managed data networks generally have a variety of tools available that enable a network manager or personnel to monitor and manage data traffic. The unfortunate part is that many companies neither purchase these tools nor have the personnel who know how to use them correctly. This means that, in many cases, when the voice and video functions are combined onto the data network, the network management function becomes more complex and yet does not benefit by having additional or better tools to deal with the new application types.

39. OK. But suppose the combined network is IP-based? Doesn't this help things?

Yes. If a common protocol such as IP is used for the converged network, many of the standard data network management function tools, such as HP OpenView, may be used to monitor activities relative to the protocols used in the new network management environment. Even though the voice or video systems may not have support for the management station being used, since the application protocol is supported, they are easily monitored and managed just like a data network. In most situations, the video or voice systems supplier has applications or new tools to provide network management functions in the IP environment for the converged functions. In any regard, problems of moving from three separate networks to a single network will involve a change of skill sets for the network management teams. The only solution to this is training for all parties involved and the realization that some people who are transferred from voice or video network management into the combined network will require additional training that the data networking people already have. Obviously, if wireless network technologies such as Bluetooth or other wireless microcell technologies are used, additional and specialized training will be required to satisfy the needs to keep the network operational.

40. My organization relies on the Internet for a lot of its corporate network applications. Doing so has saved us a lot of money because we were able to eliminate all of our private networks. We also intend to begin incorporating VoIP. Will this convergence thing have any impact on how we intend to use the Internet?

It is not uncommon at all for a company to reduce its reliance on existing private networks in favor of the Internet. In most companies, there is a gross misunderstanding of what the Internet actually is, however. The Internet is a consolidation or combination of many, many networks owned by very large companies around the world. The actual Internet backbone is, in relative terms, quite small. Most companies never connect to the backbone directly and, instead, connect to large telecommunications carriers such as AT&T, MCI WorldCom, and Sprint or to a very large hosting provider such as Exodus Communications. Therefore, what a company believes to be the Internet is actually a telecommunications company's large network that is connected to other sites located on the same network or, through peer connections, to other networks. Under ideal circumstances, the source address and the destination address between cooperating systems on the Internet would be on the same vendor's network, which would provide total control over traffic path and performance by the vendor and allow a guaranteed service level agreement (SLA). Such agreements are critical in convergence networks such as a VoIP solution to ensure the performance of the network and performance of the applications using the network.

41. Don't these large telecommunications companies have the capability to support a converged data, voice, and video network?

Most larger vendors that provide network connectivity capabilities to a company understand and deal with the convergence of data, voice, and video networking as part of their everyday business. It is still up to the customer, however, to ask the right questions of the vendor to ensure that the vendor is providing the proper network bandwidth, burst management capability, service level agreement, and overall performance capabilities at the connection points to ensure success. Obviously, security is a major concern when using the Internet as a connection methodology. Consequently, Internet and VPN security issues should also be addressed when negotiating with a provider for converged services.

42. So far you've discussed convergence from an enterprise perspective. What about network convergence in the home environment?

Well, for starters, if you have an *x*DSL connection at home (see Chapter 15), then convergence is already affecting you. For example, with a DSL connection, the same phone line providing analog support also provides a digital connection for data over the same pair of copper wires. Because DSL signaling is at a different frequency on the same copper pair as the analog telephone, the two functions do not disturb each other on the same physical pair of wires. This is an example of convergence at the signaling level. Similarly, a cable modem connection represents the convergence of data and video, and, in the very near future, it will also include voice. If you now share your *x*DSL or cable modem connection with more than one computer in your home, you have yet another type of convergence: Computers that once were standalone devices are now communicating with each other. As another illustration, consider the new IEEE 1394b standard for multimedia networking, which supports network connections as high as 1.6 Gbps. IEEE 1394b-compliant networks enable PCs, storage devices, and consumer electronic devices such as camcorders and home audio equipment to be interconnected, thereby providing a single network that merges data, video, and audio.

Convergence is also making a presence in multifamily dwellings. For example, many new apartment facilities in the United States are being prewired with high-speed Internet capability as well as PBX functionality built into the apartment. By providing both voice and data capability to each apartment, apartment providers can offer superior services, charge higher rents and make more money, and also attract technically literate tenants who can afford such connectivity. The same capability has not gone unnoticed to new home builders, who are now providing Category 5/5E UTP cable to each room in a residence as a means of connecting voice and data for new owners.

The convergence of video, voice, and data at the residence is obviously a high-growth area for telecommunications providers. Further, the agreements by large network vendors such as Cisco Systems with appliance makers such as Whirlpool mean that network and traditional industrial equipment manufacturers are obviously on the path to merging networking capabilities into traditional appliances such as refrigerators, washers and dryers, and kitchen stoves. Some demonstrations already have included the ability for a refrigerator to monitor food extraction and placement by scanning bar codes as the food is put in or removed from the refrigerator. A small processor in the refrigerator keeps track of inventory and makes this available to a residential local area network server. The server may then be available to the Internet to be scanned from work or from a remote location and find the exact contents of the refrigerator or have an on-line grocery store provider automatically deliver given items as the inventory drops to an unacceptable level. While this all seems the stuff of fantasy, it is a fact that this is going on and will surely be much greater than the current prototypical offerings. The convergence of services with the networking of home appliances and other technologies is inevitable. It's not a matter of if; it's a matter of when.

43. Throughout this discussion you have mentioned VoIP several times but have yet to describe it. Could you please give me some information about VoIP?

As noted earlier, in a VoIP network, conventional telephone calls are carried via an IP-based network such as the Internet. This means that instead of establishing a dedicated, end to end circuit for a particular telephone call, VoIP uses packet-switching technology to transmit telephone conversations. The advantage is that unlike circuit-switched networks, which have the potential to waste bandwidth, VoIP is more efficient because bandwidth can be shared (see Chapter 1). VoIP also enables users to bypass the public switched telephone network (PSTN) when making long distance calls because their conversations are being transmitted across the Internet as data packets. Thus, VoIP effectively eliminates toll charges for long distance calls and hence can result in a considerable savings in long distance charges by individuals and organizations. There is a downside to using VoIP, however. Data networks were never designed to carry voice traffic, which requires a certain QoS. For example, consider frame relay (Chapter 12), which is a popular public, WAN packet-switching protocol that provides LAN-to-LAN connectivity. One of frame relay's redeeming features is that in the presence of congestion it simply drops data frames, requiring the destination node to issue a retransmission request to the sending node. This is a very effective approach for data transmissions, but it is unacceptable for voice transmissions.

44. Why is that?

It's because voice messages are converted to data frames, which are then transmitted across a network in exactly the same manner as other "regular" data frames. In the absence of any special QoS parameters, a voice-data frame is treated as just another data frame.

45. I still don't understand why this is an issue. I thought voice conversations don't require a lot of bandwidth.

You're right; they don't. Typically, a one-way VoIP transmission requires at most 64 kbps. When the voice data are compressed, the transmission data rate requirement in some cases (depending on the compression method) is less than 8 kbps. So, yes, a voice conversation does not require a lot of bandwidth. The problem, however, is trying to make sure that the bandwidth needed is available regardless of how much traffic is on the network. Thus, a VoIP network has to guarantee that the required bandwidth will be available for all VoIP sessions, including those that take place during periods of heavy network congestion. It's like stopping in a convenience store for a cup of coffee on your way to work. Just because this activity might normally require less than 5 minutes of your time, what happens if the store is crowded one morning? What can you do to ensure that your activity will still take less than 5 minutes? In other words, how can you guarantee that the time needed for your activity does not exceed that 5-minute threshold?

This example also illustrates the other part of the VoIP problem, namely, the issue of delay or latency. If a VoIP network is going to provide the same quality we have come to expect from the PSTN, then the network must also ensure that the delay in transmitting a voice packet from sender to receiver does not exceed a specific threshold, and that the variation in delay over time (called jitter) also does not exceed a certain threshold. For example, the maximum one-way delay for transmitting a voice packet should be around 150 ms. Thus, if the round-trip delay in a VoIP network is greater than 300 ms, then "normal" conversations become interrupted. A good illustration of this is when a television reporter is interviewing someone via a satellite link. There are gaps in the conversation, some parts of the conversation have to get repeated, and the people start talking over each other's words. Putting this in the context of our convenience store example, the question that needs to be answered is, "How much of a delay can you tolerate? In other words, when you arrive at the convenience store and are greeted with congestion, how long are you able to wait: 5 minutes? 10 minutes? 20 minutes? Furthermore, what impact will this delay have on you?

46. What protocols does VoIP use to help address QoS issues?

VoIP networks employ the *real-time transport protocol* (RTP) and the *real-time transport control protocol* (RTCP), both of which are IETF standards (see RFC 1889). RTP is an end to end connection-oriented protocol that supports delay-sensitive data types such as audio and video. An RTP packet consists of a header (Figure 17.2) along with the payload (i.e., the actual audio or video stream). Thus, in a VoIP network, voice messages are first encapsulated in RTP packets, which in turn are encapsulated into UDP packets (see Figures 3.13 and 3.14).

47. Hold it. Why UDP and not TCP? Isn't UDP unreliable?

Yes, the user datagram protocol (UDP, see Chapter 3) is indeed an unreliable transport protocol whereas TCP guarantees reliable delivery. Running RTP over UDP might seem a bit strange to some people at first. However, recall from Chapter 3 that TCP is a positive acknowledgment with retransmission (PAR) protocol. This means that if a transmitting node fails to receive an acknowledgment for a previously transmitted segment within a specified time period, it will retransmit the segment. Such retransmissions cause delays, which disrupt the real-time audio or video data streams.

48. I see. That makes sense now that you explained it. What does RTCP provide?

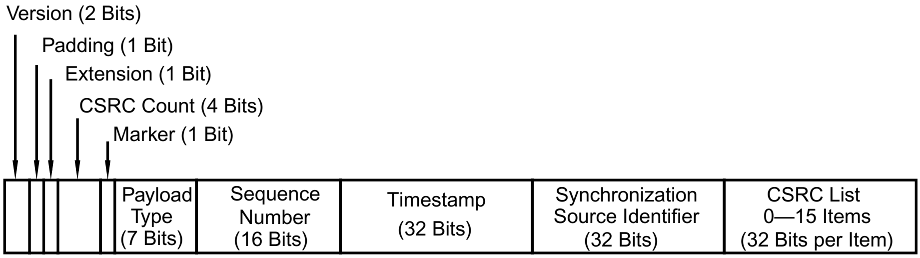
RTP's companion protocol, RTCP, provides monitoring and feedback support. Working in concert with RTP, RTCP monitors the quality of service each destination node is receiving and conveys this information to the sender. The type of feedback reported includes the packet-loss ratio (i.e., number of packets lost divided by number of packets sent) and the amount of jitter the destination node is experiencing. Although the standard does not specify how this information is to be used, the sending node (or application) can use the feedback to adjust its transmission rates or it can use the information strictly for diagnostic purposes.

49. If RTP uses UDP, then how can it provide QoS?

RTP does not directly provide any type of QoS function. Thus, it does not guarantee if packets are delivered in a timely manner (or at all), and it does not prevent packets from arriving out of sequence. Because it runs over UDP, RTP also does not assume that the underlying network is reliable and will deliver packets in sequence. These functions are provided by the lower-layer services. Bandwidth guarantees and other policy-based QoS also can be provided by the reservation resource protocol (RSVP, see RFCs 2205, 2210, and 2750).

50. Then what purpose do the sequence number and timestamp RTP header fields serve?

The sequence number and timestamp fields are used by the receiver. For example, the sequence numbers enable the receiver to reconstruct the sender's packet sequence, it can be used to determine the proper location of a packet, and it can help the receiver estimate how many packets are being lost. Together with the timing information provided by the header, the sequence numbers also enable the receiver to reconstruct the timing produced by the source so that the sound you hear, for example, is exactly as what was generated.



er ion Speci es the RTP version used; the current version is 2.

Pa ing If set, the packet contains additional padding bytes that are not part of the payload. This is needed for some encryption algorithms.

ten ion If set, then the RTP header is followed by one header extension.

ontri ution Source ount S Speci es the number of CSRC identi ers that follow the header.

ar er De ned by a speci c pro le. For example, a frame boundary may be marked in the packet stream.

Pa loa pe Speci es the format of the RTP payload and determines its interpretation by the application. For example, the payload type of an audio stream speci es the audio encoding such as pulse code modulation (PCM, see Chapter 7) being used.

Se uence um er Increments by 1 for each RTP data packet sent; may be used by the receiver to detect packet loss and to restore packet sequence (see Chapter 3).

ime tamp Re ects the time of the moment the rst byte in the RTP data packet was sampled; may be used by the receiver to remove jitter and to synchronize the receipt of the data stream.

S nc roni ation Source SS enti er Identi es the source of a stream of RTP packets. The SSRV identi er is not the IP address of the sender; it is a random number the source assigns to a new data stream. (See Appendix A.6 of RFC 1889 for an example algorithm for generating a random identi er.)

ontri uting Source S i t Speci es the contributing sources for the packet's payload. The number of identi ers is given by the CSRC count eld. A maximum of 15 sources may be identi ed.

FIGURE 17.2 Contents of a fixed real-time transport protocol (RTP) header. Source: Adapted from RFC 1889.

51. OK. I now understand some of the issues related to VoIP. What about the components of a VoIP network? What do you need to create a VoIP network?

The typical hardware components of a VoIP network include VoIP-enabled end-user devices, private branch exchanges (PBXs), and VoIP to PSTN gateways.

- A *VoIP end-user device* includes IP telephones and PC phones. An *IP telephone* is a specially designed telephone with VoIP functionality. It includes a built-in coder/decoder (codec) for digitizing and undigitizing voice messages, special software, and a network interface card (NIC, see Chapter 6) with a 10BASE-T port that enables the phone to be connected directly to a LAN. Early IP telephones required a separate power jack connection. More recent models are powered directly by the Category 5 UTP cable. IP phone manufacturers include Cisco and Nokia. A *PC phone* involves the use of a standard desktop computer running special VoIP software. PC phones can be configured in several ways. One strategy uses a speaker and microphone plugged directly into the PC. A second method uses a conventional analog phone that is plugged into a special internal or external PC adapter. Yet a third method uses a special-purpose phone that connects to the PC's serial port. Regardless of the connection method, the PC's NIC performs double duty, processing both VoIP and data packets.

- A VoIP-based PBX, commonly called a *gatekeeper*, provides functions similar to those provided by a conventional PBX, including dialtone service, internal and external call switching and routing, call setup, and special features such as call forwarding, call waiting, and conference calling. (Recall from Chapter 7 that a PBX is a telephone exchange used within an organization to provide internal telephone extensions and access to the public switched telephone network.)
- A *VoIP to PSTN gateway* converts between VoIP and PSTN phone calls. Two types of gateways are available. One type provides support for the *Signaling System 7* (SS7) protocol, the other does not. First defined by ITU-T in 1980, SS7 is an out-of-band protocol that provides signaling information for the PSTN. Thus, SS7 enables voice conversations to be transported independent of signaling information, which speeds up the time it takes to nail up and tear down a circuit. SS7 also provides additional intelligence to the PSTN, including access to toll-free numbers, calling card services, caller ID, and call forwarding. A VoIP to PSTN gateway that does not support SS7 uses in-band signaling and only supports calls from a VoIP end user-device to a traditional PSTN phone number. A VoIP to PSTN gateway that also incorporates SS7 to VoIP conversion capability enables VoIP users to access the additional services previously outlined.

To get an idea of how all of this is connected, see Figures 17.3, 17.4, and 17.5. Figure 17.3 shows a typical PSTN setup that does not involve VoIP. Figure 17.4 contains a typical LAN-based VoIP that does not involve the PSTN. In this setup, telephone calls bypass the PSTN and use the Internet to carry telephone conversations. In this scenario, as depicted, telephone calls cannot be connected between the VoIP and PSTN. In Figure 17.5, we show a full-blown VoIP network that is integrated with the PSTN. In this illustration, a VoIP to PSTN gateway is used to connect telephone calls between the VoIP network and the PSTN.

52. Since a VoIP network combines voice and data, does it use any of the multimedia protocols we discussed earlier such as MPEG or any of the H.320 family?

Yes. Most VoIP networks are H.323-based. As indicated earlier, H.323 is an ITU-T standard for packet-based multimedia communications. It is a protocol suite that comprises several protocols, including H.225 for call signaling and gatekeeper to gatekeeper communications, H.235 for security and encryption, H.323 Annex E for call connections over UDP. H.323 also supports IETF's Real-Time Transfer Protocol (RTP) for delivering packet-based audio and video streams across the Internet, as well as the Real-Time Control

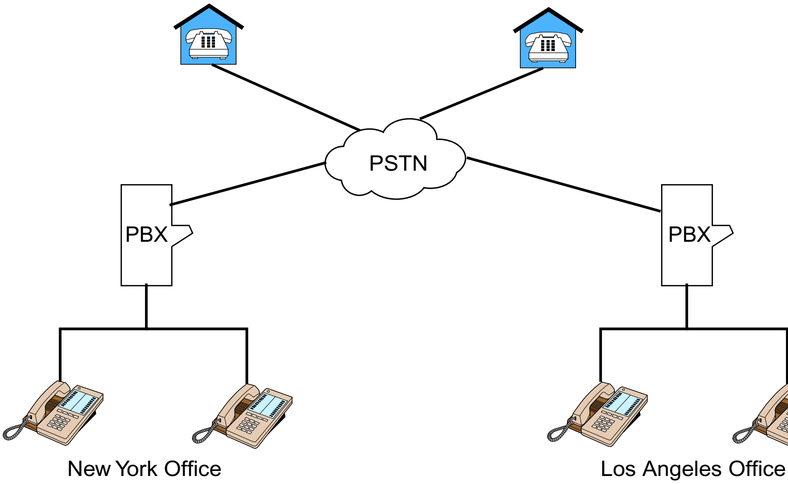


FIGURE 17.3 A conventional voice network uses the public switched telephone network (PSTN) to carry telephone conversations between calling parties. Office private branch exchanges (PBXs) connect phone calls between the PSTN and office extensions. PBXs also provide dialtone service and switch calls between extensions within an office.

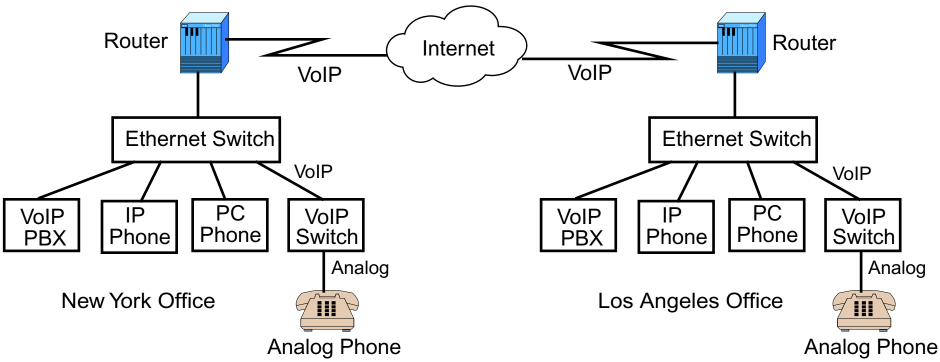


FIGURE 17.4 A LAN-based VoIP network completely bypasses the PSTN. A VoIP PBX provides similar functions as a conventional PBX but is now directly connected to the LAN via the Ethernet switch. IP phones and PC phones also have Internet connections, and analog phones are connected to a special VoIP switch, which enables them to be part of the VoIP network as well. In this scenario, calls between the PSTN and VoIP are not possible because there is no gateway that interconnects the two networks (see Figure 17.5). Source: Adapted from Carden, 2000.

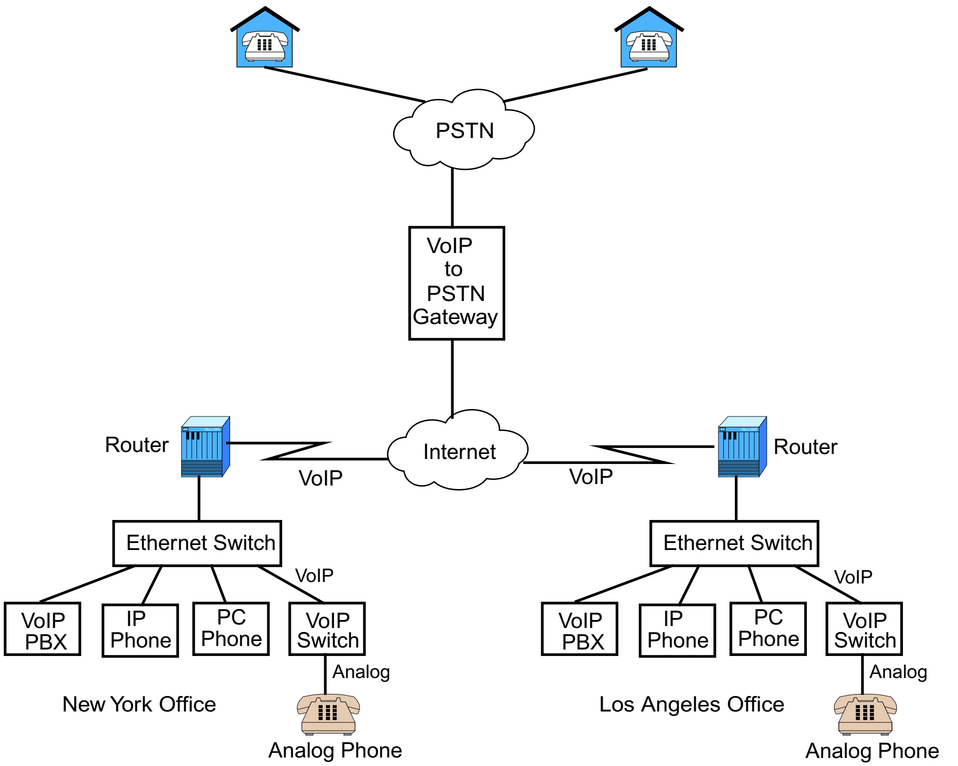


FIGURE 17.5 In a full-blown VoIP network, telephone conversations can be carried by both the PSTN and the Internet. Internet-based conversations take place via the VoIP network and connections between the PSTN and the VoIP network are handled by the VoIP to PSTN gateway.

Protocol (RTCP) for maintaining QoS information. Overall, the H.323 protocol suite enables a VoIP network to perform typical PSTN functions such as call setup/tear-down, call transfer, call forwarding, call waiting, and call holding.

53. You said "most" VoIP networks are H.323-based. Please qualify what you mean by this.

H.323 originally was developed for peer-to-peer LAN-based videoconferencing. As such, it was deficient in some areas when it came to supporting VoIP. For example, as a peer-to-peer protocol, H.323 initially did not support a network-to-network interface. Another issue was with the receipt and delivery of IP-based voice traffic. As discussed in Chapter 3, well-known IP services such as SMTP-based e-mail are assigned a specific port number on which end nodes receive and deliver their services. H.323-based voice traffic, however, does not use a well-known port number. Instead, end nodes are required to use a port number greater than 1024. H.323 is also very complex and difficult to implement because there are so many different protocols that comprise the protocol suite. Although it has gone through several revisions—the most current is version 3—to address many of its deficiencies, doing so has simply increased the protocol's complexity because many of the new changes were simply added to previous versions. As a result, these issues led to the development of a new protocol called the *Session Initiation Protocol* (SIP).

The SIP protocol suite (not to be confused with the SMDS Interface Protocol discussed in Chapter 13) is a proposed IETF standard (see RFCs 2543, 2848, and 2976) that was developed at Columbia University. SIP addresses many of H.323's deficiencies, including use of a fixed port number (the default is 5060) and complexity (SIP is a much simpler protocol to implement). SIP still does not support a network-to-network interface, though. SIP and H.323 are considered competing protocols for VoIP signaling and there is much in the trade press about SIP vs. H.323. Which one will win out is difficult to predict. In SIP's favor, it is supported by IETF and it is IP-based—two very strong reasons for wanting a SIP-based VoIP network. Unlike H.323, though, SIP is still early in its development stage, and is not as widely implemented or tested as H.323. It is interesting to note that as the two protocol suites evolve to provide improved VoIP support, H.323 and SIP are actually converging because they are incorporating key features or strengths of each other.

54. What else is there about VoIP?

Lots. We haven't even begun to break the surface. VoIP is evolving rapidly and many new protocols are being developed or old ones are being revised to make VoIP a robust technology. As with anything new in networking, patience is a virtue. We encourage you to review our comments about the standards development process in Chapter 1, as well as our comments about the components of good network design in Appendix D before you jump too deeply into implementing a VoIP network. We also suggest you review some of the resources listed in the Bibliography to become more knowledgeable about VoIP.

END OF CHAPTER COMMENTARY

The topic of network convergence represents the union of nearly every technology and application discussed in the previous chapters. To appreciate and understand what network convergence entails requires an understanding of both past and present technologies and related applications. For example, the convergence of voice and data involves understanding basic voice communications concepts that were presented in Chapters 11 and 13 as well as basic data communications concepts that were presented in Chapters 1, 2, 4, and 5. This convergence also involves packet-switching technology and is being implemented at both LAN and WAN levels. Once again, to understand the related protocols and various implementation issues—including quality of service (QoS), the differences between circuit-switching and packet-switching, Ethernet/802.3 LANs, and the Internet's ability to handle voice packets—a review of Chapters 1, 2, 3, 5, 8, 12, 13, and 15 is in order. Finally, realizing that the convergence of voice and data only begins to scratch the surface of true network convergence, we presented several examples in this chapter that should serve as the impetus for discovering other ways to merge present technology into compelling new entities. We hope that you now have a solid foundation of networking that will enable you to do just that.

This Page Intentionally Left Blank

Chapter 18

Wireless Networking

The subject of wireless communications was discussed in several earlier chapters. For example, in Chapter 4, we presented information about the two wireless transmission methods: radio frequencies and infrared. We also discussed in Chapter 4, satellite communication systems, including two high profile projects, Iridium and Globalstar, as well as the various orbits in which satellites can be placed. In Chapters 4 and 15, we discussed wireless LANs (WLANs), including HomeRF and the various IEEE 802.11 standards. Finally, in Chapter 17, we introduced Bluetooth, a short-range wireless technology that enables any Bluetooth-enabled device to communicate with each other via radio links. In this chapter, we extend our discussion of wireless concepts and present more detailed information about various wireless communication systems. If you haven't done so already, we encourage you to review the aforementioned chapters prior to reading this chapter. An outline of the major topics we discuss follows:

- Wireless Communications Overview and History (Questions 1–2)
- Wireless Data Transmission Methods (Questions 3–7)
- Cellular Telephone Networks (Questions 8–19)
- Paging Networks (Questions 20–24)
- Wireless Data Networks (Questions 25–32)
- Bluetooth Concepts vs. Wireless LANs (Questions 33–36)
- Advantages and Disadvantages of Wireless Communications (Question 37)
- The Future of Wireless Communications (Question 38)

1. What is wireless networking?

Wireless networking, or more generally, *wireless communications*, refers to the transmission of signals through a wireless medium such as air or space instead of through a physical cable. Wireless communications signals are transmitted and received via antennae (which is the plural of antenna). More specifically, a transmitting antenna radiates electromagnetic waves into the air and a receiving antenna collects these waves from the air.

2. How long has wireless communications been around?

Wireless communications dates back more than 150 years. Consider, for example, the following key wireless related events and developments:

- In 1835, Samuel Morse developed Morse code, which represented letters and numbers as dots and dashes. Within a few years, Morse also developed the electromagnetic telegraph, which enabled messages to be coded and transmitted as a series of dots and dashes.
- In 1876, Alexander Graham Bell was awarded a patent for the telephone set, which was based on the telegraph system. His basic premise was that if he could convert sound into electricity, then the electrical energy generated could be manipulated into coded messages similar to telegraph messages. A few years later, in 1880, Bell then developed an early wireless phone, called the photophone, which represented voice messages as a modulated light beam.
- In 1886, Italian inventor Guglielmo Marconi, who is considered the grandfather of wireless communications, submitted the first radio-related patent. In 1901–02, the first United States transatlantic wireless telegraph station was built by Marconi on Cape Cod, Massachusetts, and the first broadcast was transmitted from this site in 1903 between U.S. President Theodore Roosevelt and Great Britain's King Edward VII. Marconi was the recipient of the 1909 Nobel Prize in physics for his contribution to the development of wireless telegraphy.
- In 1924, Zenith sold the first portable battery-operated radio.
- In 1927, pictures of United States Secretary of Commerce, Herbert Hoover, were transmitted between Washington, DC and New York City in the world's first televised speech and first long distance television transmission.
- In 1928, John Logie Baird transmitted images of London to New York via shortwave radio.
- In 1940, AT&T introduced a cordless telephone prototype.
- In 1941, actress Hedy Lamarr and composer George Antheil received a patent for a *secret communications system*, which enabled a transmission between sending and receiving devices to be scattered across different frequencies in the radio spectrum. This frequency "scattering" made it more difficult for signals to be detected, jammed, or intercepted. The secret communications system patent, which expired in 1965, served as the basis on which the concept of spread spectrum (see Chapter 4) was developed.
- In 1954, Genie introduced the remote-control garage door opener, and in 1956, Zenith introduced the wireless television remote control device.
- In 1967, AT&T introduced the first commercial cordless telephone and began field trials of its use.
- In 1969, the first transmission between the earth and moon was broadcast.
- In 1983, AT&T and Motorola introduced cellular telephone service in Chicago.

Today, recent developments in wireless protocols, standards, and convergence technologies (see Chapter 17) have led to a plethora of very capable and affordable wireless networking products, including Web-enabled phones that combine cellular telephone and Internet technology, wireless pagers, and personal digital assistants (PDAs). These wireless Internet devices have brought the concept of wireless communications and networking to the forefront, providing users with continued and untethered access to the Web, their e-mail, and instant messaging.

3. From my perspective, wireless communications is like magic. After reading various chapters in this book, though, I now realize that there is probably a tremendous amount of behind-the-scenes technology that creates this magic.

Yes, you're right. There is a great deal of wireless networking technology that is transparent to the average user of a wireless communications system. To better understand this technology, let's begin our discussion by reviewing some of the fundamental concepts of wireless transmissions. Recall from Chapter 4 that there are two general types of wireless transmission methods: *radio transmission* and *infrared transmission*. Radio transmissions use radio frequencies (RF) to transmit information, with different portions of the radio spectrum (Figure 4.18) being assigned to different services. Infrared (IR) transmissions use electromagnetic radiation of wavelengths between radio waves and visible light, operating between 100 GHz and 100 THz (Terahertz).

4. Let me stop you here for a moment. Every time you use the term "radio," I always think of "radio-radio." You know, like FM radio stations. Is this the same thing as the radio you are talking about?

Yes, they are related and we can understand why you might be confused. The word *radio* is a general wireless communications term that designates a specific range of frequencies within the electromagnetic spectrum. This frequency range is from 3 kHz to 300 GHz. Thus, as shown in Figure 4.18, the radio frequency (RF) spectrum ranges from the extremely/very low frequency (ELF/VLF) band to just before the infrared band. Broadcast radio, which is what you are referring to, operates within this range. For example, FM radio operates in the 88–108 MHz range. It should be noted though, that the term *broadcast radio* is informally used to specify the frequency range from 30 MHz to 1 GHz. Thus, from Figure 4.18, broadcast radio also includes VHF and UHF transmissions.

5. According to Figure 4.18, microwave is also part of the RF spectrum. Does this mean that microwave is considered "radio?"

Yes. *Microwave radio* is a specific type of radio transmission that operates within the 1–100 GHz frequency range. Recall from Chapter 4 that a wavelength is a measure of the length of an electromagnetic wave. It is the distance an electrical or light signal travels in one complete cycle. The length of microwave is in the millimeter range (and thus the name *micro*wave). As a result, a microwave transmission consists of very small wavelengths that operate at ultra high, super high, and extremely high frequencies (UHF, SHF, EHF). Because the term GHz means billions of cycles per second, a microwave transmission generates billions of very small waves each second. As we discussed in Chapter 4, because of its high frequency range, microwave radio is very susceptible to environmental interferences—such as rain and fog—and attenuation. Microwave transmissions also require a line-of-sight configuration; that is, the antennae used for transmission and reception must be within each other's line of sight. Furthermore, microwave antennae are parabolic in shape so they can collect as much of an incoming signal as possible. This is because microwave transmissions operate at frequencies that are closer to the light spectrum than they are to the radio spectrum. As a result, microwave tends to be more characteristic of light, which means that microwave signals are more likely to be refracted or reflected.

6. Where does satellite communications fit into the spectrum?

Satellite communication systems are microwave-based and sometimes called *satellite microwave* or *satellite radio*. Satellite transmissions generally operate between 1 GHz and 10 GHz. However, because this frequency range has become saturated, two additional bands have been developed for use: the 12–14 GHz band and the 19–29 GHz band. Although these higher frequencies offer higher bandwidth, they also are more susceptible to attenuation and interference. These issues are being addressed, however, via advances in technology.

As we discussed in Chapter 4, a satellite communications system involves an orbiting satellite that broadcasts signals to one or more ground stations. The satellite receives a signal on one frequency band (this is called the uplink), regenerates the signal (i.e., it amplifies or repeats it), and then broadcasts it to the ground stations on another frequency band (this is called the downlink). Satellites can be placed in different orbits, including low earth orbit (LEO), medium earth orbit (MEO), and geosynchronous earth orbit (GEO) (see Chapter 4 for additional details). The number of satellites needed for global coverage (called a constellation) is a function of the altitude at which the satellites orbit; the higher the orbit, the less number of satellites required. Satellite systems traditionally have been used for television and radio broadcasts, long-distance telephone transmissions, and videoconferencing. Other satellite communications applications include global positioning systems (GPS), mobile voice and data systems such as those provided by Iridium and Globalstar (Chapter 4), direct broadcast satellite (DBS) television in which satellite television signals are transmitted directly to home TV sets, and satellite-based Internet access.

7. In what way is infrared different from radio?

As noted earlier, infrared transmissions use the infrared light band of the electromagnetic spectrum. Frequencies range in the terahertz (THz) range, which is trillions of cycles per second. In an infrared transmission, a focused light beam is sent from a transmitter to a receiver. Like microwave radio, infrared is a line-of-sight medium. Unlike microwave, though, infrared cannot penetrate walls and distances are very limited (usually up to two miles). Traditional applications include remote control devices such as those used for television sets, VCRs, and projection equipment. Infrared is also used in LAN environments such as campus networks for building to building connectivity, and can be found on various computing devices such as laptop or desktop computers and PDAs.

8. So what is happening in today's wireless world?

In today's wireless marketplace, there is a host of wireless voice/data services and applications available, including cellular telephony, paging, cordless telephony, wireless

LANs (WLANs), personal communication services (PCS), mobile voice and data systems, and PDAs. For some people, these services and applications have become so integrated into their lives that the term wireless to them means much more than simply communication without wires; it represents a specific type of lifestyle.

9. How are all of these wireless services and applications provided? For example, is there one big wireless network that supports them all?

If there were, then wireless communications would truly be magical. The reality is that each different type of communications style involves a different type of wireless network. For example, cell phones operate over a cellular telephone network, paging services are provided via a separate paging network that is different from the cellular phone network, mobile wireless voice and data services such as those provided by Iridium and Globalstar (see Chapter 4) have their own independent networks, and WLANs represent another type of wireless network. In addition to the different networks, there are also different wireless network technologies associated with each network. If that isn't enough, now consider the Bluetooth technology (Chapter 17), which enables cell phones, pagers, personal fax equipment, laptops and other computing devices to interoperate with each other over a very short distance.

10. You must be kidding. From my perspective, it doesn't look that complex.

That's because you're still looking at wireless communications as magic.

11. Is it possible for you to give me the *Readers' Digest* version of some of the technologies associated with some of these different networks?

Sure. Let's start with the *cellular communications network*. The concept of cellular communications began in the 1946–47 time frame when mobile phone service was introduced in Chicago. This system, which was called *Mobile Telephone Service* (MTS), supported vehicle to vehicle communications within the Chicago area via a radio-to-telephone interface. Eventually, the service was deployed to major metropolitan areas within the United States. This service consisted of: a large coverage area (usually a 25-mile radius from a city's downtown area); a strong, centrally located antenna that transmitted 250 watts of power; a small number of channels (typically 12); high-powered antennae mounted on high-rise buildings within a specific area, which picked up signals from the mobile transmitting devices; and *frequency reuse*—a concept in which the channels allocated for service in one area are reused in different areas. Frequency reuse is common in the radio and television broadcasting industries. For example, there are more than 300 FM radio stations within the United States that operate at 102.3 MHz, including WSLE in Cairo, GA, WISY in Canandaigua, NY, WMMJ in Bethesda, MD, KPEZ in Austin, TX, and KJSN in Modesto, CA. One of the problems with frequency reuse, though, is that if careful consideration is not given to factors such as signal strength, geographical area, and antenna design, then the signals operating at the same frequency but in different areas can interfere with each other. This is exactly what occurred with the early mobile phone service. A 250-watt signal is a very strong signal—so strong that it exceeded its designated coverage area and interfered with signals in nearby areas. In fact, in the early mobile phone systems, frequency reuse could not be implemented within a 100-mile radius. (The signals didn't know they were supposed to stop at the 25-mile radius boundary. In some ways, signals are like weeds in that they will permeate nearly everything.) This limited the number of possible simultaneous conversations.

The frequency reuse problem was resolved by first partitioning a region into a series of overlapping subregions called *cells*, and then installing at each *cell site* a base station and corresponding low-powered (about 10 watts) antennae that cover the prescribed area. This network of base stations and antennae became known as a *cellular network*. Cells are typically arranged in a honeycomb fashion and can be configured in different frequency reuse patterns. This prevents adjacent cells from using the same frequency. For example, in a 7-cell reuse pattern (Figure 18.1), seven adjacent cells cannot use the same frequency. Each cell's base station contains a transmitter/receiver (i.e., transceiver) that operates at either 800 MHz or 1900 MHz, and the antennae within a cell are either omnidirectional or vector-based. An *omnidirectional antenna* provides 360º of coverage, which means that the signal is spreading in all directions during transmission. For example, when you drop a rock into the water of a smooth pond, waves ripple out in all directions. This is what a transmission is like from an omnidirectional radio frequency burst. Omnidirectional antennae are generally located at the center of a cell so that the distance radio signals have to travel when communicating with devices entering a cell is minimized. This is important because when you lower the power output of an antenna, you also reduce the signal strength and the distance the signal can be transmitted. Placing antennae in the center of a cell is analogous to playing tennis. After you hit a return shot, you always want to get back to the center of the court so that the distance you have to travel to get to your opponent's return is minimized. A *vectored antenna* subdivides the coverage area of the cell. A common subdivision is 120º, which means that a cell is partitioned into three vectors of 120º each. One of the advantages to the vectored approach is that each vector can independently manage various frequency-related issues such as channel allocation and signal strength. For example, in a 4-cell reuse pattern that employs omnidirectional antennae, each base station can support 102 simultaneous conversations. However, in a 7-cell reuse pattern that employs 120º-vectored antennae, each base station can support 22 independent sets of 58 simultaneous conversations, thereby increasing the relative capacity of a cell. Although it might appear that vectored antennae are located in the center of a cell, they are not; they are actually at the edge of each cell created by the vectoring application (see Figure 18.2). The coverage area of a cell site can range from 0.6 mile to 30 miles in radius depending on such factors as topography, population, amount of traffic, and network protocols. For example, a cell site that includes tunnels or subways will have a small coverage radius, whereas a very open, flat area with no obstructions will have a larger coverage radius. In some regions, a cell-hierarchy is used that consists of *macro cells* (large coverage area), *micro cells* (smaller coverage area), and *pico cells* (very small coverage radius). For the most part, though, a cell site's range is usually limited to a 3- to 5-mile radius. Note that one of the effects of partitioning a cell into smaller cells is increased system capacity.

12. Wow! Could you summarize this into a more concrete and coherent form?

Okay, we'll try. Let's first list the major components of a cellular network. These include: mobile or handheld devices such as cell phones; base stations, which use low-pow-

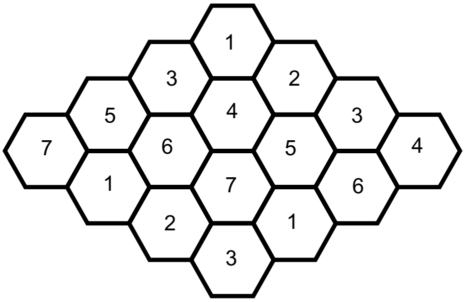


FIGURE 18.1 Cellular networks partition a region into a series of overlapping subregions called cells, which are arranged in a honeycomb fashion. Because frequencies are limited, a frequency reuse pattern is applied to prevent the same frequency from being used more than once by adjacent cells. Thus, in a 7-cell reuse pattern, seven adjacent cells are not permitted to use the same frequency.

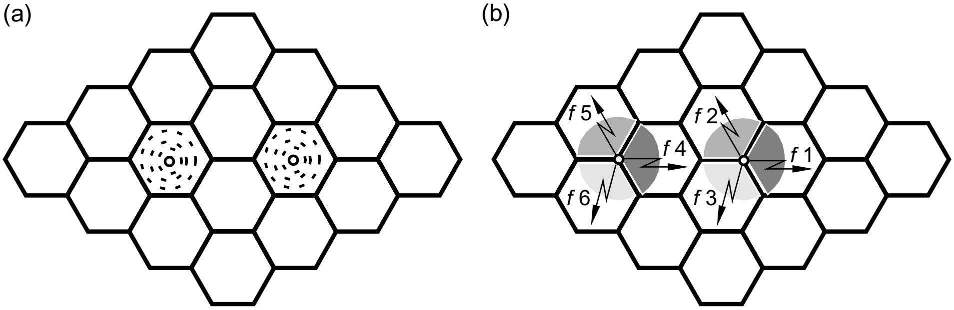


FIGURE 18.2 Omnidirectional antennae are placed in the center of a cell as shown in (a). Vector-based antennae, however, are placed at the edge of several cells as shown in (b), not at the center. Each sectored cell region uses different frequencies than the other sectors and cells.

ered radio transceivers, and corresponding antennae; a mobile telecommunications switching office (MTSO), which is sometimes called the mobile switching center or MSC; and the public switched telephone network (PSTN). To establish a cellular network within a geographical region, you first apportion the region into a series of small, overlapping parcels of land called cells. Thus, a cell is nothing more than a geographical area that is defined by the boundaries you established for it. Within each cell, you determine the best location to place a base station and antennae. This location is called the cell site. When a mobile device enters a cell (i.e., it enters a specific territory or region), it communicates with the base station via a dedicated channel. The channel assigned to support this conversation consists of paired frequencies—the base station uses one frequency to communicate with the mobile device and the mobile device uses a second frequency to communicate with the base station. In a very simple scenario (Figure 18.3), a mobile device initiates a call within a cell. This call (i.e., the signals associated with the call) is picked up by the cell site's

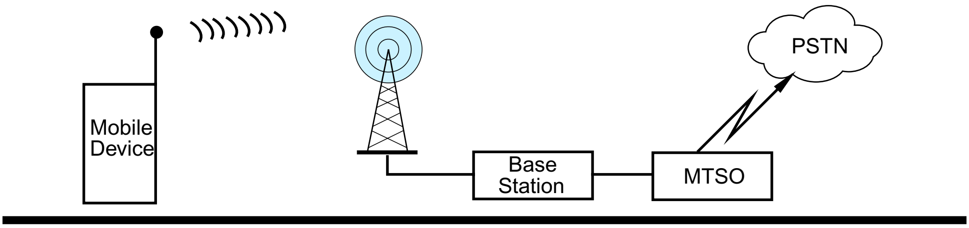


FIGURE 18.3 In a simple mobile transmission, a mobile device such as a cell phone transmits a signal to a cell site's antenna, which passes the signal to the base station's transceiver. The signal is then routed to a mobile telecommunications switching office (MTSO) where it is processed and routed to the public switched telephone network (PSTN).

antennae, passed to the base station's transceiver, and is then routed to the MTSO. The call is then processed and routed to the PSTN.

13. OK. This helps but it also raises two more questions. First, based on the frequency reuse concept, adjacent cells cannot use the same frequencies. So what happens when you move from one cell to another?

If a call is in progress and the mobile device is moving away from the cell site within the cell, its signal will become weaker. When this happens, a *handoff process* occurs where the call's signal is transferred to the cell site of the cell the mobile device is approaching (Figure 18.4). This handoff process is controlled by the MTSO. Upon detecting a weak signal from the mobile device as it leaves a cell, the MTSO transmits a broadcast message to the cell sites located within the area. This broadcast message asks the sites to report the strength at which they are receiving the mobile device's signal. After it receives the responses from these sites, the MTSO identifies the cell site that is receiving the strongest signal and instructs this site to setup a communications path with the cell site that is currently responsible for the call. A new frequency is also allocated for the new cell site. When this path is established, the MTSO directs the mobile device (i.e., the cell phone) to begin communicating on the new frequency that has been assigned to the new cell site. With the Signaling System 7 (SS7) protocol (see Chapter 17), the cellular call handoff process is done automatically within 100 ms.

14. My second question is do all cellular networks use the same transmission methods?

No. That would make it much too easy and make too much sense. There are many different and incompatible cell phone to base station transmission technologies in use today. Commonly referred to as transport mechanisms or access technologies, the three primary ones are *frequency division multiple access* (FDMA), *time division multiple access* (TDMA), and *code division multiple access* (CDMA). A brief description of each follows:

Frequency Division Multiple Access (FDMA) FDMA is wireless communication's counterpart to the wired world's frequency division multiplexing (FDM, see Chapter 4). The assigned frequency range within a cell is partitioned into narrower, multiple frequency bands, each of which constitutes a separate communications channel. Thus, telephone calls

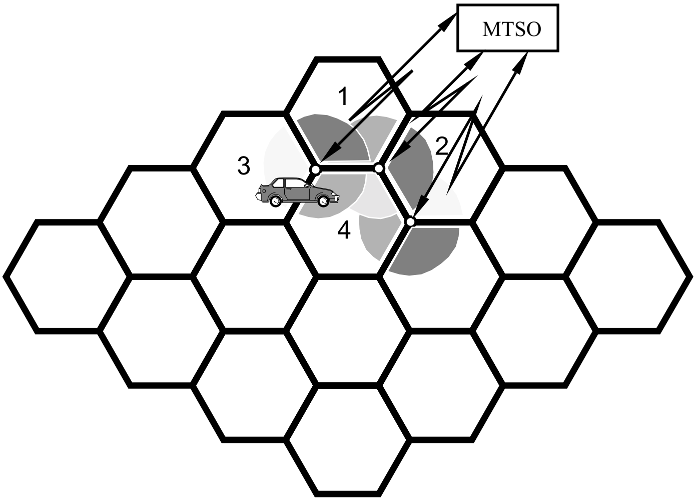


FIGURE 18.4 When a mobile unit roams from one cell to another, a handoff process is initiated. The MTSO first notices that the unit's signal is becoming weaker. The MTSO then sends out a broadcast requesting all cell sites in the area to determine the unit's signal strength. Once the MTSO determines the cell site that is receiving the strongest signal, it directs this cite to establish a communications link with the current cell site. When the link is established, the MTSO directs the mobile unit to go to the frequency of the new cell cite. Source: Adapted from Bates and Gregory, 1998.

are separated by frequencies similar to the way radio stations are separated by frequencies. That is, one transmission (a telephone call) occurs on one frequency, a second transmission takes place on a second frequency, and so forth. FDMA is used in analog cellular networks such as the *Advanced Mobile Phone System* (AMPS), which is discussed later.

Time Division Multiple Access (TDMA) TDMA is a digital technique (i.e., it first digitizes conversations) and employs exactly the same access method as the wired world's time division multiplexing (TDM, see Chapter 4). More specifically, TDMA partitions a cellular channel into three time slots, combines the corresponding digitized conversations into a single digital stream and then transports this stream on a single channel using the same frequency. Thus, in a TDMA-based cellular network, conversations take turns transmitting over a single channel, with each time slot being permanently assigned to the same channel. In contrast to FDMA-based networks, which splits a frequency into three subfrequencies to carry three conversations, TDMA-based networks carries these same three conversations using three time slots on the same channel, giving it triple the capacity of FDMA-based networks. TDMA was the first digital standard to be developed. It is used both by Global System for Mobile (GSM) Communications and Interim Standard 136 (IS-136), both of which are discussed later in the chapter. Just so you don't get confused, you should also be aware that IS-136 is colloquially called "TDMA" in North America. So TDMA is both a technology and a specific standard using that technology.

Code Division Multiple Access (CDMA) CDMA is a digital wireless communications access technology that uses spread spectrum (see Chapter 4) to increase system capacity and voice quality. In spread spectrum, the bandwidth of a signal is spread over a range of different radio frequencies. This results in a frequency range that is much wider than what was used initially. By incorporating spread spectrum into a wireless communications access strategy, more simultaneous conversations can be supported using the resulting "wider band" frequency than can be supported using either FDMA or TDMA techniques. Thus, CDMA's spread spectrum approach provides much greater coverage, which reduces the number of cell sites that are needed within a cellular network. Unlike FDMA, which separates a channel into multiple frequencies, or TDMA, which partitions a channel by time, CDMA separates cellular conversations by code. Operating in a manner similar to a UDP datagram service (see Chapter 3), every bit of every conversation gets assigned a specific code prior to transmission. Upon receiving these coded packets, the receiver reassembles the conversation from the coded bits. CDMA's digitally encoded spread spectrum transmissions also provide a much higher degree of security. Qualcomm Incorporated, which is headquartered in San Diego, CA, innovated CDMA technology and currently licenses its CDMA patent portfolio to leading telecommunications equipment manufacturers throughout the world. The term CDMA is also used to refer to Interim Standard 95 (IS-95), which is the most common CDMA-based standard in operation today.

15. Wait a minute. Qualcomm? You mean the Eudora e-mail company?

Yes, Qualcomm the Eudora company is also Qualcomm the CDMA company. Qualcomm acquired Eudora from the University of Illinois-Urbana, in 1992, and eventually commercialized the product. (Eudora was originally written by Steve Dorner at UI-C.) However, since its inception, Qualcomm's primary focus has always been (and continues to be) wireless communications. Today, CDMA technology is regarded by many as the world's most advanced digital wireless technology and one of its most recent products, cdma2000 1x, is the basis for the first commercial third-generation (3G) standard. In fact, because the Universal Mobile Telephone Standard (UMTS, see below), is also based on CDMA, it is probable that all third-generation cellular networks will adopt the technology.

16. What do you mean by "third-generation?"

As is the case with most technologies, the growth of cellular communications has been part of an ongoing evolutionary process. Throughout a technology's evolution, marketing people, who try to be creative but usually end up subscribing to platitudes, often like to distinguish products and services from different "eras" by using the term "generation." Please bear in mind, though, that this does not take effect until at least the second stage of a technology's evolution. Moreover, if a technology is able to evolve further yet (e.g., to a third or fourth stage), then products and services from previous generations are referred to as "legacy." In the context of the present discussion, cellular networks are currently in their second generation but are rapidly approaching their third generation, or 3G, as the trade people like to say. Using this naming idiom, first- and second-generation cellular networks are fast becoming legacy systems.

17. What are the three generations of cellular networks?

First-generation, or 1G, cellular networks, are primarily analog-based and characterized by the concept of mobility; that is, they were designed for mobile telephone communications. The Mobile Telephone Service (MTS), which we mentioned earlier, and its successor, the *Improved Mobile Telephone Service* (IMTS), which was introduced in 1964, were the first two mobile telephone networks in the United States. These two systems, however, did not implement a frequency reuse strategy and hence are not considered "true" cellular networks.

The first true 1G cellular network is the *Advanced Mobile Phone System* (AMPS), which was developed by AT&T and Motorola and deployed in 1983–84 time frame. AMPs operates in the 800 MHz band and supports 832 channels. The mobile device to base station channel (called the *reverse path*) operates between 824–849 MHz, and the base station to mobile device channel (called the *forward path*) operates between 870–890 MHz. These communications channels are 30 kHz wide and use FDMA as their access technology. AMPS is very popular in the United States, South America, China, and Australia. Analog-based derivatives of AMPS include the defunct *Narrowband AMPS* (N-AMPS), which increased system capacity to 2412 channels by using 10kHz-wide channels instead of 30 kHz-wide channels, and the *Total Access Communications System* (TACS), which was deployed in the United Kingdom in 1985, as well as in Japan where it is called JTAC.

Second-generation, or 2G, as well as 2.5G cellular networks are digital-based. They are characterized by increased system capacity and improved quality of service over 1G systems, and 2.5G systems support voice and data communications. There are many different 2/2.5G cellular networks and all employ incompatible standards (see http://www.cellular.co.za/celltech.htm). Examples include a digital version of AMPS called D-AMPS, IS-95, IS-136, cellular digital packet data (CDPD), and the Global System for Mobile (GSM) Communications. (There are many more.)

- *Digital Advanced Mobile Phone System* (D-AMPS) is the digital version of AMPS. It uses the same 30 kHz channels as AMPS but employs TDMA for its access technology instead of FDMA. As noted earlier, this effectively triples system capacity because each channel can now support three calls instead of one. An AMPS/D-AMPS cellular network supports either analog or digital cellular phones; the digital phones operate in the 1900 MHz band.
- *IS-95* refers to any CDMA-based cellular network that operates at 800 MHz. Thus, an IS-95 network operates at AMPS frequencies but employs spread-spectrum technology and uses a special coding scheme for its transport mechanism. IS-95A provides 14.4 kbps data transfer rates and IS-95B, which is referred to as a 2.5 generation technology, increases data transfer rates to 64 kbps. (*Note:* IS-95 is sometimes called *cdmaOne*.)
- *IS-136* refers to any TDMA-based cellular network that operates at either 800 MHz or 1900 MHz. Two examples include D-AMPS and GSM (see below).
- *Cellular Digital Packet Data* (CDPD) is a data communications strategy that runs over an AMPS network. Instead of deploying dedicated wireless data-only networks such as the Advanced Radio Data Information Services (Ardis) and the

BellSouth Wireless Data (formerly known as Ram), CDPD provides a mechanism for transmitting data over existing voice-based cellular networks.

- *Global System for Mobile* (GSM) *Communications* is a de facto cellular network standard that was developed to provide cellular compatibility throughout Europe. It has since become the international standard of choice and is now deployed in over 100 countries, including parts of the United States. Versions of GSM are TDMA-based. GSM operates in the 800–900 MHz range as well as in the 1800–1900 MHz range. It is also compatible with ISDN (see Chapter 11). GSM serves as the basis for other cellular network standards, including the *Personal Communications Services* (PCS) network, which is deployed by Sprint in the United States. PCS operates in the 1900 MHz band. Although GSM is the heart of PCS, and is even sometimes called GSM, the two cellular networks are incompatible.
- *General Packet Radio Service* (GPRS) is a nonvoice 2.5G packet-switching technology upgrade that makes Internet applications accessible via GSM or IS-136 cellular networks. Peak data transfer rates are 171 kbps under ideal circumstances, which means that the transmission uses no error-correction and is able to reserve all eight GSM time slots. Actual data rates are more in the neighborhood of 30 kbps. Unlike current cellular services such as short message services (SMS), which use cell phone to cell phone communications, users with a GPRS-enabled phone are able to access any Internet application or site that has an IP address. This includes Web pages, file transfers, e-mail, remote LANs, and multimedia files. GPRS is commercially available on most GSM networks. However, actual bandwidth and latency are still much poorer than initially projected. For more information about GPRS, see http://www.gsmworld.com.

Third-generation, or 3G, cellular networks are digital-based and characterized by higher capacity and higher-speed data transfer rates compared to 2G or 2.5G systems. They also support global roaming, which implies that you can use your cell phone anywhere in the world, and "always-on" coverage. Most current and future 3G technology standards are CDMA-based, which implies that IS-95-based 2G and 2.5G systems are in the best position to be upgraded to 3G technology. Examples of 3G CDMA-based standards include cdma2000 1x, cdma2000 1xEV, and the Universal Mobile Telephone Standard (UMTS).

- *cdma2000 1x* is the first commercial 3G standard. It supports data transfer rates from 153 kbps to 307 kbps, doubles voice capacity and standby times compared to IS-95, and is backward compatible with IS-95A/B systems. The world's first commercial deployment of cdma2000 was in 2000 by Korea's SK Telecom.
- *cdma2000 1xEV* is an enhancement of cdma2000—the "EV" is for evolution. It supports up to 2.4-Mbps data transfer rates operating on a standard 1.25 MHz channel. The specification was developed by the Third Generation Partnership Project 2 (3GPP2), which comprises telecommunications standards bodies from China, Japan, Korea, and North America. The specification also provides a data-only version (cdma2000 1xEV-DO) optimized for packet data. cdma2000 1xEV was deployed in commercial networks in South Korea and Japan in 2001.

- *Universal Mobile Telephone Standard* (UMTS) is currently being developed with an expected deployment beginning in Japan in 2002 and in Europe 2004. It is based on CDMA access technology, supports theoretical data transfer rates of 2 Mbps, and operates at 2 GHz. UMTS specifies two air interfaces: time-division CDMA (TD-CDMA) and Wideband CDMA (W-CDMA). The difference between the two specifications lies in the way they perform duplexing: W-CDMA uses frequency-division duplexing (FDD), which means that the uplink and downlink use two different frequencies; TD-CDMA uses time-division duplexing (TDD); thus, the up and down links use different time slots. W-CDMA, which is sometimes referred to as Broadband CDMA (B-CDMA), provides a 2-Mbps peak data rate and operates on a 5-MHz channel. W-CDMA lends itself better to symmetrical traffic such as voice transmissions, whereas TD-CDMA is more efficient for asymmetrical traffic such as bulk downloads. As of this writing, successful W-CDMA trials have been carried out in Europe and Japan. Additional information about UMTS is available at http://www.umts-forum.org.

18. I didn't realize there are so many different cellular network standards.

Right. Not many people do, and, as we mentioned several times before, all of them are incompatible because they all rely on different standards and access technologies. As a result, there are no cell phones that support all the different access technologies or cellular standards. Some use CDMA, some use TDMA, and some use GSM depending on where you go in the world. Worse yet, some countries use derivatives of one or more of the traditional signaling methods, which makes cellular access in that country unique to that country. For instance, many providers in the United States deploy CDMA-based networks. Japan on the other hand, has its own standard called *Personal Digital Cellular* (PDC), which is TDMA-based and hence incompatible with the United States's version. Therefore, a cell phone designed using United States specifications will not work in Japan and vice versa. In Europe, GSM is the most popular cellular transmission standard and is incompatible with the standards used in Japan and the United States. Some Asian countries' networks are TDMA-based, which again makes them incompatible with cellular networks in North America and Europe. Some cellular phone manufacturers such as Motorola have solved some of the problems by designing and building telephones that support multiple wireless network transmission types. For example, the Motorola i2000 supports multiple cellular protocols and works in nearly every country except Japan. So, although, we are not there yet, we are approaching the period where we will one day have true global roaming and interoperability using the same phone.

19. What's the current status in the United States? For example, what standard do the big carriers use?

As of this writing, the two most widespread cellular standards in the United States are GSM and CDMA. Cingular's and Voicestream's networks are GSM-based, and Verizon and Sprint PCS use CDMA. AT&T Wireless is currently building a GSM-based network. Although CDMA is considered the more technically advanced standard, it has the same primary deficiency as GSM, namely, both standards are circuit-switched. This makes it cost-prohibitive for subscribers to maintain continuous Internet connections. Japan's PDC, however, uses packet-switching technology, which enables subscribers to maintain their Internet connections without using any bandwidth; subscribers only get charged for voice calls. U.S. cellular network providers will not be able to provide their subscribers a completely packet-switched solution until they upgrade their networks to 3G standards.

20. The cell phone I have also doubles as a pager. Is this paging service provided via a separate network that is different from the cellular phone network?

In the strictest sense, there are indeed two separate networks at work: the cellular phone network and a separate, standalone paging network. Some cellular providers, however, also operate a distinct paging network or have a partnership with a paging service. This enables the cell phone carrier to send paging messages over a paging network and have these messages received by a specific telephone number. This is not any different from how a regular pager operates. In this scenario, the circuitry for receiving pages from a paging network is included in the cellular phone. Some vendors have also created gateways that allow short e-mail messages to be sent to a pager circuit in a cell phone from a desktop Internet e-mail client. The e-mail is typically sent to the telephone number of the cell phone, which is hosted by an e-mail server operated by the carrier. The carrier then forwards the message to the paging network and it is sent to the cell phone's paging circuitry. Therefore, a cell phone that supports voice and Internet paging capabilities is actually connected to two distinct wireless networks at the same time.

21. What do you mean by "in the strictest sense?"

It's complicated. In using this phrase, we are referring to the conventional definition of a page and the manner in which paging services historically were provided. The concept of paging was first conceived in 1939, and the first commercial paging systems were deployed in 1950. These early paging networks were one-way systems that used dedicated receiving devices called pagers. Pagers began as simple tone-only devices that render a distinct sound when they receive the transmitted signal. Pagers eventually evolved to numeric devices that display numerical messages (e.g., a telephone number) to alphanumeric devices that display text-based information consisting of letters and numbers. When paging networks were initially deployed, the FCC allocated the 35-MHz frequency band for paging services. As the paging concept evolved, additional frequencies were allocated by the FCC and licensed to paging providers. These included the 45-MHz, 150-MHz, and 900-MHz bands. Most conventional, standalone pagers today listen to specific FM subcarrier frequencies in the 150-MHz band and paging companies contract with local FM radio stations to use their transmission system to send paging messages throughout the network.

The standalone, locally based paging networks eventually expanded into national- and globally based systems. Today, paging networks comprise centrally located antennae interconnected via microwave, leased lines, or packet-switching networks. National providers, which are licensed to operate in the 900-MHz band, also use satellite communication systems that interconnect land-based systems. Nationwide paging systems also have cooperative arrangements with other paging or radio networks to send a message throughout the entire network for people who are mobile and require getting their pages in different cities or locations. Global paging services use satellite ground stations that are located in most airports around the world. The ground stations communicate with local radio stations to forward paging messages. Pages from the master ground station are uplinked to a satellite, which then downlinks the paging messages to the various ground stations for retransmission through a city network.

In addition to expanding their coverage areas, one-way paging systems also evolved into two-way systems. This two-way paging concept was introduced in the United States in the mid-1990s. In a two-way system, pagers are no longer simple receivers, but instead are transceivers. They contain a small transmitter that can communicate either through a radio network or a cell paging network to send small messages back to a master ground station to respond to pages. Two-way paging can be provided partially or fully. Partial two-way paging enables the pager to acknowledge receipt of the page; full two-way paging enables the recipient to use his or her pager to transmit a return message. Most two-way systems are proprietary in nature.

Today, the paging concept has been incorporated into other devices such as cell phones or personal digital assistants (PDAs), and several cellular network providers use the PCS frequency band (1900-MHz range) to provide paging services. This is where the "in the strictest sense" comes into play. Some PCS carriers provide *short message services* (SMS), which they market as a paging service. The interesting part of this is that these cellular providers are not FCC-licensed paging service providers. In other words, when viewed from technical and regulatory perspectives, PCS-based "paging" through cell phones, PDAs, or other devices is not conventional paging. So, in the strictest sense, SMS is not a paging service, SMS messages are not really pages, and PCS-based paging devices are not really pagers. Most consumers, however, don't really understand or care about these differences, which makes it easier for marketing people to market PCS-based SMS services as a paging service. So, if we relax the definitions of a paging service, paging message, and paging device, then in the not so strictest sense, SMS is a paging service, SMS messages are pages, and PCS-based devices such as cell phones and PDAs double as pagers. Furthermore, because SMS is provided via the GSM cellular standard, then the "page" (i.e., SMS message) you receive is not being provided by a separate paging network. Thus, in the not so strictest sense, given today's technology, cellular phone networks that feature SMS and paging networks are not viewed as distinct systems.

22. Exactly what is SMS if it is not a paging service?

SMS is a feature of the de facto GSM cellular standard (discussed earlier) and is based on the store-and-forward transmission concept (see Chapter 2). Messages up to 160 characters long are sent to an SMS center where the message is stored, error-checked, and then transmitted to the recipient. The SMS center also transmits a delivery confirmation message to the sender. SMS messages can include any alphanumeric character as well as nontext data such as binary formatted messages, and can be transmitted and received simultaneously with a GSM-based cell call. Pragmatically, SMS messages really are like alphanumeric pages. They just don't fit the conventional idiom of a paging system. SMS messages can provide a variety of information, including weather reports, stock quotes, and sports scores, as well as Web-based information.

Today, there are a host of wireless Web devices, including pagers, PDAs, and Web-enabled cell phones. The de facto standard that enables these devices to provide Internet communications as well as advanced telephony services is the *Wireless Application Protocol* (WAP). In a WAP environment, an intermediate WAP gateway interconnects a conventional Web server and an end-user mobile device. This gateway provides a translation service between HTTP-based Web server requests and responses and WAP-based mobile device requests and responses. WAP uses a special markup language called the *Wireless Markup Language* (WML), which is similar to HTML. The next version of WAP (2.0) is expected to be more compatible with Internet standards, including TCP/IP and a subset of HTML called Compact HTML, which will eliminate the need for a special WAP gateway. For more information about WAP, see http://www.wapforum.org.

23. So what's the bottom line? Is my cell phone-pager really connected to two separate wireless networks?

Yes. A cell phone that has both voice and paging capabilities is actually connected to two distinct wireless networks at the same time. Also keep in mind that regardless of how a service is marketed, it is important to realize that, from a technical perspective, cellular phone service is a real-time system whereas paging service is not. When someone places a call to your cell phone number, the call is first processed through your service provider's network. It then is most likely routed through the public switched telephone network (PSTN), which in turn passes it to the recipient's cellular service provider's network. Once this circuit is established, a real-time conversation takes place. A paging system on the other hand, is a store-and-forward system. When someone sends you a page, the message is routed to a paging terminal or station where it is processed, validated, and encoded for RF transmission. This message is then forwarded to other paging terminals en route to the destination terminal that services the recipient's paging device. From here the message is decoded and transmitted to the recipient's pager. No real-time conversation takes place, and a separate paging network is being used that is different from the cellular network.

24. I guess this means that my Web-enabled cell phone and Web-enabled PDA are also connected to separate networks.

Yes.

25. OK. Let's shift the discussion to wireless data networks.

The topic of wireless data networks was initially discussed in Chapter 4 where we introduced wireless local area networks (WLANs), including HomeRF and IEEE 802.11-based LANs. Given the popularity of IEEE 802.11 LANs, we will restrict our discussion of WLANS in this chapter to IEEE 802.11. Recall from Chapter 4 that there are three versions of the IEEE 802.11 standard. The first version, IEEE 802.11, supports data transmission rates of 1 Mbps or 2 Mbps and specifies three different physical layers: two are RF-based (direct sequence spread spectrum and frequency hopping spread spectrum), and one is infrared-based (diffused IR). The second version, IEEE 802.11b, supports data transmission rates of 5.5 Mbps and 11 Mbps. It is also compatible with the initial 802.11 standard and hence supports 1 Mbps and 2 Mbps rates as well. IEEE 802.11b is commonly referred to as *Wi-Fi*, for wireless fidelity, to denote IEEE 802.11b-compliant products, which foster interoperability. The third version, IEEE 802.1a, supports data transmission rates of up to 54 Mbps. All three standards use the CSMA/CA media access method (see Chapter 5) for their MAC sublayers.

26. How does a WLAN work?

Before we respond to your question directly, let's first review some of basic concepts related to wireless systems. Most wireless transmission systems in use today, including wireless LANs and cellular phone networks, generally involve a network of base stations that are traditionally wired together in some manner. When a wireless system transmits information, a base station within signal range receives the information and sends it along a traditional ground network. For instance, when a cell telephone is transmitting to another telephone somewhere in the world, the transmitting circuitry of the cell phone is communicating with an antenna somewhere nearby on a given negotiated frequency. This frequency is established when the cell phone begins transmitting over the network after a number is dialed and the cell phone's send button is pressed. The local base station that detects the strongest signal receives the transmitted data from the cell phone. In a cellular phone network of base stations, more than one base station might "hear" the transmission from the cell phone at the same time. Because the base stations are connected to each other, they can negotiate which base station will act as the master for a transmission signal from a cell phone. If the cell phone stays relatively stationary, then the same base station continues to receive data from the cell phone and forwards the data through the base network to the destination telephone network. If the cell phone is moving, as in the case of transmitting from a car, then, as we described earlier in the chapter, base stations interoperate with each other to determine the direction the signal is traveling and the next base station that is to receive the handoff. Once the handoff is complete, this new base station communicates directly with the cell phone. Through this method, the base stations coordinate the handling of the transmitting cell phone as it moves from place to place.

Let's now extend this scenario to wireless data networks. In an IEEE 802.11 LAN environment, base stations are called *access points* (APs) and each computer that is connected to the network contains a wireless 802.11 NIC. APs are directly connected to a wired LAN and act as bridge or router that interconnects the wired network to the wireless network (see Figure 4.19). APs perform two major functions: authentication and connection identification. The authentication process confirms that the wireless device is authorized to access the services provided by the wired LAN. A wireless device can be authenticated by its hardware or IP address, a password, or any other unique identifier. Connection identification involves a handshaking process that identifies which AP will serve as the master for the wireless device. An AP receives and sends transmissions to and from the various attached clients on the network. APs are typically within a maximum of 1500 feet of all the clients in a network "cluster." In cellular networks, this "cluster" is known as a cell; in 802.11 jargon, it is called a *basic service set*, or BSS. Since most readers are probably more comfortable with the cell terminology, we'll continue to use it here. As discussed in Chapter 4 and noted above, clients may communicate with their designated AP via one of two transmission methods: frequency hopping or a direct signaling spread spectrum. Both transmission systems use a variety of frequencies during the transmission between the AP and clients to find the most optimal frequency for the transmission and to avoid traffic congestion on a specific frequency. Depending on the type used, APs might be limited to the number of simultaneously transmitting clients they can support. This means that the network topology might consist of many APs, each servicing a cell of clients operating on specific frequency groups. In this way, multiple APs can support hundreds of machines in close proximity.

In addition to this hybrid wired-wireless configuration, WLANs can also be configured in a peer-to-peer topology. In this scenario, wireless 802.11 clients are equipped with software to support AP functionality. This makes direct communication between devices possible without having to use any intermediate device (see Figure 18.5). A common application of a peer-to-peer WLAN is a conference room setting where wireless workstations or other devices are placed in close proximity to each other (e.g., less than a 300-foot radius). A more simple but nondata example of a peer-to-peer wireless transmission is a walkie-talkie, which is also called the new Family Channel transmission system in the United States. These hand-held units are very small and can transmit and receive messages via specific frequencies identified by channel numbers on the units. No base station is required and the units can communicate with each other over a common frequency. Another example of peer-to-peer wireless networking is the infrared wireless transmission between two personal digital assistants (PDAs). This allows data transmissions between the two devices over a distance of a couple of meters. Each PDA is equipped with a transceiver and does not require any other intermediate equipment to facilitate communications. This type of networking may or may not require line of sight connectivity. Some peer-to-peer wireless network devices use omnidirectional transmission as well. Peer-to peer-wireless networks are not limited to the type of transmission system used, but they are usually limited to how many other receivers a transmitting station can connect to simultaneously.

27. How do WLANs handle roaming?

In the wired world, if you move your desktop unit to a different location within your company (e.g., to a different building or another floor), chances are you (or a network administrator) will have to reconfigure your machine for network access. The same is true in the wireless world. The difference is that users of wireless devices expect to be able to move from location to location without ever having to reconfigure their machines. This expectation is not unrealistic. After all, one of the advantages to wireless data networking is the ability to freely roam without being tied to a specific location. To support roaming in a WLAN environment, network cells (i.e., basic service sets) are grouped into multicells called an *extended service set* (ESS). Multicells comprise multiple APs, which effectively provide broader coverage over a larger geographical area, increase the number of users that can access network resources and services, and increase the aggregate throughput of the WLAN because less stations are sharing the throughput available within a given cell. The concept of multicells or ESS is very similar to the way in which cellular networks are organized. Furthermore, the handoff process in a WLAN environment is also similar to that of a cellular phone network. For example, as a wireless device moves into a new cell, the multiple APs communicate with each other just as cellular base stations do to determine which

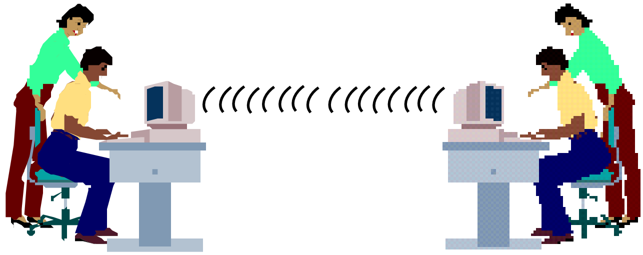


FIGURE 18.5 In a peer-to-peer wireless topology, each node is equipped with software that enables it to be an access point, thus eliminating the need for any intermediate access point. (Contrast with a hybrid wired-wireless topology as shown in Figure 4.19.)

AP will become the "master." It should be noted that the 802.11b standard does not actually specify how handoffs are performed. Proprietary InterAccess Point Protocols (IAPPs) are available from various vendors, though.

If the APs involved in a roaming negotiation process are all connected to the same LAN, then the process is relatively simple and straightforward. If the APs are connected to different LANs or subnets, however, then a network reconfiguration process that generally involves some type of human intervention must be performed. To make roaming a user-transparent process, proprietary solutions are available. There is also an IETF standard called *Mobile IP* (see RFC 2002), which provides IP address resolution and tunneling capabilities (see Chapter 3). Mobile IP makes it possible for users to roam within or outside their company's network but still maintain their home-network-based IP address. In other words, as a wireless laptop user, for example, moves from a subnet in one cell to a different subnet in another cell, the laptop's network connection is constantly maintained without any need to reconfigure the laptop with a new IP address. Mobile IP conceptually works as follows (see Figure 18.6): The home network's edge router (i.e., the router that connects the enterprise network to the Internet) runs special Mobile IP software. (Other routers within the enterprise network can also run Mobile IP software.) When a mobile computing device (e.g., a wireless laptop) enters a new cell or subnet and tries to access network services from the home network, the mobile device detects a foreign router, which must also be running Mobile IP software. Through the Mobile IP software, the mobile device receives a "care-of-address" and registers with the foreign router. A link is established to the home network router, which in turn authenticates the mobile device. Assuming everything is in order, the home network router establishes a tunnel to the mobile device. Note that none of the network service provider's internal routers need to run Mobile IP software; only the two edge routers have to run this software.

28. I see that many establishments such as airports, hotels, and stores like Starbucks are now offering wireless Internet access in their establishments.

What you are referring to is what the mainstream press calls "public Wi-Fi." Recall that Wi-Fi stands for wireless fidelity. In nontechnical circles, Wi-Fi means IEEE 802.11b-

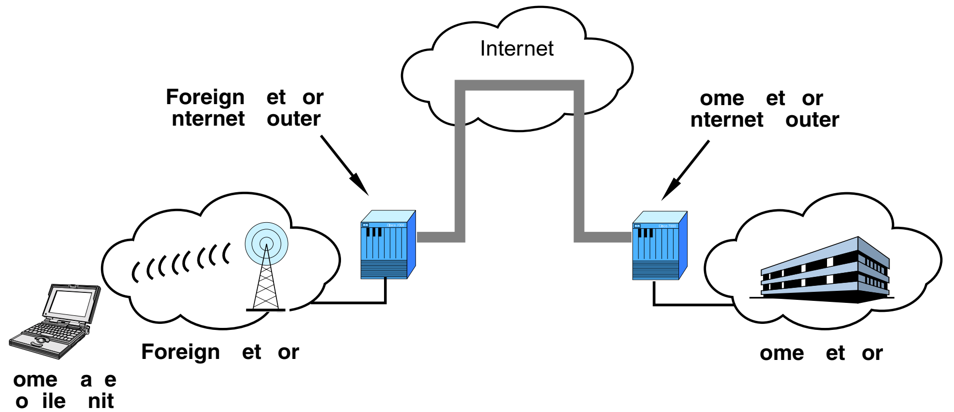


FIGURE 18.6 Mobile IP enables a roaming home-based mobile unit to maintain its IP configuration and connection to its home network. When the unit attempts to communicate with its home network, it detects a foreign router. The mobile unit registers with the foreign router and receives a special care of address (COA). A communications path is then established between the mobile unit's COA and the home network. The home network Internet router authenticates the mobile unit and a temporary tunnel is established between the two sites. Source: Adapted from Wexler, 2001.

based LANs or wireless Ethernet. However, as we discussed earlier, Wi-Fi is a label assigned to 802.11b hardware to signify that it has satisfied the Wireless Ethernet Compatibility Alliance's (WECA) rigid tests for interoperability. Thus, Wi-Fi simply indicates that an 802.11b device adheres to the IEEE 802.11b specification and will interoperate with other such devices "out of the box."

Prior to IEEE's ratification of 802.11b, wireless data networking was more of a niche market and products were expensive. When the 802.11b standard was approved, wireless data networking was validated by the general consumer because for the first time the data transmission rate of a WLAN was on par with conventional wired Ethernet/802.3's 10-Mbps rate. In addition to acceptable data transmission rates, 802.11b networks are similar to newer-model cordless telephones: both operate at 2.4 Ghz and both have approximately the same operating ranges—150 feet through walls and 300 feet in an open area. As a result of these factors, local wireless Ethernet/802.3 networks suddenly became fast and simple to establish. With the increased interest in and demand for 802.11b products, local wireless Ethernet/802.3 networks also became affordable.

Given 802.11b's attractive features, the concept of providing public Internet access via wireless Ethernet/802.3 LANs became a reality. Today, hotels, conference centers, and airports have begun establishing public 802.11b networks. Even storefront businesses, including coffee shops such as Starbucks and bookstores have jumped on the "public Wi-Fi network" bandwagon. Starbucks, for example, is expected to have public Internet access via 802.11b wireless networks in approximately 4000 of its stores throughout the United States and Canada before the end of 2003. Thus, users with a mobile device that contains an 802.11b adapter can access the Internet via any public "Wi-Fi network" without having to connect their computers into a wired Ethernet/802.3 outlet or into a telephone set for dialup access.

29. If a public Wi-Fi network is similar to a cordless phone, then does this mean that it is just as susceptible to eavesdropping as cordless phone conversations?

You touched on a very important issue related to wireless networking, namely, security. The inherent nature of RF communications lends itself to eavesdropping and 802.11b LANs are of no exception. As an illustration, consider the concept of a police scanner. If you are within range and tuned to the same frequency, then you can pick up police transmissions. The same concept is true for 802.11b networks. To understand this a little better, consider the following scenario: Let's assume you have an *x*DSL or cable modem home-based Internet connection. Let's further assume that this connection is part of an IEEE 802.11b WLAN that uses a Wi-Fi access point such as Apple's AirPort wireless hub to provide a single point of access to the Internet. This configuration not only provides you with wireless Internet access from different locations throughout your house, but it also provides anyone with a Wi-Fi-equipped laptop within 150 feet of your AirPort hub access to your Internet connection as well. Thus, as long as you are in range, you can be part of someone else's wireless LAN. This "feature" of RF and Wi-Fi networks has been exploited to the point where some business districts or neighborhoods have created Internet-free zones where users can access the Internet for free from their Wi-Fi equipped laptops. In fact, the Greater San Francisco Bay Area contains several Internet-free zones, and the Bay Area Wireless Users Group (BAWUG) maintains a list of free access points at its Web site (http://www.bawug.org). To better understand the magnitude of the Wi-Fi security issue, you are encouraged to review some of the articles available at the BAWUG Web site.

30. Doesn't Wi-Fi incorporate any security measures?

Yes it does. IEEE 802.11b supports Wired Equivalent Privacy (WEP) encryption. The problem, though, which is well-documented, is that WEP uses 40-bit cryptographic keys. Recall from Chapter 16 that key-size is critical to an encryption algorithm's strength: the larger the key the more secure the encrypted transmission. Given that DES, a 56-bit key encryption algorithm was cracked recently, WEP's 40-bit key makes 802.11b LANs extremely vulnerable to security breaches. Key size is not WEP's only problem, though. WEP also typically employs a single-key strategy, which means that every network node uses the same key, and keys must be distributed among nodes manually. Thus, not only is WEP technically deficient, it is also inconvenient to use and, hence, many people opt not to invoke it. This is worse yet because WEP at least provides some measure of protection.

31. Some people have a different opinion and believe that no protection is better than a false sense of protection, which is what WEP apparently provides to the uninformed user. Has anything been done to resolve this problem?

Yes. There have been several approaches to the problem. One approach centers around increasing the key size. Some 802.11b vendors, for example, offer proprietary-based 128-bit key encryption as an option in some of their hardware devices. According to a recently published paper, though, increasing the key size to 128 bits does not make it significantly harder to crack an encrypted transmission (see http://www.crypto.com/papers/others/rc4_ksaproc.ps). To enhance the security of 802.11b transmissions, some experts suggest using virtual private networks (VPNs, see Chapter 7), IPSec (Chapter 7), or *Remote Authentication Dial-in User Service* (RADIUS) servers to provide centralized identification and authentication. Still another approach is to release programs expressly designed to crack wireless transmissions. The presumption here is that by placing such programs in the public domain, businesses will become more careful about transmitting sensitive data via 802.11b LANs and upgrade their security measures. Two such programs, AirSnort and WEPCrack, are currently available. Finally, another approach is to address the problem with technology. Two IEEE working groups are doing just that. IEEE 802.1x is addressing overall network security and authentication and IEEE 802.11i is developing a new version of WEP that uses an approach based on IPsec (see Chapter 7). A recently released proprietary solution based on these latest technical developments is Cisco's Aironet product.

32. OK. Let's revisit an earlier question (i.e., question 23): Suppose I now have a pager, a cell phone, and a wireless adapter for my PDA, a Wi-Fi connector for my laptop, and I use an infrared connection for synchronization. I am actually connected to five different types of wireless networks?

Yes. Further keep in mind that many of those networks you are directly connected to might also be connected to different types of wireless networks on their backbones.

33. Where does Bluetooth fit into all of this?

Bluetooth, which was initially introduced in Chapter 17, is a cable replacement technology; it is not a network technology like 802.11b. Bluetooth's underlying design philosophy is cable elimination. As discussed in Chapter 17, Bluetooth enables mobile PCs, mobile phones, and other portable telecommunications devices to be interconnected via short-range radio links instead of conventional and often inconvenient cable attachments. In other words, it provides a radio interface connection instead of a cable attachment.

34. So Bluetooth involves more than just wireless data networking.

Correct. Bluetooth, which is named after Harald Bluetooth, a 10th-century Viking king, is a convergence technology (Chapter 17). If you believe the hype, Bluetooth promises a true anytime, anywhere wireless world. For example, a Bluetooth-enabled phone can become an all-in-one wireless device that will serve as an at-home cordless telephone, a cellular telephone when you are mobile, a walkie-talkie, a personal intercom system, a TV/VCR/DVD/stereo remote control unit, a garage door opener, a bug zapper, and an interface to the Internet, enabling you to send and receive e-mail messages and download Web content. (We're just kidding about the bug zapper.) Bluetooth also supports some 802.11b functions such as wireless printing and PDA connectivity. In fact, Bluetooth even operates at the same frequency as 802.11b: 2.4 GHz. Bluetooth, however, has a peak data transmission rate of 1 Mbps, making it less likely to be used for wireless data networking that involves bandwidth-intensive applications. In one sense, Bluetooth and Wi-Fi can be viewed as complementary technologies instead of competing technologies. The Bluetooth Web site (http://www.bluetooth.com) contains several usage examples.

35. If both technologies operate at the same frequency, won't their signals interfere with each other?

According to the experts, if Bluetooth and Wi-Fi are properly configured, then their respective transmissions will not conflict with each other enabling the two technologies to coexist. If you're still a little concerned about the possibility for signal interference, recall that 802.11a, which is the next evolution of Wi-Fi, operates at the 5-GHz frequency band. So by the time both 802.11a and Bluetooth are fully deployed (probably in 2004), the concern over potential transmission conflicts should be allayed. Speaking of potential frequency interference, you might be interested in knowing that Bluetooth also restricts the sending device's radio transmitter's output power to be exactly what a receiving device requires. Thus, if a receiver indicates that it is only a few feet away, then the transmitter modifies its signal strength to be commensurate with that distance.

36. What is Bluetooth's current status?

As of this writing, Bluetooth's deployment is nowhere near its initial projection. For example, it was predicted that more than 100 million Bluetooth-enabled end-user devices would be shipped by the end of 2001. In reality, the number of units shipped is approximately one-fifth of what was projected. Most manufacturers simply have not incorporated the technology into their products. One of the reasons cited for Bluetooth's nominal marketplace presence is that consumers don't yet need Bluetooth's wireless telecommunications applications, whereas its wireless data networking applications, which are practical, are being nicely satisfied by Wi-Fi. Thus, although Bluetooth and Wi-Fi are not really considered competing technologies, they are indeed competing with each other. Given these and other issues, Bluetooth's deployment has been delayed by at least a year.

37. What are some of the advantages and disadvantages of wireless communications?

The primary advantages of any wireless communications strategy include fast setup times and convenience. Installing a wireless infrastructure is by far easier and faster than installing a wired infrastructure. There is no need to drill through walls or lay cables, two very time consuming and messy tasks. The second advantage, convenience, is obvious. With wireless communications you are not tethered to a specific location by wires. True wireless communications is, as you indicated earlier in the chapter, like magic. You can go anywhere and still be connected via your mobile device.

On the flip side, though, wireless communications also has several disadvantages, including security, interference, and speed. Wireless security issues were discussed earlier so let's spend a few minutes discussing interference. In a perfect world, an RF signal follows a single path between the transmitter and receiver. Unfortunately, not all wireless transmissions occur under ideal conditions. Any type of RF transmission may encounter interference on a given frequency range by a natural phenomenon such as atmospheric static or even more interesting problems such as solar flares and sun spots. As a signal radiates, just like ripples in the pond, the signal may strike into the interference, which effectively blocks the transmission of the outbound signal to the receiver. Using the pond analogy, this would be like the signal striking a rock that is protruding from the surface of the pond. If the receiving antenna were on the other side of the rock, then the transmitted signals would not reach the receiving antenna and the receiver would not get the data from the transmitting station. If an RF signal bounces off an obstruction such as a building or the side of a mountain, then secondary signal paths are created. This can spell trouble for the transmission because instead of receiving a single signal from a single path, the receiver will receive multiple signals from multiple paths. This is similar to multimode fiber (Chapter 4) in which the distance light rays travel through the cable varies. Some rays travel longer distances from sending and receiving nodes; others travel shorter distances. The cladding layer reflects any stray light waves, causing signal distortion at the receiving end. In a similar manner, when secondary RF signals arrive at the receiver, they will have a different delay than the primary signal. In wireless jargon, these secondary signals will be out of phase with the primary signal. Out of phase signals can lead to high error rates. One way to address this problem is to equip the receiver with more than one antenna and incorporate some sort of switch to identify the best signal. Another issue related to signals bouncing off objects en route to their destination is out of order signals. Thus, the receiver must be able to reassemble signals in the correct order. A technology called *vector orthogonal frequency division multiplexing* (VOFDM), which is similar to OFDM (Chapter 4), provides this function. In short, wireless transmissions have special problems that are not encountered on wired networks. Although wired networks have their share of transmission interference problems such as near-end crosstalk (NEXT), these problems are much less likely to occur than on a wireless network of any type. This means that the technologies required for wireless networking need additional error checking and a different type of network topology than wired networks.

A third disadvantage is speed. Wireless networks are indeed getting faster and faster as evidenced by the evolving 802.11 standards. However, the data transmission rates of wireless networks still cannot compare to the wired-based speeds available today. For example IEEE 802.11a's 54 Mbps data rate is less than 1 percent of 10 Gbps Ethernet. Because of limitations with frequency ranges, signal strength, natural interference issues, and many other anomalies, there is substantial science required to increase speed dramatically in the short term on wireless networks. The flexibility of wireless networks does compensate for the lack of speed in many environments, but not all. There is a considerable amount of work under way that addresses wireless speed and distance issues. As usual, though, wireless networks benefitting from new speed and distance standards will never be fast enough or soon enough to satisfy the consumption requirements of most companies and individuals.

38. Let's assume that wireless networking's disadvantages are resolved with technology. I suspect that the convenience factor will make wireless networking's future very promising. What do you see happening?

Although we are not futurists, there are some signs that enable us to offer a peak into the not too distant future. A brief description of some future wireless applications follows:

Cellular Telephone Networks As of this writing, there is a great deal of market penetration in voice wireless networking. That technology will continue to increase in market acceptance as more and more of the population adopts cellular telephone technology with a younger and younger market. This will accelerate as the price of cellular use drops. New technologies such as 3G allow the convergence of video, voice and data in a seamless wireless environment. Already in Japan, for example, a 3G service known as *i-mode wireless Internet* has been deployed. I-mode requires specially designed 3G-enabled telephones that have traditional cellular, pager, and microbrowser Internet access via a 2-inch screen built directly into the phone. The phone also contains color television and broadband wireless video capabilities equivalent to cable television. Thus, a single handset provides a host of personal communications and personal entertainment capability. I-mode is expected to be deployed in the United States within the next year or so by AT&T.

The concept of a logical, single, worldwide cellular network might also be possible by the recent introduction of *software-defined radio* (SDR), which effectively transforms cellular phone networks into software-based systems. Although an SDR-based cellular network is probably at least 5 years away, SDR has several potential benefits. For example: (a) users can download software that will enable their cell phones to operate on different cell networks based on a different technology; (b) through software downloads, wireless providers can quickly move their subscribers to less crowded frequency bands, which could reserve some radio spectrum; and (c) users can download software directly to their handsets that will enable them to upgrade their devices or add new services.

Wireless Appliances Wireless capabilities are also being added to appliances, automobiles, and other personal communications devices to make it easier to communicate with individuals and share data and information. For example, wireless capabilities available in Asia allow a small video screen in car to (a) monitor traffic conditions, (b) provide satellite-based communications and entertainment, (c) monitor the car's status and transmit that status to a car dealer's service shop, and (d) automatically report the car's position to a service bureau or tracking company in case the car is stolen.

Professional Use Educational institutions are using wireless systems to interconnect students and school resources 24 hours a day. Classroom assignments are given out via wireless networks and homework returned in the same manner. Students using electronic books over wireless networks no longer have to worry about book inventory from local paper stock to be able to do research and learn about topics presented in class. Wireless networks also enable learning and physically impaired students to access the same information as anyone else in a manner that best suits their ability to assimilate information. Wireless technology is also being applied to the medical profession and many other vertical markets to allow quick access to patient data, financial data, records of all types, and the ability for direct interaction between doctors, patients, and other individuals such as pharmacy technicians. New laws in most countries around the world, however, may restrict some wireless growth in the medical profession due to security concerns.

Wireless Local Loop (WLL) As we mentioned earlier, installing any type of cable is labor-intensive, time-consuming, and nonportable. When installing MANs and WANs, additional complications such as geographic rights-of-way must be secured to dig up the ground to bury cable or use existing telephone poles for new cable runs. All this effort is part of what is called the *local loop* and is critical to the network infrastructures in most cities. Recently, however, many countries have opened up their airwaves to permit high-speed wireless transmissions that are roughly equivalent to the speeds of buried cables. Thus, it is now possible to place a tower and antennae within a neighborhood to provide a wireless local loop connection to homes or small business. For example, currently a major telecommunications carrier in the United States is providing a wireless residential community network that allows homes to be connected to a network at a speed of 10 Mbps. This wireless network completely bypasses the existing local loop cable infrastructure. Usually, a local telephone company owns the local loop and leases it to one or more telecommunications carriers at a premium price. By using wireless communications to bypass the local loop, a telecommunications carrier can reduce to the premise cost of a communications channel for the customer. The end result is that the customer can get a superior service at equal to or less than the cost that is traditionally imposed by a copper-based service. In addition, the local neighborhood wireless systems usually provide a multimedia capability of voice, video, and high-speed data transmission at very reasonable prices. This means that traditional copper- and fiber-based backbones are beginning to yield toward a wireless environment. The IEEE 802 committee has established the 802.16 working group to provide a standardized approach to WLL. The concept of a WLL is also especially useful in emerging countries throughout the world where a cable infrastructure does not currently exist. Places such as China and Africa, for example, which do not have even rudimentary copper capability, could be made operational with high-speed and superior communications capabilities in a very short time, and at reduced cost when compared to emplacing a fiber or copper infrastructure.

END-OF-CHAPTER COMMENTARY

You should now have a good understanding of the fundamental concepts and terms related to the current state of wireless communications and networking. As indicated in the chapter introduction, several earlier chapters also contained material related to wireless communications, most notably, Chapters 4 and 17. You are encouraged to review this earlier information relative to the material contained in this chapter.

Appendix A

Vendor Ethernet/802.3 Prefixes

Ethernet/802.3 vendor prefixes are the first six (leftmost) hexadecimal digits of an Ethernet/802.3 address. Vendor prefixes, officially called *organizationally unique identifiers* (OUIs), are assigned by IEEE, which maintains a Web site of all publicly released OUI assignments. (Not all organizations make their assigned prefixes public.) This Web site, located at http://standards.ieee.org/regauth/oui/index.html, is updated quarterly. Note that OUI assignments do not always match the original equipment manufacturer (OEM) of a network interface. Some organizations elect to license their vendor codes to other manufacturers. In such cases, the prefix will reflect the OEM and not the IEEE assignment. For example, the Ethernet/802.3 prefix 00400B is officially assigned to Cisco Systems, but field reports show that it appears on Crescendo devices. As a result, a second, unofficial list of prefixes has been published that contains reports of prefixes observed in the field. This second list, which is subject to errors, is available at http://map-ne.com/Ethernet/Ethernet.txt. Page constraints prevent us from listing a complete table that merges the official and unofficial lists in this appendix. Nevertheless, the reader is encouraged to visit either of the Web sites given to receive the most current list of vendor Ethernet/802.3 prefixes.

This Page Intentionally Left Blank

Appendix B

Using Parity for Single-Bit Error Correction

Let's define a *codeword* to have the following structure (each X_i is a bit in the data set):

$$X_1 X_2 X_3 X_4 X_5 X_6 X_7 X_8 X_9 X_{10} X_{11} X_{12} \ldots$$

Also, let every bit with a subscript that is a power of 2 be a *check bit*, denoted by r_j, which will contain redundant information for error correction. Thus, X_1, X_2, X_4, and X_8 are check bits and we replace them with r_1, r_2, r_4, and r_8, respectively.

$$\boldsymbol{r_1} \, \boldsymbol{r_2} \, X_3 \, \boldsymbol{r_4} \, X_5 X_6 X_7 \, \boldsymbol{r_8} \, X_9 X_{10} X_{11} X_{12} \ldots$$

Since the X_i bits are predetermined (they represent user data), we need a rule for forming check bits. This rule is based on parity and is defined as follows:

> Each check bit, r_j, is formed by collecting all corresponding X_i bits. These are determined by expressing the subscript of each X_i as the sum of a power of 2. For example, X_3 corresponds to r_1 and r_2 because its subscript, 3, is equal to the sum of check bits r_1's and r_2's subscripts (i.e., 3 = 1 + 2). Once the set of X_i bits is identified for each r_j, the check bit then forces the parity of each set to be even or odd, depending on the parity selected.

This rule is implemented by first expressing the subscript of each X_i bit as a power of 2:

- X_3 corresponds to r_1 and r_2 (3 = 1 + 2)
- X_5 corresponds to r_1 and r_4 (5 = 1 + 4)
- X_6 corresponds to r_2 and r_4 (6 = 2 + 4)
- X_7 corresponds to r_1, r_2, and r_4 (7 = 1 + 2 + 4)
- X_9 corresponds to r_1 and r_8 (9 = 1 + 8)
- X_{10} corresponds to r_2 and r_8 (10 = 2 + 8)
- X_{11} corresponds to r_1, r_2, and r_8 (11 = 1 + 2 + 8)
- X_{12} corresponds to r_4 and r_8 (12 = 4 + 8)

Using the list generated above, we now identify all X_i bits that correspond to each r_j:

- r_1 corresponds to $\{X_3, X_5, X_7, X_9, X_{11}\}$
- r_2 corresponds to $\{X_3, X_6, X_7, X_{10}, X_{11}\}$
- r_4 corresponds to $\{X_5, X_6, X_7, X_{12}\}$
- r_8 corresponds to $\{X_9, X_{10}, X_{11}, X_{12}\}$

Given the preceding information, we now construct a codeword for a specific data set. Let's assume that the data string to be transmitted consists of the bit pattern 10001100. Let's further assume even parity. Our goal is to generate a single-bit error correction code for the given eight data bits. This codeword will be of the form:

		$\mathbf{b_1}$		$\mathbf{b_2}$	$\mathbf{b_3}$	$\mathbf{b_4}$		$\mathbf{b_5}$	$\mathbf{b_6}$	$\mathbf{b_7}$	$\mathbf{b_8}$
r_1	r_2	X_3	r_4	X_5	X_6	X_7	r_8	X_9	X_{10}	X_{11}	X_{12}

Substituting the data bits into this structure we have:

		$\mathbf{b_1}$		$\mathbf{b_2}$	$\mathbf{b_3}$	$\mathbf{b_4}$		$\mathbf{b_5}$	$\mathbf{b_6}$	$\mathbf{b_7}$	$\mathbf{b_8}$
r_1	r_2	1	r_4	0	0	0	r_8	1	1	0	0

We now determine the check bits (assuming even parity).

- Since r_1 corresponds to $\{X_3, X_5, X_7, X_9, X_{11}\}$, r_1 is based on the parity of bits 10010. Thus, r_1's parity is 0.
- Since r_2 corresponds to $\{X_3, X_6, X_7, X_{10}, X_{11}\}$, r_2 is based on the parity of bits 10010. Thus, r_2's parity is 0.
- Since r_4 corresponds to $\{X_5, X_6, X_7, X_{12}\}$, r_4 is based on the parity of bits 0000. Thus, r_4's parity is 0.
- Since r_8 corresponds to $\{X_9, X_{10}, X_{11}, X_{12}\}$, r_8 is based on the parity of bits 1100. Thus, r_8's parity is 0.

As a result, the codeword is:

		$\mathbf{b_1}$		$\mathbf{b_2}$	$\mathbf{b_3}$	$\mathbf{b_4}$		$\mathbf{b_5}$	$\mathbf{b_6}$	$\mathbf{b_7}$	$\mathbf{b_8}$
0	0	1	0	0	0	0	0	1	1	0	0

This is the data string that's transmitted. Let's now assume the receiving node receives the string 001001001100. A visual inspection clearly reveals that the sixth bit from the left is incorrect; it's 1 when it should be 0. Unfortunately, the receiving node cannot see this. It must rely on an algorithm to detect and then to correct the error. This is done as follows:

1. Check the parity of each check bit using the data string that was received.
   - r_1 = 10010. Given even parity, $r_1 = 0$ = Correct (C)
   - r_2 = 11010. Given even parity, $r_2 = 1$ = Error (E)
   - r_4 = 0100. Given even parity, $r_4 = 1$ = Error (E)
   - r_8 = 1100. Given even parity, $r_8 = 0$ = Correct (C)
2. Based on the check bits, which are based on parity, r_1 = C, r_2 = E, r_4 = E, and r_8 = C. Let C = 0 and E = 1. If we multiply the subscript of each check bit by its corresponding C or E value we get the following:

$$\begin{aligned} &\mathrm{C}(1) + \mathrm{E}(2) + \mathrm{E}(4) + \mathrm{C}(8) \\ &= 0(1) + 1(2) + 1(4) + 0(8) \\ &= 0 + 2 + 4 + 0 \\ &= 6 \end{aligned}$$

Only check bits 2 and 4 are in error. This implies that bit 6 is incorrect because only bit 6 is checked by r_2 and r_4. Thus, the sixth bit is in error and should be complemented.

Appendix C

Guidelines for Installing UTP Cable

1. **Know and understand the related premises wiring standards.** It is important that you are knowledgeable of the published standards related to UTP premises wiring. Network cable installation is governed by a standard known as EIA/TIA-568. This standard was jointly developed by the Electronic Industries Association (EIA) and the Telecommunications Industry (TIA). EIA/TIA-568 is a North American standard that specifies the type of cable that is permitted for a given speed, the type of connectors that can be used for a given cable, and the network topology that is permitted when installing cables. The standard also defines the performance specifications that cables and connectors must meet. In short, EIA/TIA-568 represents a comprehensive standard for premises wiring that addresses network design and performance characteristics for physical media. Familiarize yourself with this standard as well as EIA/TIA-569 (*Commercial Building Standard for Telecommunications Pathways and Spaces*) standards. Although a non-standard cabling system might work, this is something you should not chance. Also, do not let the issue of cost take precedence over standards. It is better to spend a little more money for something that is consistent with published standards than not. Standards were written for a purpose and should not be ignored.
2. **Check local building codes.** Prior to any wiring installation, contact the facilities people of your company to ensure that the planned cable plant is consistent with electrical and fire codes. Also consult the following: Section 800 of the National Electrical Code for telecommunications cable installations; EIA/TIA-607, the standard on the grounding and bonding of a building premises cabling plant; and IEEE 1100-1992, *Recommended Practice for Powering and Grounding Sensitive Electronic Equipment* (known as the *Emerald Book*).
3. **Think globally and locally.** If your cable installation is parochial, you are encouraged to think about how it might impact the entire organization. At the very least, and as a courtesy, contact your organization's designated network administrator and share your plans with this individual.
4. **Plan and document.** Not enough can be said about these two activities. Planning is extremely important to ensure that all factors have been considered including: who is responsible for installation, termination, and certification of the installation; who will

manage the cable plant; whether existing cable should be used; what minimum specifications the cable plant must meet; what interruptions there will be to company business prior to, during, and after the installation; what facilities modifications are needed to facilitate and support the installation; what future needs might be and how the current installation lends itself to meeting these needs; and so forth. As for documentation, it is extremely important that network maps be constructed and maintained. There should be a general topology map and a wire map that identifies cable lengths, termination points, grounding information, and so on. In our experiences, we found it easier to maintain handwritten maps than to use a software program. Copies of these maps should be in a centralized location and at key sites throughout an organization. Modifications to the cable plant should be documented at the time they are made.

5. **Be sensitive to physical security and the physical environment.** Include security issues as part of the planning process. Try to acquire dedicated space with a locked door to serve as wiring closets. Also make certain these areas are clean, air-conditioned, and free of corrosive material (see Box 16.3.).
6. **Test the cable on the spool and after installation.** This is important. If you only test the cable after installing it, you cannot be certain if any detected postinstallation problems are the result of the installation process or bad cable.
7. **Label the cables.** Regardless of how diligent you are in documenting the cable plant, there will be instances when your map is not readily available when examining the cable plant. By labeling both ends of a cable with Avery stickers or their equivalent, you will have a built-in map that identifies each cable, its termination point and length, and electrical characteristics, among others.
8. **Inspect the installation, ensure the cable has been tested properly and accurately, and get a warranty from the cable installer.** Check for the following: Make certain cables have been installed neatly and securely; look for cinch marks and worn insulation (there should be none); be sure cables are not resting directly on ceiling tiles, have not been pulled near anything that radiates heat, are not near electrical fixtures, and have not been placed where people can step on them or roll over them in a chair; make sure that the minimum bend radius has not been exceeded; and ensure that cable runs do not exceed maximum recommended lengths. Also, make sure the cable plant is certified; that is, it meets all published certification standards such as those found in the EIA/TIA Technical Service Bulletin (TSB)-67. Finally, since a certified installation does not guarantee quality of work, obtain a warranty (at least 10 years) on the installed cable.
9. **Consider STP rather than UTP.** Before committing to UTP, you might want to consider installing STP. This would be prudent especially if the installation site is a source of high electromagnetic interference (EMI).
10. **Install a sufficient number of pairs.** Consider installing 16 pairs to each location—four pairs for a LAN, four pairs for voice, and eight pairs (four each) for alternate LANs. Although using fewer pairs might be functional, it is much more difficult to manage in a live network. Most networks will be upgraded in the future, and this will mean parallel network connectivity for a period of time while the new network is tested and the old one is running.

Appendix D

Network Design and Analysis Guidelines; Network Politics

Network design and analysis refer to the essential methods required to properly design a network. A properly generated network design provides the following benefits:

- Proper analysis of existing equipment for network installation.
- List of requirements for network installation.
- Proper configuration of network components for optimum cost savings.
- A flexible and adaptable network topology.
- Correct selection of network hardware and software.
- Documentation of the network for future enhancements and modifications.
- Migration path into future network technologies without redesign.
- A long network life cycle.
- Interconnect paths and methods for multiple network architectures.
- User analysis and configuration of network resources for optimal use.
- Network management plan and methodology to reduce downtime and allow for maximum use of available resources.
- Expectations for performance, reliability, and usability.
- Optimal programming environment for network applications.
- Training needs for programmers, users, and network managers.
- Recurring expense forecasting and budgeting methods.
- Network support needs (programming, management, user support).
- Use of mathematical modeling tools to help ensure the success of the network design and topology.
- Optimal design to prevent network congestion, queuing delay, and proper placement of routing and management resources on the network.

Unfortunately, many network administrators do not take the time to perform a network design and analysis. Instead, they rely exclusively on vendor recommendations. Network design and analysis involve much more than ordering hardware and software from a vendor. If you want to trust a vendor, that is your choice. However, few vendors are qualified network designers, and those who are do not offer this service free of charge. Be forewarned: You get what you pay for, and in the end, you will be the person who will be

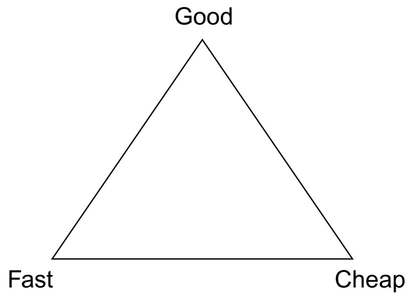


FIGURE D.1 This triangle is called "Truman's triangle," named after its inventor, Truman Reynolds. Truman's triangle serves as a good guide for anything that is network related. Pick any two vertices of the triangle, and then apply the word *not* to the remaining description (vertex). For example, if you want a network that is fast and good, it will *not* be cheap.

blamed for poor network performance or problems, not the vendor (see Figure D.1). *Networks are neither trivial nor cheap.* If implemented incorrectly, expect to spend more time correcting problems, increased network downtime, and increased stress in your life.

The Steps of Network Design and Analysis

1. **Identify the Need.** Today, networks are an integral part of nearly every organization, and network managers are constantly faced with decisions to upgrade an existing network or install a new network. Before any action is taken, make sure you really need to upgrade or install a network. Don't be pressured into a new installation or an upgrade. Thus, before you swap out your token ring network for Ethernet/802.3, before you decide to replace your 10-Mbps Ethernet hubs with switches, or before you decide to upgrade your backbone from Fast Ethernet to Gigabit Ethernet, justify the need for doing so.
2. **Identify Function and Cost.** What is the network supposed to do and how much is it going to cost? The first part of this question can be answered by defining what functionality the network is to offer. If it is e-mail, file transfer, task-to-task communications, great, but write it down. In addition to functionality, there is the issue of cost. Networks are just like systems—they have a life cycle. Networks require periodic upgrading and expansion. They have recurring costs such as software and hardware maintenance, training, telco service, and replacement equipment purchases. They require personnel to manage and maintain the network components and might require custom software development, which implies costs for software engineering or application programming. Networks are expensive to install and operate over a period of time because they are service intensive.
3. **Site Survey.** A site survey involves the careful examination of company facilities, building architecture, phone facilities, and existing computer hardware and software components. It also includes the examination of existing contracts, power facilities, HVAC facilities, wireways and wire centers, electromagnetic interference possibilities, radio frequency interference possibilities, safety issues, security issues, building wiring and

fire codes, electrical codes, reception and shipping facilities, building maintenance capabilities, on-site or vendor maintenance capabilities, and other related items. As a result, a thorough inspection of the network site is necessary. Furthermore, this inspection should not be done alone. You should include the director of management facilities or equivalent, as well as managers of departments that might be affected.

4. **Basic Design, Data Collection/Reduction, and Data Analysis.** This step involves the use of mathematical modeling tools (manual and computer-based) to address issues such as data flow ratios, probabilities of error, queuing delays, interconnect problems, least-cost network topological layout, routing paths, and redundancy paths. Following the modeling of the network, a financial analysis is done to determine how much the network is going to cost to implement, start up, maintain, and expand. An assessment analysis is also performed to identify networks that are most useful (closest to desired functionality) and least useful (on the right track, but not closest to desired combination of price, performance, and ease of use). Finally, an analysis of personnel needs and operational considerations is performed. This analysis describes the type of personnel necessary to get the job done, what kind of personnel will be needed for the day-to-day support of the network and its related components, and the costs associated with such personnel.
5. **Formal Design Document.** The last step in the design and analysis of a network is the formal design document. This document is a summary of all the previous steps. It documents the rationale for the network design and includes a description of the components, a network topology, a wiring diagram, expansion capabilities, expected life cycle, applications support environment, network management environment, potential problems, data throughput analysis, testing and verifications procedures, identification of network installation resources, an implementation timetable, personnel and training needs, cost analysis, and risks. This document is the backbone of the network design and serves as a guideline for implementation and expansion.

Why go through all of this? The answer is simple and complex (sort of the yin and yang of networks): It's proper business procedure and it reduces potential risk. Many people approach network design and analysis in the same manner as buying stocks—they use the "gut-feeling" approach; they rely on emotion instead of reading a prospectus, doing some research, or hiring a financial planner or stockbroker. They say that network design and analysis activities are not necessary to install wire, hardware, and software. They also "prove" this by identifying other installed networks, which did not undergo a design and analysis, but are working without any problems. This might indeed be the case. Sometimes it works and sometimes it doesn't. However, just as you might get lucky in the stock market using a gut-feeling approach, chances are you will be less fortunate lacking the design and analysis of a network. In fact, studies have been done that indicate a gut-feeling approach works only about 25% of the time. When looking at network design and analysis, the main mistake many companies make is they approach a network in the same manner that they approach the "self-broker" methodology. This is not prudent because networks have some fairly serious restrictions that require both formal training and experience before someone is in a position to design a network properly.

Components of Good Network Design

Good network design is characterized by the following main points:

1. **It meets or exceeds the needs defined in the specification.** This is self-explanatory.
2. **Its capabilities are obvious and beneficial.** If a network is obviously useful and beneficial, the life of the network can be more easily justified, and the corporation will utilize the network and its capabilities in all appropriate areas. Thus, the capabilities of a network should be demonstrable upon demand so that its merits are recognized by management. If this can be done, then the needs of a network can be addressed properly and funding for the network is assured.
3. **It is cost effective and cost predictable.** Cost effective is obvious; cost predictable, however, is not. Over the period of life of any network, there is a need to predict the costs the network will incur upon corporate finances. Some obvious costs include component upgrades, software maintenance, hardware maintenance, and operational management tools and personnel. Some of the less obvious costs that need to be predictable are documentation, training, code maintenance, system downtime due to network component failure, productivity delays due to congestion or network failure, and consulting assistance. These costs should be part of the design so no surprises surface after the fact.
4. **It is capable of being managed by both system-manager-level and network-manager-level personnel.** This is necessary to protect a company from the eventual exit of trained network personnel. Both network and systems people should be capable of managing an organization's network.
5. **It is user transparent.** Users should not have to know the intricacies associated with the network. They should be able to use it in the easiest and most efficient manner possible.
6. **It is easily expanded.** A network should be capable of expansion without redesign. Management structures change, corporate directions change, people change. There is no reason that the network should not be able to change as well.
7. **It is well-documented.** Proper documentation of how a network was configured, why, and the politics behind it is critical to future support, expansion, and interpretation of the original goals of the network. Furthermore, any network-related activities (e.g., errors detected and how they were resolved, changes made to configuration files, breaches of security) should be logged for future reference and network history.
8. **The technology is state of the practice, not state of the art.** State of the practice means that the network architecture reflects proven, useful, current technology that is not leading edge. Why not leading edge? The reason is simple. Leading edge technology is good for daredevils, but it has no place in a business or engineering environment unless it is the only reasonable method to do a job. Leading edge technology is usually bug- and problem-laden, few people understand it, it imposes unnecessary risk upon the functionality of a network, and it might not receive wide acceptance upon review by the industry and other vendors. State of the practice technology reflects what is in use. It might not be the latest on the market, but it is proven, reliable, manageable, cost predictable, and there is plenty of talent to manage the technology that is

available. When working with and configuring networks, it is essential that cost and risk constantly be kept in mind.

9. **It is supportable and maintainable at all node locations.** Regardless of where a network node is located, network managers should be able to have access to that location to provide necessary support or maintenance if needed.
10. **Network diagnostics and management are thought out and available.** If network diagnostic and management issues are not addressed a priori, then users suffer and network managers cannot be expected to manage the network properly. As a result, attention must be given to such issues as management strategies, type of diagnostic tools needed, appropriate training and support for network managers, and disaster-recovery procedures.
11. **The network provides future interconnectivity.** Do not configure a network for one type of computer. Although the battle for the desktop essentially has been won by the Intel platform and Microsoft Windows, this does not mean that you only need to consider this hardware-software platform. The network should provide for interconnectivity among different products and architectures that either currently exist or might emerge as a result of network and corporate expansion.
12. **The network has predictable performance as loading changes.** System performance will change over the life of any system. When a network is initially configured, it is generally underutilized. As time passes, though, it becomes overutilized to the point where its performance is impacted negatively. This underutilization/overutilization syndrome causes problems in network design. As a result, when considering network performance, it is necessary to look at not only the network topology and components, but also the loading history of critical processors and applications to ensure that there is a clear picture of how fast and how well things are going to perform.
13. **Its load on networked systems is predictable and reasonable.** Increased network use will eventually impact the performance of networked systems. For example, hub, switch, or router performance can degrade, and end user workstations (particularly those running a single-task operating system) can freeze up. A good network design will take these issues under consideration and provide some means for addressing them.
14. **It provides security adequate to the corporate or application needs.** Accept the following dictum: Networks are not secure. Although it is possible to make it quite difficult for someone to break into a network, you cannot keep out a professional. If someone wants to get into your network badly enough, there is very little you can do to stop it from happening. To ensure the success of a network, a security audit is performed during the design phase to identify sources of network and component vulnerability and what can be gained through penetration. This information is then presented to management, which will decide the resources needed to address these deficiencies.
15. **It supports system upgrades and enhancements.** It is impractical to expect network software and hardware to be upgraded simultaneously within all systems as new hardware and software become available. As a rule, any systems upgrade will disrupt operations on an entire network if the upgrades are not carefully considered networkwide. A good network design will allow for network and systems upgrades and provide for a

method to do both. Good network design requires that the future be considered as well as the present.

16. **It survives the politics of the company and provides for political needs.** Network design must encompass not only the needs of the technical end, but also the needs of the practical end. That is, it must address the day-to-day needs of the corporation as well as the political needs of its management. For a network to survive the political environment, care must be taken to allow proper placement of the network in the political environment. If the network is not properly introduced, placed, and controlled by the appropriate level of management, the political machine can turn on the network and associated personnel and proceed to squash everything in its path.

Network Politics

When designing and configuring networks, funny things creep in that make no logical sense, have no basis for reasonable reality, and are of questionable use. These things are called *politics*. In a networked environment, politics take on a whole new role. The more nodes of a network, the more likely politics will invade the design and operation of the network, especially when competition exists between the various departments networked together. Politics are further exacerbated if competing departments depend on each other for technical support, assistance, and worst of all, funding. Another problem is that of machine ownership. Although all machines are owned by the corporation, department managers get very possessive over the systems and resources they have a day-to-day interest in using and adopt a "my machine syndrome." Avoiding network politics generally tends to increase the political problem as a whole. To help keep politics to a minimum, consider the following items:

1. Spend time describing and documenting to users what you do and why.
2. Always get the local management at each location involved.
3. Do not underestimate any network user.
4. Good rapport is all-important when dealing with personnel from different departments. Get to know the users and managers at all locations and keep everyone in the loop.
5. Do not take an attitude of "I'm technical and don't do politics." This will increase your political problems in the long run. In the networked environment, politics are part of the design and support. Get used to it.
6. If possible, get some sensitization training. There is nothing worse to a user than a surly answer from the support organization. Try to show some empathy toward a user, but keep the sympathy to a minimum.
7. If you do not write well, get some training. Written communication is very important to tracing problems in the political structure.

If a political situation arises, take action but do not react to the situation. Many political moves occur due to power plays, inconsideration, or just to instigate a situation. If one arises, do not let your emotions get in the way of solid, logical reasoning and fact. You also must learn to discern a political problem from a real nonpolitical problem. If there is a system down, that's a real problem; don't escalate it into a political one.

Appendix E

X.25

In 1976, the Consultative Committee for International Telegraph and Telephone (CCITT) formally approved "Recommendation X.25: Interface Between Data Terminal Equipment (DTE) and Data Circuit-Terminating Equipment (DCE) for Terminals Operating in the Packet Mode on Public Data Networks." This standard, which has been periodically revised throughout the 1980s and 1990s, corresponds (although not exactly) to OSI's first three layers. In the original standard, X.25 referred to these layers as levels, namely, level 1, level 2, and level 3. This terminology was discontinued in later revisions of the standard, but it is still occasionally used in the literature today. Although many people refer to it as one, X.25 is not a network. Instead, it is a formal set of specifications for connecting data terminal equipment (e.g., a computer or a terminal) to a public packet-switched public data network (see Figure E.1). The most recent version of this protocol, adopted in 1996, also specifies how this connection is established when access is made by a dedicated circuit. X.25 provides data transmission speeds up to 2 Mbps and provides error-free transmissions. X.25 is an international standard for data communications in wide area networking (it does not support voice communications, though) and is extremely robust in transmitting data over noisy lines. It is also one of the oldest WAN protocols.

X.25's Physical Layer

X.25's physical layer addresses the manner in which data are transferred across the DTE-DCE link. This includes the physical and electrical characteristics of the media, the type of connectors, and the signaling scheme. In short, it specifies the interface between the DTE and DCE. Instead of defining these interface characteristics, X.25 references the X.21 interface standard. X.25 also supports V.35 as well as the EIA RS-232C standard for the DTE-DCE interface.

X.25's Link Access Layer

The link access layer of X.25 provides reliable end-to-end data transmissions across the DTE-DCE link. X.25's layer 2 is defined by the *Link Access Protocol-Balanced* (LAP-B), which provides link access specifications for frame composition, flow-control procedures, and error-checking methods. LAP-B is what ISDN's *H* channel uses for its layer-2 protocol for packet-switched connections. LAP-B is also a subset of the *High-Level Data Link*

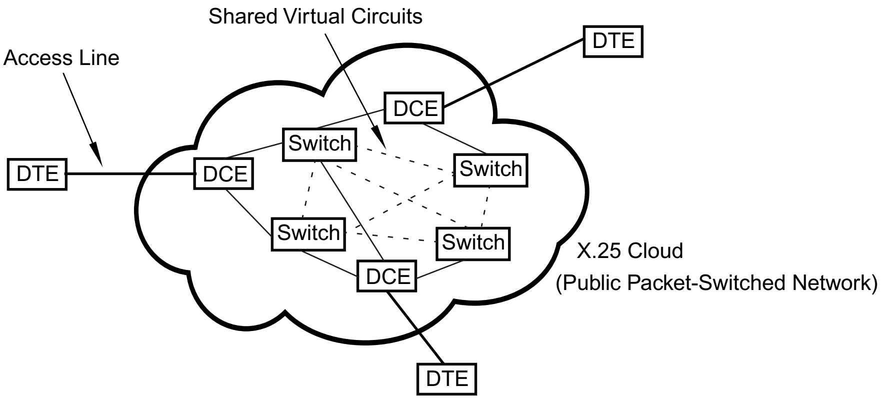
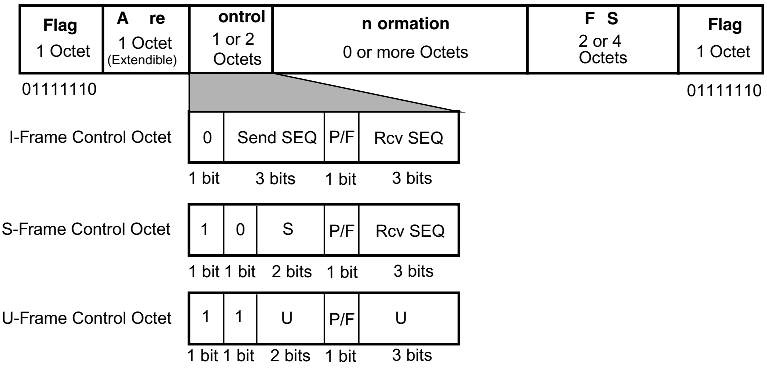


FIGURE E.1 X.25 is a set of specifications that addresses the interface between the user's equipment and the network. In X.25 terminology, the user's equipment is called data terminal equipment (DTE) and typically comprises a computer as well as any modem or line interfaces located on the customer's premises. Access to the network is provided via a physical link between the DTE and the data circuit-terminating equipment (DCE), which serves as a connection device between the user's end equipment and the network switches (i.e., the service provider's equipment). X.25 does not specify how the WAN operates; it simply provides a standard for accessing the WAN.

Control (HDLC) protocol, which is one of the foremost and influential data link control protocols ever developed. Since many current data link control protocols are based on HDLC, the general format of HDLC frames is shown in Figure E-2.

Excluding the flag fields, which contain the start-stop bit pattern 01111110 to mark the beginning and ending of a frame (bit-stuffing is used to guarantee the uniqueness of this start-stop flag), HDLC frames are partitioned into at most four fields: address, control, information, and frame check sequence (FCS). Furthermore, all fields except the data field are of fixed length. When there are no frames being transmitted across the DTE-DCE link, the start-stop flag sequence is usually repeated indefinitely. A brief description of these fields follows:

- **Address**—The address field identifies the sending or receiving node and has a standard length of eight bits, although it can be extended in multiples of seven bits to accommodate longer address sizes. The first bit of the address octet is either 1 or 0, and the remaining seven bits specify a station's address. If the leading bit is 1, then the address field contains the last part of the address. A leading bit of 0 specifies that the address field is being extended and contains at least one more octet. When only two devices are interconnected (i.e., there is only one station at each end of a link), then received frames will have exactly the same address. Hence, the address field is not required in this type of configuration—known as *balanced mode*—but is included primarily to maintain a standard frame format. The address field is necessary, however, in *unbalanced mode*, which involves the interconnection of multiple devices in parallel. An unbalanced mode configuration consists of a primary station that is interconnected to



ontrol Fiel e cription or ac Frame pe

Frame ontrol ctet

¥ Send SEQ This is the sequence number of the frame. Either a three-bit (modulo 8) or seven-bit (modulo 128) sequence number may be used. In the case of the latter, the control octet must be extended to two octets.

¥ P/F (See separate description below.)

¥ Rcv SEQ This is the sequence number of the rst unacknowledged frame. It is determined by adding 1 to the sequence number of the last acknowledged frame.

S Frame ontrol ctet

¥ S This two-bit eld is used to provide ow and error control. Four commands or responses are supported.

Receive Ready (RR): Used as a positive acknowledgment or to indicate that a station is ready to receive I-frames.

Receive Not Ready (RNR): Used as a positive acknowledgment or to indicate that a station is not ready to receive I-frames. RR and RNR together provide a stop-go type of ow control.

Reject (REJ): Informs sender that speci c I-frames were either not received or discarded due to errors and hence should be retransmitted. This represents a negative acknowledgment and infers a Go-Back-N protocol.

Selective Reject (SREJ): Similar to REJ except only select frames are retransmitted instead of all the frames with sequence numbers corresponding to a particular gap.

¥ P/F (See separate description below.)

¥ Rcv SEQ (Same as I-frame.)

Frame ontrol ctet

¥ U The U elds are used to provide additional link control information. The two- and three-bit elds are combined to provide a single ve-bit eld, which supports up to 32 commands or responses. Commands used to set the desired mode of operation, including extended mode for seven-bit sequence numbers, follow. (*Note*: SABM and SABME are used in X.25.)

Set Asynchronous Response Mode (SARM) and *Set Asynchronous Response Mode Extended (SARME)*
Set Asynchronous Balanced Mode (SABM) and *Set Asynchronous Balanced Mode Extended (SABME)*
Set Normal Response Mode (SNRM) and *Set Normal Response Mode Extended (SNRME)*
Disconnect: Terminates a previously selected mode and places link in an idle state; used for a normal disconnect.

Selected U responses include the following:

Unnumbered Acknowledgment (UA): Acknowledges receipt and acceptance of a U command such as SARME.

Frame Reject (FRMR): Speci es that a received frame was not formatted properly (e.g., I-frame did not contain an integral number of octets).

Disconnect Mode (DM): Ungraceful link termination; used as an error-recovery procedure.

P F it e cription

The P/F bit is known as the *poll bit* when sent in a command; it is the *final bit* in a response. It is used as a form of handshaking: When a P bit is set in a command, it must be matched by an F bit in the corresponding response. In *normal mode*, a primary station will initially send a command with the P bit set requesting I-frames. The sender of I-frames then sets the F bit to indicate the end of a response. In *asynchronous mode*, every S or U command sent with the P bit set must have the F bit set in corresponding responses. Furthermore, an I-frame with the P bit set, must have a corresponding RR, RNR, or REJ frame with the F bit set.

FIGURE E.2 Frame format of the high-level data link control (HDLC) protocol. X.25's Link Access Protocol-Balanced (LAP-B) is a subset of HDLC.

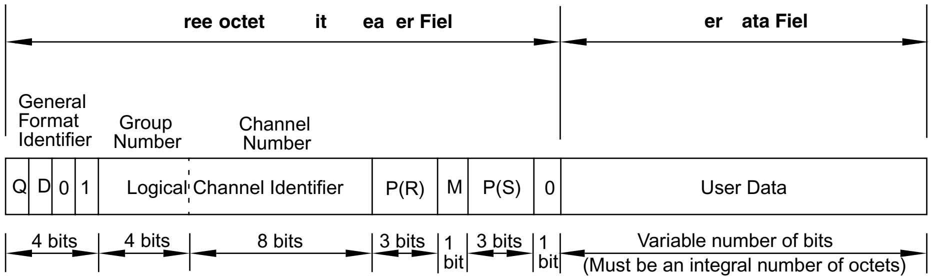
at least one secondary station. A primary station controls the link and issues commands; secondary stations are controlled by the primary station and issue responses to the primary station's commands. Since a primary station maintains separate logical links to secondary stations, the address field is essential for unbalanced mode configurations.

- **Control**—The control field specifies the type of frame being transmitted. As can be seen from Figure E.2, HDLC defines three different types of frames. *I-frames* are information frames that are used to send actual user data. Designated by a leading 0 bit in the control octet, I-frames are the only frames that can transport user data and hence are regarded as HDLC's principal frame type. *S-frames* are supervisory frames and contain only the flag, address, control, and FCS fields since they do not carry user data. Designated by a leading 10 bit-pair in the control octet, S-frames are used to transport sequence numbers that represent acknowledgments for received frames. S-frames are also used for flow and error control. *U-frames* are unnumbered frames that provide additional link control information. U-frames are designated by a leading 11 bit-pair in the control octet. A description of the control octets for each of these frame types is provided in Figure E.2.
- **Information**—The information field contains actual user data. Although theoretically this field can contain a variable number of bits, in practice it usually comprises an integral number of octets.
- **FCS**—The frame check sequence field provides error control via either the 16-bit CRC-CCITT or alternatively through CRC-32.

X.25's frame format mirrors HDLC's frame structure. As a subset of HDLC, X.25 only provides *asynchronous balanced mode* (ABM), which is set as part of the U-frame control octet shown in Figure E.2. HDLC-ABM is a connection-oriented service that uses seven-bit sequence numbers (modulo 128). Thus, an extended control field is used.

X.25's Packet Layer

X.25's packet layer specifies a virtual circuit service for transporting X.25 packets across the packet-switched network. Two services are available: permanent virtual circuit and virtual call. Both provide a point-to-point, connection-oriented transport mechanism. X.25's *permanent virtual circuit service* is a fixed virtual circuit that requires no call initiations. This is similar to a leased line service except the circuit is virtual, which implies a logical connection instead of a physical one. This means that the bandwidth is shared among multiple sites instead of being dedicated between two sites. X.25's *virtual call service* is similar to the permanent virtual circuit service except it represents a standard virtual circuit connection. This means that a virtual circuit between two DTEs is dynamically established. This requires a call initiation procedure in which the initiating DTE transmits a request packet to the destination DTE requesting that a link be established. The destination DTE in turn responds with a call accepted packet that accepts the initiating DTE's request and confirms that the circuit is established. X.25 can support up to 4096 simultaneous DTE-DTE virtual circuits on a single DTE-DCE link. The format of an X.25 data packet is shown in Figure E.3.



General Format enti er This four-bit eld speci es the packet format. In general, the GFI eld is ??01 for packets with 3-bit sequence numbers and ??10 for packets with 7-bit sequence numbers. The eld shown speci es that the packet uses 3-bit sequence numbers. The D bit is for delivery con rmation; if set, then end-to-end packet acknowledgment is required. The Q bit is a data quali er bit.

ogical annel enti er This 12-bit eld speci es the virtual circuit number; it is composed of a 4-bit group number and an 8-bit channel number.

P This is the packet receive sequence number.

This is the more data bit. If set to 1, then it indicates to the receiving DTE that the current packet was unable to support the entire user data, and hence, subsequent data packets contain the remaining part of the data.

P S This is the packet send sequence number.

er ata This eld contains the actual user data. The maximum length of this eld varies, but it is generally 128 octets. Other maximum lengths allowed vary from 16 to 4096 octets. As noted above, this eld must contain an integral number of octets.

ote

1. The packet format shown is that of a data packet. Other types of packets include control packets, as well as receive ready (RR), receive not ready (RNR), and reject (REJ) packets, which are similar to the corresponding LAP-B S-frames. The difference is that these packets operate at the packet layer.
2. X.25 headers are either 24 bits or 32 bits in length. For example, data packets that use 7-bit sequence numbers have a 32-bit header.
3. Some packet headers also contain an 8-bit *packet type identifier* eld, which identi es the type of packet (e.g., a call request packet).

FIGURE E.3 Format of an X.25 data packet.

This Page Intentionally Left Blank

Glossary

AAL See *ATM Adaptation Layer*.

AAUI See *Apple Attachment Unit Interface*.

Access Line The *local loop* in *frame relay*. Also called a *port connection*.

Access Point A wireless repeater that is connected to a wired network; it provides wireless communications devices access to the services and resources of the wired network.

Active Monitor A station on a token ring network that oversees the ring and ensures that it is functioning properly. Also called a *monitor station*.

Address A unique number assigned to a device to identify its location within a network. An address also can uniquely identify a network application process.

Addressing A network concept that describes the process of assigning unique identification numbers (called *addresses*) to a networked device.

Address Resolution Protocol (ARP) An Internet protocol that binds a node's *IP address* to its corresponding *MAC* sublayer (hardware) address.

ADoS See *Application-Level Denial Of Service*.

ADSL See *Asynchronous Digital Subscriber Line*.

ADSL Lite A slower *ADSL* in which downstream rates equal 1 Mbps and upstream rates equal 128 kbps. Intended primarily for homes. Also called *G.Lite*.

Advanced Encryption Standard (AES) A new encryption algorithm, formally called Rijndael (pronounced "Rain Doll," or "Rhine doll") designed to replace *DES*.

Advanced Mobile Phone System (AMPS) A first *generation* analog-based *cellular communications network* that operates in the 800 MHz band, supports 832 channels, and uses *FDMA* as its access technology.

AES See *Advanced Encryption Standard*.

Alignment Error An Ethernet/802.3 frame that does not end on a "byte-boundary."

Always On/Dynamic ISDN (AO/DI) An initiative from the Vendor's ISDN Association (VIA) in which a portion of the *D* channel, which is always active and constantly connected to the provider's switch, is used to transmit user packet data.

AM See *Amplitude Modulation*.

Ambient Noise Electrical *noise* that is always present and is generated primarily by transmission equipment such as transmitters, receivers, and repeaters. Ambient noise also can be induced by external sources such as fluorescent light transformers, electrical facilities, and heat. Ambient noise makes it difficult for receiving equipment to distinguish between incoming signals. Also called *thermal noise*.

Amplitude A characteristic of a carrier signal that represents the signal's strength. It is the level of voltage on a wire or the intensity of a light beam on a fiber-optic cable. Mathematically, the amplitude of a periodic function is one-half the difference between the maximum and minimum values of the function.

Amplitude Modulation (AM) A *modulation* technique in which a carrier signal's strength is altered by varying the signal's voltage.

Amplitude-Shift Keying (ASK) Similar to *amplitude modulation* except amplitude-shift keying is the more appropriate term when used in the context of converting digital data (or signals) into analog signals at the sending end and then back to digital form at the receiving end. The amplitude of a signal is altered so that it conforms to digital data (0s and 1s); one amplitude is used to represent a binary 0 and a second amplitude is used to represent a binary 1.

AMPS See *Advanced Mobile Phone System*.

Analog Any physical device or signal that varies continuously in strength or quantity over an infinite range of voltages or currents. An example is voltage in a circuit.

Analog Communication Any communication method based on analog principles. In analog communications, signals flow across a wire in the form of electromagnetic waves. These waves resemble a sine curve and have the following three characteristics: *amplitude*, which is the level of voltage on a wire (or the intensity of a light beam when dealing with fiber-optic cable); *frequency*, which is the number of oscillations, or cycles, of a wave in a specified length of time; and *phase*, which is the point a wave has advanced within its cycle. Typically associated with voice transmission rather than data transmission because voice transmission facilities, such as the telephone, were initially analog based.

Analog Data Any data type that is continuous in nature and varies in strength and quantity over time. Analog data can be represented as either *analog signals* or *digital signals*.

Analog Signal Any signal that has continuously varying waveforms represented by a sine wave, which has characteristics of amplitude, frequency, and phase. Examples include conventional audio and video.

Analog-to-Digital Conversion The process that converts analog data, which is in the form of a sine wave, into digital signals, which are represented as 0s and 1s. This conversion is necessary so that analog data such as voice communications can be transmitted across a digital network. The most common approach for doing this is a process known as *pulse-code modulation.*

AO/DI See *Always On/Dynamic ISDN*.

Apple Attachment Unit Interface (AAUI) Apple Computer Corporation's proprietary *AUI*.

Appliance See *Network Appliance*.

Application Gateway Firewall See *Proxy Server*.

Application-Level Denial Of Service (ADoS) A DoS attack that is orchestrated via a specific application program such as e-mail.

Application Program Software that performs a specific function such as e-mail.

Application Protocol Defines how an application is to be implemented on a network. Also includes specific user programs for interacting with an application.

Application Service Provider (ASP) An organization that provides its subscribers with designated applications or technologies via the Internet. Also called content service provider.

ARP See *Address Resolution Protocol.*

AS See *Autonomous System.*

ASK See *Amplitude-Shift Keying*.

ASP See *Application Service Provider*.

Asymmetric Cryptography See *Cryptography.*

Asynchronous Communication A data transmission method that requires the sending node to encapsulate special start and stop bits within each unit of data being transmitted. Thus, data can be transferred at any time by the sending node without the receiving node having any advance notification of the transfer.

Asynchronous Digital Subscriber Line (ADSL) A *DSL* variant in which traffic is transmitted at different rates in different directions. Downstream rates range from 1.5 to 9 Mbps; upstream rates range from 16 kbps to 1 Mbps. Rates depend on line quality and local loop distance. Suitable for Internet or intranet access, video-on-demand, database access, and remote LAN access.

Asynchronous Transfer Mode (ATM) A connection-oriented, full-duplex, and point-to-point high-speed cell-switched network architecture that was created in the late 1980s/early 1990s to apply circuit-switching concepts to data networks. Designed to carry data in 53-octet cells, ATM can be used to transmit data, voice, and video—separately or simultaneously—over the same network path. Although not based on any specific physical layer protocol, ATM is generally carried over *SONET.* Also known as *cell relay* to distinguish it from *frame relay.*

ATM See *Asynchronous Transfer Mode*.

ATM Adaptation Layer (AAL) An ATM layer that interprets the type and format of user data messages and then translates these messages into ATM format by packaging them into the 48-byte payload portion of an ATM cell. The AAL's interpretation of data type and format is based on the specific class of service assigned to the data by the application. The AAL provides support for four different service classes and provides five different AAL types to accommodate a particular service class. *AAL1* is used for data that require connection-oriented constant-bit rate transmissions (e.g., voice transmissions); *AAL2* is used for data that require connection-oriented variable-bit rate transmissions (e.g., a videoconferencing application); *AAL3* and *AAL4* are used for connection-oriented or connectionless variable-bit rate transmissions (e.g., bursty data typical of LAN applications such as those found on frame relay and SMDS networks); and AAL5, which is an improvement to AAL3, is used for transmissions in which higher-layer protocols provide error recovery.

Attachment Unit Interface (AUI) A 15-pin "universal" connector that allows a device to be connected to UTP, thick or thin coax, or fiber-optic cable via an external transceiver.

Attenuation The decrease in signal strength, which occurs as the signal travels through a circuit or along a cable. The longer the cable, the greater the attenuation. Also, the higher the frequency of the signal, the greater the attenuation.

AUI See *Attachment Unit Interface.*

Authentication In the context of network security, a systematic method for confirming the identity of an entity. For example, in a secure Web transaction between a client and a server, an authentication procedure is used to establish proof of identity between the two nodes.

Autonomous System (AS) A collection of networks controlled by a single administrative authority that shares a common routing strategy. Routers connecting networks within an AS trust each other and exchange routing information using a mutually agreed upon routing protocol. Also known as a routing domain or protocol area.

Autowrapping The "self-healing" of a token or FDDI ring that has been cut in a single spot. The break in the active ring is corrected by establishing a loopback connection to the inactive ring. This creates a single virtual ring and allows the network to continue to function at full speed.

Backbone Switch One application of an Ethernet switch in which the switch serves as the backbone for the entire LAN. In this application, the network topology is called a "collapsed backbone."

Backward Explicit Congestion Notification (BECN) A one-bit field in a *frame relay* frame that is set to 1 by a frame relay switch to denote that a frame transmitted toward the sending node experienced congestion.

Bandwidth In *analog communications*, the total capacity of a communications channel measured in hertz (Hz). It is the difference between the highest and lowest frequencies capable of being carried over a channel. The greater the bandwidth, the more signals that can be carried over a given frequency range. In *digital communications* and networking, bandwidth is the theoretical capacity of a communications channel expressed in bits per second (bps), which is called *data rate*.

Bandwidth On Demand Interoperability Network Group (BONDING) A protocol that aggregates two *ISDN B* channels into a single 128-Mbps circuit.

Barrel Connector A cylindrical shaped connector used to connect two segments of coaxial cable.

Baseband Cable Uses the entire bandwidth of the cable to carry a single signal.

Basic Rate Interface (BRI) An *ISDN* basic access channel that comprises two 64-kbps *B* channels, one 16-kbps *D* channel, and 48 bits of overhead used for framing and other functions. Commonly written as $2B + D$.

Baud A unit of signaling speed, named after the French engineer Jean Maurice Emile Baudot (1845–1903). It is another term used to express the capacity of a channel, but is different from bits per second.

Baud Rate A measure of the number of times line conditions (i.e., frequency, amplitude, voltage, or phase) change each second. At low speeds (under 300 bps), data rate (measured in bps) and baud rate are the same because signaling methods are relatively simple. As speed increases, signaling methods become more complex. Baud rate then differs from data rate because several bits are typically encoded per baud. That is, each signal can represent more than one bit of information.

***B* Channel** A 64-kbps ISDN clear channel (no signaling information is sent on the channel) used to transmit computer data (text and graphics), digitized voice, and digitized video. Most basic ISDN services are based on multiple *B* channels. Also called a *bearer channel*.

Bearer Channel See *B Channel*.

BECN See *Backward Explicit Congestion Notification*.

Bend Radius The radius in which cable (copper or fiber) can be curved or “bent” without breaking. Fiber is much more flexible than copper cable and can be bent in much smaller radii than equivalent copper.

B-ISDN See *Broadband ISDN*.

Bit-Stuffing A data link process that is used to guarantee the uniqueness of a flag bit so that user data are not interpreted as a given start-stop bit pattern.

Bit-Time A unit of measure equal to 0.1 μs. Thus, a one-bit transmission requires 0.1 μs. Transmitting a 64-byte Ethernet/802.3 frame requires 512 bit-times, or 51.2 μs.

Bluetooth An emerging wireless convergence technology designed to interconnect various devices, including computers, mobile phones, mobile computers, and hand-held or portable terminals using short-range (approximately 10 meters) radio links.

BNC Connector A type of connector used with thin coaxial cable. There are several interpretations of BNC, including bayonet Neill-Concelman (named after its developers), bayonet nut connector, barrel nut connector, and British national connector.

BONDING See *Bandwidth On Demand Interoperability Network Group*.

BRI See *Basic Rate Interface*.

Bridge A layer-2 device that interconnects two or more individual LANs or LAN segments. A transparent bridge is used in Ethernet/802.3 and 802.5 (token ring) networks; a source routing bridge (introduced by IBM) is used exclusively in token ring networks. Bridges keep local traffic local, but forward traffic destined for a remote network. Forwarding/filtering decisions are based on MAC sublayer (i.e., hardware) addresses. Bridges partition Ethernet/802.3 networks into multiple collision domains.

Broadband Cable Shares the bandwidth of a coaxial cable among multiple signals.

Broadband ISDN (B-ISDN) An extension of ISDN that provides full-duplex data transmission at OC-12 rates (622.08 Mbps) and is designed for delivery of interactive services (e.g., videoconferencing and video surveillance) and distribution services (e.g., cable TV and high-definition TV). B-ISDN is also the basis for *ATM*.

Broadcast A data transmission that is destined to all hosts connected to a network. A broadcast message is a special *multicast* message.

Broadcast Design A network configuration that consists of nodes sharing a single communications channel. Every node connected to this shared medium “hears” each other’s transmissions.

Broadcast Storm A network phenomenon that occurs when several broadcast messages are transmitted at the same time. Broadcast storms can use up a substantial amount of network bandwidth and in many cases can cause a network to crash or shut down.

Brouter A combination bridge-router; a bridge with routing capabilities.

Buffering Switch See *Store-and-Forward*.

Bus Design A specific design based on a broadcast topology. All nodes are directly connected to the same communications channel.

Cable See *Wire*.

Cable Modem A modem that uses cable television lines for data communications. These lines use broadband coaxial cable, which has a multitude of frequencies available and significantly higher bandwidth than the UTP cable used by the telcos. Cable modems provide an Ethernet/802.3 network interface that enables a computer to connect to the cable. Once connected, it is as if the PC were connected to an Ethernet/802.3 LAN. The connection is always "up," and multimegabit data rates are possible. Depending on the cable operator and service, current upstream rates for cable modems are somewhere between 500 Kbps to 3 Mbps; downstream rates range from 10 to 30 Mbps.

Capacitance The property of a circuit that permits it to store an electrical charge. The capacitance of a cable determines its ability to carry a signal without distortion. The lower the capacitance, the longer the distance a signal can travel before signal distortion becomes unacceptable.

Carrier Sense Multiple Access (CSMA) A protocol that serves as the basis for various *random access protocols*. CSMA-based protocols include *1-persistent CSMA, nonpersistent CSMA, CSMA with Collision Detection (CSMA/CD),* and *CSMA with Collision Avoidance (CSMA/CA)*.

Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) A variant of *CSMA/CD* except that it specifies an implementation scheme for *collision avoidance* instead of *collision detection*.

Carrier Sense Multiple Access with Collision Detection (CSMA/CD) A variant of either *1-persistent* or *nonpersistent CSMA* that specifies what a node is to do upon detecting a collision. One-persistent CSMA/CD is the MAC sublayer protocol used in Ethernet/802.3 LANs.

Carrier Sense Protocol A network protocol that requires nodes to listen (i.e., "sense") for the "sound" of another node's transmission prior to accessing a shared channel.

Carrier Signal An analog signal that can be modulated with a second data-carrying signal. The carrier does not convey any information until the signal is altered using a specific modulation technique such as *amplitude modulation*, *frequency modulation*, or *phase modulation*. It is the modulation that conveys the information.

CCITT See *Consultative Committee for International Telephony and Telegraphy.*

CDDI See *Copper Distributed Data Interface*.

CDMA See *Code Division Multiple Access*.

CDPD See *Cellular Digital Packet Data*.

Cell A unit of data that is transmitted across a network. Similar to a data *frame*. When used in the context of *ATM*, a cell contains exactly 53-bytes, with 48 bytes for user data and 5 bytes for overhead. Also refers to a geographical area of a *cellular communications network.*

Cells in Frames (CIF) A method of transporting *ATM* protocols over *Ethernet* and *token ring LANs*. CIF is a LAN technology that provides LANs with ATM features, including *QoS* and the seamless integration of data, voice, and video.

Cell Site A specific location within a *cell* in which the base station and supporting antennae of a *cellular communications network* are located.

Cellular Communications Network A network of base stations and antennae that are interconnected to provide wireless voice and data communications.

Cellular Digital Packet Data (CDPD) A data communications strategy that runs over an *AMPS* network. and provides a mechanism for transmitting data over existing voice-based *cellular communications networks*.

Centralized System A single computer that provides all the computing resources for all offices and departments within an organization via computer *terminals* that are connected to the centralized system.

CERT See *Computer Emergency Response Team*.

Channel Service Unit (CSU) A device used for terminating T-carrier circuits. A CSU regenerates the signal, monitors the line for electrical anomalies, provides proper electrical termination, performs framing, and provides remote loopback testing for diagnosing line problems. Usually combined with a *DSU* to form a single unit called a *CSU/DSU* or DSU/CSU.

Channel Service Unit/Data Service Unit (CSU/DSU) A device that combines the functions of a *CSU* and a *DSU*. A CSU/DSU works exclusively with digital signals; it provides an interface between a digital computing device and a digital transmission medium.

Check Bits See *Redundancy Bits*.

Checksum A parameter used to detect errors. Checksums are calculated using a predetermined *generator polynomial* and assigned to a specific checksum field of a data frame.

CIDR See *Classless Interdomain Routing*.

CIF See *Cells in Frames*.

Ciphertext A coded message. See also *Encryption*.

CIR See *Committed Information Rate.*

Circuit Gateway Firewall A device or product that involves monitoring the session setup between a system and the user security options relative to that system for a particular user. For instance, a circuit gateway might check user IDs and passwords, or it might implement proxy connection authorization or other types of authentication services. A circuit firewall is also responsible for logging who came from where and went to what.

Circuit-Switched Network A network design in which a dedicated physical circuit is established between the source and destination nodes before any data transmission can take place. Furthermore, this circuit must remain in place for the duration of a transmission.

Circuit-Switching A switching technique in which a dedicated circuit path is established between two entities on demand. This path is established prior to any data transmissions and is used exclusively by the connected parties until the connection is terminated. Contrast with *packet-switching*.

CIX See *Commercial Internet Exchange*.

Class I Repeater A type of repeater used in *Fast Ethernet LANs*. Class I repeaters support both of Fast Ethernet's signaling schemes—100BASE-T4 and 100BASE-TX/FX.

Class II Repeater A type of repeater used in *Fast Ethernet LANs*. Class II repeaters support only one Fast Ethernet signaling scheme—100BASE-T4 or 100BASE-TX/FX.

Classless Interdomain Routing (CIDR) A routing mechanism that allows sites to advertise multiple *IPv4* Class C networks by using a single prefix.

Class of Service (CoS) A data prioritization scheme that tags data with a specific priority level. Higher-priority data get delivered before lower-priority data.

CLEC See *Competitive Local Exchange Carrier.*

Client A networked device that requests resources from a *server.*

Client/Server A model or paradigm that describes network services and the programs used by end users to access these services. The client side (or front end) provides a user with an interface for requesting services from the network, and the server side (or back end) is responsible for accepting user requests for services and providing these services transparent to the user.

Coaxial Cable A type of cable that consists of a single-wire conductor, surrounded by a dielectric material and two types of shielding—a foil shield and a braided shield—arranged concentrically and encased in a PVC or Teflon outer jacket.

Codec A device that consists of a built-in encoder and decoder. The term itself means "coder/decoder." A codec is used for *analog-to-digital conversions* in which analog data are converted into digital signals for transmission across a digital network. A codec is also used to decode these digital signals back into analog form at the receiving end.

Code Division Multiple Access (CDMA) A digital wireless communications access technology that uses *spread spectrum* to increase system capacity and voice quality of a *cellular communications network.*

Codeword A data frame that comprises both user data and *redundancy bits*, which are used for *error control.*

Collapsed Backbone A network topology in which all LAN segments are interconnected via a bridge or switch, which serves as the network backbone.

Collision What happens when two or more nodes attempt to transmit data simultaneously on an Ethernet/802.3 network.

Collision Domain A "field" within a single Ethernet/802.3 network where two nodes can cause a collision. In the case of a single-segmented Ethernet/802.3 LAN, the independent segment represents the collision domain; in a multisegmented Ethernet/802.3 LAN, the collective segments comprise the collision domain.

Commercial Internet Exchange (CIX) A subscription organization consisting of a consortium of commercial and nonprofit regional network providers that began offering Internet service independent of the NSFNET backbone and without NSF's restriction on traffic type. Today, CIX serves as an Internet interconnect site similar to a *NAP.*

Committed Burst (B_c) In *frame relay*, the maximum amount of data a provider guarantees to deliver within a specified time period, T. $CIR = B_c / T$. Most providers use a 1-second time interval to calculate the average amount of bandwidth utilization. Thus, CIR is usually equal to B_c. The difference between these two parameters is their units. CIR is measured in bps; B_c is measured in bits. See also *Excessive Burst.*

Committed Information Rate (CIR) The amount of throughput a *frame relay* provider guarantees to support under normal network loads. A CIR, which is assigned to a PVC when the network is initially configured, can range from 16 kbps to T3 (44.8 Mbps) and is the minimum guaranteed throughput of a PVC. If a PVC's assigned CIR is greater than or equal to the average amount of traffic transmitted across a PVC over a specified period of time (e.g., 1 second), then data transmissions are guaranteed. If the assigned CIR is less than this average, then data transmissions are not guaranteed.

Competitive Local Exchange Carrier (CLEC) A new telecommunication service provider formed after the Telecommunications Act of 1996 in the United States.

Compression A process that codes repetitive patterns within a data set. Compressed files can be sent at a faster rate than uncompressed files.

Computer Emergency Response Team (CERT) A formal organization operated by the Software Engineering Institute at Carnegie-Mellon University and dedicated to addressing computer and network security issues. CERT also serves as a clearinghouse for identifying and resolving security "holes" in network-related software or operating systems.

Computer Network A collection of computers and other devices that use a common network protocol to share resources with each other over a network medium.

Conductor That part of a wire which serves as the medium for the physical signal. It is composed of either copper wire, glass, or plastic fiber. In the case of copper, the wire can be stranded (composed of several thin wires) or solid (a single "thick" strand). Furthermore, the thickness of a wire is given in terms of gauge, which represents the conductor's diameter. The lower the gauge, the thicker the wire. Most often, wire gauges are expressed in terms of AWG—American Wire Gauge—which is a classification system for copper wire based on a wire's cross-section diameter.

Congestion A situation when a network is consumed with excessive network traffic (i.e., lots of packets), resulting in performance degradation. Congestion occurs when routers are too slow, causing queues to lengthen, or when routers are too fast, causing queues to build up whenever input traffic is greater than the capacity of output lines. The ultimate level of congestion is known as *deadlock*, which occurs when one router cannot proceed until a second router does something, and the second router cannot proceed because it is waiting for the first router to do something. Congestion control is provided by layer 3 of the *OSI model*.

Connectionless Service A type of service in which messages are partitioned into *packets* and routed through the network. Each packet is independent of the other packets that carry parts of the message, and each packet carries a destination address. Unlike connection-oriented service, no physical link is established between sending and receiving nodes prior to data transmission.

Connection-Oriented Service A type of service in which prior to the transfer of data a physical (and virtual) link is established between the sending and receiving nodes. This link remains in effect for the duration of the session. After the session is completed, the link is removed. Characteristics of a connection-oriented service include wasted bandwidth (link must remain established even during idle periods of a transmission), a high potential for a hung network (there is always a possibility that a link will not be terminated), and guaranteed sequential arrival of packets at the destination node.

Connector A layer-1 device that attaches network components together.

Consortia Standards Network standards that are designed and agreed upon by a group of vendors who have formed a consortium for the express purpose of achieving a common goal. These vendors pledge their support for the standards being developed by the consortium and also develop and market products based on these mutually agreed upon sets of standards.

Consultative Committee for International Telephony and Telegraphy (CCITT) An international standards organization, which is now part of *ITU*.

Contention A phenomenon in which more than one node competes to access a shared medium simultaneously.

Contention Protocol A network protocol that specifies the procedures nodes are to follow when competing for access to the same communications channel at the same time. Also called *Random Access Protocol.*

Copper Distributed Data Interface (CDDI) An interface that provides a 100-Mbps data transmission rate over copper. A CDDI network is similar to an FDDI network. CDDI also is restricted to connections between concentrators on the ring and single-attachment devices, not for the ring itself.

CoS See *Class of Service*.

CPE See *Customer Premises Equipment.*

CRC See *Cyclic Redundancy Check.*

CRC Checksum The result of a polynomial division that uses a predetermined *generator polynomial* as the divisor.

CRC Error An invalid *CRC* checksum.

Crosstalk Electrical interference (i.e., *noise*) that occurs when energy radiated from one wire pair of a twisted-pair wire "spills over" into another pair. In one type of crosstalk, called *near-end crosstalk* (NEXT), a signal on the transmit pair is so strong that it radiates to the receive pair. A direct consequence of this spilled-over radiation is that the receiving device cannot decipher the real signal.

Cryptography The practice or art of encoding messages into secret code or cipher. In the context of network security, cryptography can be implemented either symmetrically or asymmetrically. *Symmetric cryptography* implies that a secret (i.e., private) key is used to code and decode messages. In general, n people who want to establish a secret communication among them require $(n)(n-1)/2$ private keys. *Asymmetric cryptography* implies that each person engaged in secret communications maintains a public-private key pair. The public key is published; the private key is kept secret. The successive application of each key is then used either to code or to decode a message.

Cryptology The study of secret communications or speech.

CSMA See *Carrier Sense Multiple Access*.

CSMA/CA See *Carrier Sense Multiple Access with Collision Avoidance.*

CSMA/CD See *Carrier Sense Multiple Access with Collision Detection.*

CSP See *Content Service Provider*.

CSU See *Channel Service Unit.*

CSU/DSU See *Channel Service Unit/Data Service Unit.*

Customer Premises Equipment (CPE) Any telecommunication device that is owned and housed at a customer site.

Cut-Through A network switch architecture in which switches begin forwarding frames from one switch port to another as soon as the frame's destination address is read.

Cyclic Redundancy Check (CRC) An *error detection* method that constructs a polynomial whose terms' coefficients are the values of each of the bits of a data frame. This polynomial is divided by a predetermined *generator polynomial*. The remainder of this division, called the CRC *checksum*, is then assigned to a frame's checksum field. The most common CRC used in most LAN protocols is CRC-32, a 32-bit checksum.

DAM See *Demand Access Multiplexing.*

D-AMPS See *Digital Advanced Mobile Phone System*

DAS See *Dual Attachment Station.*

Data Communications Equipment (DCE) Generally used as a synonym for *modem.* A DCE device is placed between *DTEs* and is responsible for establishing, maintaining, and terminating the link connecting the two DTEs.

Data Encryption Standard (DES) A specific coding technique developed by the National Institute of Standards and Technology (formerly the National Bureau of Standards) and IBM for protecting sensitive data during transmission.

Datagram A grouping of bits organized as a logical unit of data at the network layer. *IP* datagrams serve as the Internet's primary unit of information. In the OSI model, a datagram is generically referred to as a *packet*.

Datagram Packet-Switching See *Packet-Switching*.

Data Link Connection Identifier (DLCI) In *frame relay*, virtual circuit addresses assigned to *PVC*s or *SVC*s.

Data Link Layer The second layer (layer 2) of the OSI model. The data link layer regulates and formats transmission of information from software on a node to the network cabling facilities. This layer is partitioned into two sublayers: the *logical link control* sublayer (LLC), which provides framing, flow control, and error control; and the *media access control* sublayer (MAC), which specifies the manner in which nodes access a shared medium.

Data Rate A measure of the amount of data that can be transferred over a communications medium in a given period. Data rate is measured in bits per second (bps) and can vary considerably from one type of channel to another.

Data Service Unit (DSU) A device used for terminating a T-carrier circuit. A DSU provides the interface (usually V.35, a type of serial interface) for connecting a remote bridge, router, or switch to a T-carrier circuit. The DSU also provides flow control between the network and the *CSU*. DSUs are usually combined with a CSU to form a single unit called a CSU/DSU.

Data Terminal Equipment (DTE) End *devices* that communicate through their serial ports or expansion buses. Computers (PCs, workstations) are examples of DTEs. See also *Data Communications Equipment*.

DB Connector Layer-1 device that serves as an interface between a computer and a peripheral device such as a printer or external modem. DB stands for "data bus."

DCE See *Data Communications Equipment*.

DCE-to-DCE Rate The speed at which two modems "talk" to each other. This rate is fixed and is a function of a modem's speed. Typical rates are 14,400 bps (*V.32*), 28,800 bps (*V.34*), and 57,600 bps (*V.90*).

***D* Channel** A 16-kbps or 64-kbps *ISDN* circuit that is used to carry signal and control information for circuit-switched user data. The *D* channel transmits call initiation (call setup) and termination (call teardown) information between an ISDN device and the telco's central office for each *B* channel. The *D* channel also can be used to transmit packet-switched user data (provided that no signal or control information is needed), data from security alarm signals of remote sensing devices that detect fire or intruders, and low-speed information acquired from telemetry services such as meter reading. The D stands for "delta."

DDoS See *Distributed Denial of Service*.

Deadlock See *Congestion*.

Decentralized System Computer systems that are independent of each other and maintain separate databases germane to specific activities.

Decryption The process of taking an encrypted (coded) message and translating it into its original meaningful form.

De Facto Standards Network standards, placed in the public domain, that have been met with widespread industry acceptance instead of formal approval from a standards organizations. (De facto is Latin for "from the fact.")

De Jure Standards Network standards approved by a formal accredited standards organization such as ANSI or ITU. (De jure is Latin for "by right, according to law.")

Demand Access Multiplexing (DAM) A multiplexing technique in which a pool of frequencies is managed by a "traffic coordinator." Pairs of communications frequencies are assigned to a requesting station—one pair for transmission, a second pair for reception ("demand"). These two pairs of frequencies are connected to another set of frequencies ("access"). When one or both stations are finished communicating, the allocated frequencies are deallocated and returned to the frequency pool, where they are made available for other incoming requests ("multiplexing").

Demand Priority A *MAC* sublayer protocol used in *100VG-AnyLAN* networks. Demand priority specifies the manner in which repeater hubs poll their ports to identify which nodes have data to transmit and the order of these transmissions.

Denial of Service (DoS) A type of network attack that denies users access to the services of the system that was attacked. A DoS attack does not actually compromise or damage a host. It simply shuts down the host or service by blocking out legitimate traffic.

DES See *Data Encryption Standard*.

Desktop Another name for a networked device. See also *Workstation*.

Device Any entity that is connected to a network. Examples include terminals, printers, computers, or special network-related hardware units such as communication servers, repeaters, bridges, switches, and routers. Local or sending devices originate communications; remote or receiving devices are the recipients of such communications.

Differential Manchester Encoding A data transmission encoding scheme similar to *Manchester encoding*—each bit-period is partitioned into two intervals, and a midbit transition between "high" and "low" occurs during each bit-period. In differential Manchester coding, though, the interpretation of these low-to-high and high-to-low midbit transitions is a function of the previous bit-period. The presence of a transition at the beginning of a bit-period is coded 0, and the absence of a transition at the beginning of a bit-period is coded 1.

Diffused IR A "broadcast" infrared transmission method in which a transmitter "floods" a specific area with a strong infrared signal that is spread over a wide angle. The IR signal is transmitted by reflecting off ceilings, walls, and other surfaces.

Digital Any device or signal that varies discretely in strength or quantity between two values, usually 0 and 1: 0 implies "off"; 1 implies "on." Digital signals are represented as binary digits called "bits" and are *discrete*.

Digital Advanced Mobile Phone System (D-AMPS) The digital version of *AMPS*. It uses the same 30 kHz channels as AMPS but employs *TDMA* for its access technology instead of *FDMA*.

Digital Certificate An electronic passport that consists of a numerical pattern, value, or key and is used for personal identification. Creating a digital certificate involves a user identifying a specific personal trait to a trusted third party, which issues the certificate.

Digital Communication Any type of communication in which data are represented in the form of binary digits.

Digital Data Any data that have been converted into a binary code such as ASCII.

Digital Signal An electronic signal that is represented using 0s and 1s instead of as a continuum of voltages, as is the case with *analog signals*. Analog data are usually converted to digital signals via *pulse-code modulation*; digital data are converted to digital signals using any one of several techniques, including *Manchester encoding*, *differential Manchester encoding*, and *NRZI*.

Digital Signature A security authorization method in which a user "signs" a document so that the document's authenticity can be confirmed by checking the signature. A digital signature proves a message was not modified.

Digital Subscriber Line (DSL) A technology that enables data, voice, and video to be mixed and carried over standard analog (copper) telephone lines. This is accomplished by using the unused frequencies that are available on a telephone line. Thus, DSL can deliver data services without interfering with voice transmissions. There are at least nine DSL variants: *ADSL, ADSL Lite, HDSL, HDSL 2, IDSL, RADSL, SDSL, UDSL,* and *VDSL*.

Digital Subscriber Loop The formal term for the *local loop*, which is the circuit between a *customer's premises equipment (CPE)* and the telco's equipment.

Digital-to-Analog Conversion A process used to convert digital data such as computer output into analog signals for transmission across an analog circuit path. Common strategies for effecting this conversion include *amplitude-shift keying* (*ASK*), *frequency-shift keying* (*FSK*), *phase-shift keying* (*PSK*), and *quadrature amplitude modulation* (*QAM*).

Digital-to-Digital Conversion A process used to convert digital data into digital signals for transmission over a suitable facility. Common strategies include *Manchester encoding*, *differential Manchester encoding*, and *NRZI*.

DIN Connector Similar to a *DB connector*, but circular instead of rectangular and typically used to connect a keyboard to a computer. DIN stands for "Deutsche Industrie Norm," a German industrial standard.

Directed IR A "point-to-point" infrared transmission method that requires an unobstructed line-of-sight connection between transmitter and receiver. It is basically a "point and beam" medium.

Direct Sequence Spread Spectrum (DSSS) A physical layer technology used in wireless LANs (IEEE 802.11). DSSS operates by spreading a signal over a wide range of the 2.4-GHz band.

Discard Eligibility A field in a *frame relay* frame, which, if set to 1 by an end node, denotes that the frame can be discarded in the presence of congestion. Discarded frames will then be retransmitted at a later time when congestion has subsided.

Distance-Vector Algorithm A routing algorithm that determines the distance between source and destination nodes by calculating the number of router hops a packet traverses en route from the source network to the destination network. An example of a distance-vector algorithm is the Bellman-Ford algorithm.

Distributed Denial of Service (DDoS) A highly malevolent form of a *DoS* attack that involves placing hundreds or thousands of small programs on different computers connected to a network. These programs, called agents or "zombies," are controlled by a master console somewhere on the network. When commanded to do so, the zombies go into attack mode and begin sending messages to the specified destination, completely overwhelming the destination system as well as the network pathways to get to the system.

Distributed Queue Dual Bus (DQDB) A data link layer protocol (IEEE 802.6) that specifies the medium access method for *MANs*. Used in *SMDS*.

Distributed System Computers that are linked together to provide, in a transparent manner, the required computing resources and information-processing needs of an entire organization. Distributed systems bear the greatest resemblance to computer networks.

DLCI See *Data Link Connection Identifier.*

DNS See *Domain Name Service.*

Domain Name A logical name assigned to an *IP address* and used as another type of addressing construct for identifying Internet nodes. The translation between logical name and IP address is called name resolution, which is provided by a *domain name service*.

Domain Name Service (DNS) An Internet translation service that resolves *domain names* to *IP addresses*, and vice versa. Domain name service is provided by DNS servers.

DoS See *Denial of Service*.

DQDB See *Distributed Queue Dual Bus*.

DS-0 A single digital voice channel rated at 64 kbps. The notation *DS-0* stands for "digital signal at level 0," which refers to a voice channel multiplexed into a digital signal.

DS-1 A digital signal that carries 24 *DS-0* channels plus one 8-kbps channel reserved for framing for an aggregate bandwidth of 1.544 Mbps. A *T1* circuit carries a DS-1 signal.

DS-2 A digital signal that carries 4 *DS-1* channels for an aggregate bandwidth of 6.312 Mbps. A *T2* circuit carries a DS-2 signal.

DS-3 A digital signal that carries 28 *DS-1* channels for an aggregate bandwidth of 44.736 Mbps. A *T3* circuit carries a DS-3 signal.

DS-4 A digital signal that carries 168 *DS-1* channels for an aggregate bandwidth of 274.176 Mbps. A *T4* circuit carries a DS-4 signal.

DSL See *Digital Subscriber Line*.

DSL Access Multiplexer (DSLAM) A device that aggregates *DSL* signals so they can be transferred directly into a data switch for transmission across the telco's data network backbone.

DSLAM See *DSL Access Multiplexer.*

DSSS See *Direct Sequence Spread Spectrum*.

DSU See *Data Service Unit*.

DTE See *Data Terminal Equipment*.

DTE-to-DCE Rate The speed at which a computer "talks" to its modem. Typical rates include a 4:1 compression ratio between DTE and DCE speeds. Thus, for a *V.34* modem (28,800 bps), the DTE-DCE rate is 115,200 bps. This rate is user configurable.

Dual-Attachment Station (DAS) An FDDI node that is connected to two full dual-fiber rings and has the ability to reconfigure the network to form a valid network from components of the two rings in case of a failure. A DAS is also called a *Class A* node.

E-1 The multiplexing of 30 separate 64-kbps voice channels, plus one 64-kbps control channel, into a single wideband digital signal rated at 2.048 Mbps. E-1 is the basic telecommunications service used in Europe.

E-2 A multiplexed circuit that combines 4 *E-1* circuits and has an aggregate bandwidth of 8.448 Mbps.

E-3 A multiplexed circuit that combines 16 *E-1* circuits and has an aggregate bandwidth of 34.368 Mbps.

E-4 A multiplexed circuit that combines 64 *E-1* circuits and has an aggregate bandwidth of 139.264 Mbps.

E-5 A multiplexed circuit that combines 256 *E-1* circuits and has an aggregate bandwidth of 565.148 Mbps.

E.164 An ITU-T standard network addressing format that resembles telephone numbers. E.164 addresses are 15 decimal digits long and include a country code, an area or city code, and a local number. Country codes are two or three digits long and consist of a zone code followed by a one- or two-digit national identifier. Area or city codes are up to four digits long. If an address contains fewer than 15 digits, then it is padded with hexadecimal Fs. The United States and Canada use the zone code 1 followed by a three-digit area code and a seven-digit local number in lieu of country codes.

E-commerce Short for "electronic commerce," which involves using the Internet for credit card purchases of items such as automobiles, airline tickets, computer hardware and software, and books.

EGP See *Exterior Gateway Protocol*.

EIGRP See *Enhanced IGRP*.

Encapsulation A process in which a *packet* or *frame* is enclosed or "wrapped" in a specific protocol header. For example, *routers* typically perform protocol encapsulation in which packets from one network protocol are wrapped into the header of another network protocol so the packet can be transmitted to a different network. Also called *tunneling*.

Encryption The process of coding a message so that it is incomprehensible to unauthorized users. When retrieved by authorized users, encrypted messages are then reconverted (i.e., decoded) into meaningful text. Encrypted output is called *ciphertext*.

Enhanced IGRP (EIGRP) A routing protocol designed by Cisco that combines the best features of distance-vector and link-state routing protocols.

Enum An IETF protocol that maps *E.164* telephone numbers to a *uniform resource identifier (URI)*, which enables users to use telephone numbers to access *Internet* services. Enum-enabled applications demonstrate the convergence of the *public switched telephone network (PSTN)* and the Internet.

Error Control The process of guaranteeing reliable delivery of data. Error control can be provided through *error detection* or *error correction*.

Error Correction The process in which a destination node, upon detecting a data transmission error, has sufficient information to correct the error autonomously. Error correction implies *error detection*.

Error Detection The process in which a destination node detects a data transmission error and requests a retransmission from the sending node. Error detection is also called error correction through retransmission.

Ethernet A local area network protocol developed jointly by Xerox, Intel, and Digital Equipment Corporation (DEC) at the Xerox Palo Alto Research Center (PARC) in the mid-1970s. The name "Ethernet" was derived from the old electromagnetic theoretical substance called luminiferous ether, which was formerly believed to be the invisible universal element that bound together the entire universe and all its associated parts. Thus, an "ether" net is a network that connects all components attached to the "net."

Excessive Burst (B_e) In *frame relay*, the maximum amount of uncommitted data a provider will attempt to deliver within a specified time period. A provider will guarantee a *committed burst* of B_c bits and will attempt to deliver (but not guarantee) a maximum of $B_c + B_e$ bits.

Exchange Access SMDS (XA-SMDS) A special *SMDS* service through which *local exchange carriers* offer SMDS to *interexchange carriers* for delivery across *LATAs*.

Exterior Gateway Protocol (EGP) Any Internet interdomain routing protocol used to exchange routing information with other autonomous systems. Also refers to a specific EGP defined in RFC 904. Another EGP is the Border Gateway Protocol (BGP), defined in RFC 1105 and RFC 1771. Both EGP and BGP are part of the *TCP/IP* protocol suite. Of the two, however, BGP has evolved into a robust Internet routing protocol and the term "border gateway protocol" is used in favor of the term "exterior gateway protocol."

Extranet An interconnection from an internal intranet to a customer or noncompany network that is not the Internet connection.

4B/5B A data encoding method, which stands for *four bits in five baud*, or *four-bit to five-bit*, used in FDDI networks.

5-4-3 Repeater Placement Rule A general rule of thumb to follow when configuring an Ethernet/802.3 LAN to ensure that it follows IEEE specifications. The 5-4-3 repeater placement rule requires no more than five segments of up to 500 m each, no more than four repeaters, and no more than three segments with end nodes connected to them. This rule is also known as the 4-repeater rule or the 5-4-3-2-1 rule. In the latter, the 2 implies that two of the five segments are used as interrepeater links, and the 1 implies that a configuration using the maximum parameters permitted results into one collision domain.

568SC Connector See *SC Connector.*

Fast Ethernet 100-Mbps *Ethernet* (*IEEE 802.3u*). Three different media specifications are defined: 100BASE-TX, 100BASE-T4, and 100BASE-FX.

FDDI See *Fiber Distributed Data Interface*.

FDDI-II A now-defunct second-generation *FDDI* technology that was intended to handle traditional FDDI network traffic as well as synchronous, circuit-switched *PCM* data for voice or *ISDN* systems.

FECN See *Forward Explicit Congestion Notification*.

Federal Internet Exchange (FIX) An Internet interconnect site similar to a *NAP*.

FDMA See *Frequency Division Multiple Access*.

FHSS See *Frequency Hopping Spread Spectrum*.

Fiber Distributed Data Interface (FDDI) An ANSI standard, X3T9.5, created in 1986 for interconnecting computer systems and network devices typically via a fiber ring topology at 100 Mbps.

Fiber-Optic Cable A type of cable that carries data signals in the form of modulated light beams. The cable's conductor can be either glass or plastic. Fiber-optic cable is immune to electromagnetic interference (EMI) and other types of externally induced noise, including lightning; it is unaffected by most physical factors such as vibration; its size is smaller and its weight is lighter than copper; it has much lower attenuation per unit of length than copper; and it can support very high bandwidth. Two general types are available: *single-mode fiber* and *multimode fiber*.

Fibre Channel A family of ANSI standards that defines a specific communications interface for high-speed data transfers between different hardware systems. Applications include the medical profession, where large images (e.g., 100-MB+ x-rays) are transferred from a scanner to a computer to a screen, and the electronic publishing industry, where large files are transferred from a designer/creator's machine to a publisher's computer. It has also become the "backbone" of high-speed data storage systems.

File Transfer Protocol (FTP) A TCP/IP application protocol used for transferring files between two systems.

Firewall A device or product that allows systems or network managers to restrict access to components on a network. Five generally accepted types of firewalls are used on Internet connections: *frame-filtering, packet-filtering, circuit gateways, stateful* and *application gateways,* and *proxy servers*.

FIX See *Federal Internet Exchange*.

Flow Control A process that controls the rate at which data messages are exchanged between two nodes. Flow control provides a mechanism to ensure that a sending node does not overwhelm a receiving node during data transmission.

FM See *Frequency Modulation*.

Forward Explicit Congestion Notification (FECN) A one-bit field in a *frame relay* frame that is set to 1 by a frame relay switch to denote that a frame transmitted toward the receiving node experienced congestion.

Fractional T1 *T1* service that is sold in 64-kbps increments.

FRAD See *Frame Relay Access Device*.

Fragmenting A process in which a *packet* is broken into smaller units to accommodate the maximum transmission unit a physical network is capable of supporting. Fragmented packets are sent to the destination separately and then reassembled at the destination node before they are passed to the higher levels. In *IP*, reassembly of a *datagram* occurs at the destination node and not at any of the intermediary nodes the packet traverses.

Frame A formatted sequence of bits that incorporates both data and control information.

Frame-Filtering Firewall A *firewall* device or product that filters (permits or denies access) at the *data link layer* by examining frames for both layout and content.

Frame Relay A public *WAN* packet-switching protocol that provides LAN-to-LAN connectivity. Its name implies what it does, namely, relays frames across a network between two sites. Frame relay was originally part of the *ISDN* standard.

Frame Relay Access Device (FRAD) Any *frame relay* end node.

Framing A *data link layer* process that partitions a bit stream into discrete units or blocks of data called *frames*.

Frequency The number of times an electromagnetic signal repeats itself (i.e., the identical signal is continuously generated) within a specific time period. A frequency rate of one cycle per second is defined as 1 *hertz* (Hz).

Frequency Division Multiplexing (FDM) A multiplexing technique that partitions the available transmission frequency range into narrower bands (subfrequencies), each of which is a separate channel. FDM-based transmissions are parallel in nature.

Frequency Division Multiple Access (FDMA) A *cellular communications network* access technology similar to *frequency division multiplexing*. The assigned frequency range within a *cell* is partitioned into narrower, multiple frequency bands, each of which constitutes a separate communications channel.

Frequency Hopping Spread Spectrum (FHSS) A physical layer technology used in wireless LANs (*IEEE 802.11*). FHSS operates by transmitting short bursts of data on different frequencies. One burst is transmitted on one frequency, a second burst is transmitted on a second and different frequency, and so forth.

Frequency Modulation (FM) A *modulation* technique in which the *frequency* of an electromagnetic wave is altered so that the signal can carry more information.

Frequency Reuse A concept in which the same radio frequencies are used in different regions to maximize the use of limited frequency bands.

Frequency-Shift Keying (FSK) Similar to *frequency modulation* except frequency-shift keying is the more appropriate term when used in the context of converting digital data (or signals) into analog signals at the sending end and then back to digital form at the receiving end. The frequency of a signal is altered so that it conforms to digital data (0s and 1s); one frequency is used to represent a binary 0 and a second frequency is used to represent a binary 1.

FSK See *Frequency-Shift Keying*.

FTP See *File Transfer Protocol*.

Full-Duplex Ethernet Ethernet/802.3 LANs that are switched-based and hence collision-free. In full-duplex Ethernet, both sending and receiving channels can operate simultaneously since the receiving channel does not have to listen for collisions.

Full-Duplex Transmission A data transmission method that involves the simultaneous sending and receiving of data in both directions.

GAN See *Global Area Network*.

Gateway A software application that converts between different application protocols. The host on which this software resides is called a *gateway machine*. Historically, this term also refers to a *router* in the *IP* community.

General Packet Radio Service (GPRS) A nonvoice packet switching technology that makes *Internet* applications accessible via *GSM* or *IS-136 cellular communications networks*. Peak data transfer rates are 170 kbps. Unlike current cellular services such as *short message services*, a GPRS-enabled phone enables users to access any Internet application or site that has an IP address.

Generation A term used to specify the stages of a technology's evolution. For example, first-generation (1G) refers to the initial development of a technology, and second-generation (2G) refers to the first major enhancement to this technology.

Geostationary Earth Orbit (GEO) Satellite A satellite placed into orbit at an altitude of 22,000 miles (36,000 km) above the equator. GEO satellites traverse their orbits at approximately the same rate as the earth rotates. Thus, the satellite appears stationary with respect to the earth's rotation. Also call *geosynchronous earth orbit*. Only eight GEO satellites are needed to provide global communications coverage.

Gigabit Ethernet 1000-Mbps Ethernet (*IEEE 802.3z*).

G.Lite See *ADSL Lite*.

Global Area Network (GAN) A collection of *WANs* that spans the globe.

Global System for Mobile (GSM) Communications A de facto cellular network standard that was developed to provide cellular compatibility throughout Europe. It has since become the international standard of choice and is now deployed in over 100 countries, including parts of the United States. Versions of GSM are either *TDMA*-based or *CDMA*-based. GSM operates in the 800–900 MHz range as well as in the 1800–1900 MHz range. It is also compatible with *ISDN* and is the basis for other *cellular communications network* standards, including the personal communications services network.

GOSIP See *Government OSI Profile*.

Government OSI Profile (GOSIP) A U.S. government directive that mandated all government organizations to purchase *OSI*-compliant networking products beginning in 1992. In 1995, however, GOSIP was modified to include *TCP/IP* as an acceptable protocol suite for GOSIP compliance.

GPRS See *General Packet Radio Service*.

Graded-Index Multimode Fiber A type of multimode fiber in which variations in the density of the core medium change its index of refraction such that light is refracted (i.e., bends) toward the center of the fiber.

GSM See *Global System for Mobile* (GSM) *Communications*.

H.323 An ITU standard for packet-based *multimedia networking*.

Half-Duplex Transmission A data transmission method in which data may travel in either direction—from sender to receiver or from receiver to sender—but only one unit can send at any one time. While one node is in send mode, the other is in receive mode.

Handoff The process used in *cellular communications networks* that enables mobile devices to maintain a constant connection as they roam from one *cell* to another.

Harmonic Motion The basic model for vibratory or oscillatory motion. Examples include mechanical oscillators such as mass-spring systems and pendulums; periodic motion found in the earth sciences such as water waves, tides, and climatic cycles; and electromagnetic waves such as alternating electric currents, sound waves, light waves, radio waves, and television waves.

***H* Channel** An *ISDN* channel used for transmitting user data (not signal or control information) at higher transmission rates than a *B* channel provides. Four *H* channels are defined: *H0* (six *B* channels; 384 kbps); *H10* (United States specific; aggregates 23 *B* channels; 1.472 Mbps); *H11* (equivalent of North American DS-1; 24 *B* channels; 1.536 Mbps); and *H12* (European specific; comprises 30 *B* channels; 1.920 Mbps).

HDSL See *High bit-rate Digital Subscriber Line*.

HDSL 2 A modified *HDSL* designed and packaged for corporate clients.

Hertz A measure of frequency in cycles per second. A frequency rate of one cycle per second is defined as 1 hertz (Hz). Named in honor of Heinrich Rudolph Hertz (1857–1894), a German physicist who in the late 1880s was the first to produce radio waves artificially.

HFC See *Hybrid Fiber Cable*.

High bit-rate Digital Subscriber Line (HDSL) A *DSL* variant that provides symmetrical service at *T1* rates over two pairs of *UTP* and *E1* rates over three pairs of UTP. Telephone service is not supported. Applications include connecting PBXs and serving as an alternative to T1/E1. HDSL is suitable for campus networks and *ISPs*.

Hold-down A strategy used by *RIP* that requires routers not to update their routing tables with any new information they receive for a prescribed period of time, called the hold-down time. Designed to prevent routing loops. Hold-down is not standardized.

Hop The passage of a *packet* through an intermediate gateway (*router*) en route to another network. For example, if a packet transverses through two routers in reaching its final destination, then we say the destination is two hops away.

Host A networked computer system (see *Workstation*). Also describes a computer system that provides service to users (see *Server*).

HTTP See *Hypertext Transfer Protocol*.

Hub Generically, any *device* that connects two or more network segments or supports several different media. Examples include repeaters, switches, and concentrators.

Hybrid Fiber Cable (HFC) A cable TV system that has *fiber-optic cable* between the head end and neighborhood distribution sites, but coaxial cable between the neighborhood distribution and residential homes and businesses.

Hybrid Switching A data transmission method that combines the principles of circuit- and *packet-switching*. This technique first partitions a message into packets (packet-switching) and transmits each packet via a dedicated circuit (*circuit-switching*). As soon as a packet is ready for transmission, a circuit meeting appropriate bandwidth requirements is established between the sending and receiving nodes. When the packet reaches its destination, the circuit is broken down so that it can be used again.

Hypertext Transfer Protocol (HTTP) A TCP/IP application protocol on which the World Wide Web (WWW) is based. It is a request-response protocol in which an HTTP client program establishes a TCP connection to an HTTP server program and requests specific services from the server. Request messages are made through a *user agent* such as a Web browser, and response messages are provided by the server after it has received and interpreted the request message.

IAB See *Internet Architecture Board.*

IANA See *Internet Assigned Numbers Authority.*

IBM Cable System (ICS) A copper wire classification system established by IBM that specifies nine cable "types" (1 through 9). Of the nine types defined, specifications are available for only seven; types 4 and 7 have no specifications.

ICANN See *Internet Corporation for Assigned Names and Numbers.*

ICMP See *Internet Control Message Protocol*.

ICS See *IBM Cable System.*

IDSL See *ISDN-like Digital Subscriber Line*.

IEC See *Interexchange Carrier.*

IEEE See *Institute of Electrical and Electronics Engineers*.

IEEE 802 The primary *IEEE* standard for the 802.*x* series for *LANs* and *MANs*.

IEEE 802.1 *IEEE* standard that defines an architectural overview of *LANs*.

IEEE 802.2 *IEEE* standard that defines the logical link control, which describes services for the transmission of data between two nodes.

IEEE 802.3 *IEEE* standard that defines the *carrier sense multiple access/collision detection (CSMA/CD)* access method commonly referred to as *Ethernet*. Supplements include 802.3c (10-Mbps Ethernet); 802.3u (100-Mbps Ethernet, known as *Fast Ethernet*); 802.3z and 802.3ab (1000-Mbps Ethernet, known as *Gigabit Ethernet*); and 802.3ae (10,000-Mbps Ethernet, known as *10 Gbps Ethernet*).

IEEE 802.4 *IEEE* standard that defines the token bus network access method.

IEEE 802.5 *IEEE* standard that defines the logical ring *LAN* that uses a token passing access method; known also as *token ring*.

IEEE 802.6 *IEEE* standard that defines metropolitan area networks (*MANs*).

IEEE 802.7 *IEEE* standard that defines broadband *LANs* (capable of delivering video, data, and voice traffic).

IEEE 802.9 *IEEE* standard that defines integrated digital and video networking—integrated services LANs (ISLANs).

IEEE 802.10 *IEEE* standard that defines standards for interoperable *LAN/MAN* security services.

IEEE 802.11 *IEEE* standard that defines standards for wireless media access control and physical layer specifications.

IEEE 802.12 *IEEE* standard that defines the "demand priority" access method for 100-Mbps LANs; known also as 100 Base-VG or *100VG-AnyLAN*.

IEEE 802.13 Defines nothing—IEEE was concerned about the superstitious overtones associated with 13.

IEEE 802.14 *IEEE* standard that defines a standard for cable TV-based broadband communication.

IEEE 802.16 *IEEE* standard that defines the specifications for the *wireless local loop*.

IETF See *Internet Engineering Task Force.*

IGP See *Interior Gateway Protocol*.

IGRP See *Interior Gateway Routing Protocol*.

ILEC See *Incumbent Local Exchange Carrier.*

I-Mode A Japanese-based wireless Internet initiative that provides *cellular communications network* subscribers to access via their handsets a host of personal communications and personal entertainment services, including traditional cellular voice communications, pages, Internet access, and color television and broadband wireless video services equivalent to cable television. The *i* in i-mode stands for "information."

Impedance A measure of the opposition to the flow of electric current in an alternating current circuit. Measured in ohms (abbreviated by the Greek symbol omega, Ω), impedance is a function of capacitance, resistance, and inductance. Impedance mismatches, caused by mixing cables of different types with different characteristic impedances, can result in signal distortion.

Impulse Noise Electrical *noise* that consists of intermittent, undesirable signals induced by external sources such as lightning, switching equipment, and heavy electrically operated machinery such as elevator motors and photocopy machines. Impulse noise increases or decreases a circuit's signal level, which causes the receiving equipment to misinterpret the signal.

Incumbent Local Exchange Carrier (ILEC) The contemporary name given to the seven *RBOCs* and GTE relative to the United States Telecommunications Act of 1996. With mergers, only four remain: Ameritech, SBC, Verizon Communications, and GTE.

Infrared (IR) A line-of-sight transmission method that uses electromagnetic radiation of wavelengths between radio waves and visible light, operating between 100 GHz and 100 THz (terahertz). IR transmission can occur in one of two ways: *directed* and *diffused*.

Institute of Electrical and Electronics Engineers (IEEE) A professional society of engineers, scientists, and students. One of its many activities is to act as a coordinating body for computing and communication standards.

Insulation Material surrounding the *conductor* of a wire. The insulation serves as a protective "barrier" to the conductor by preventing the signal from "escaping" and preventing electrical interference from "entering."

Integrated Services Digital Network (ISDN) A carrier service that is offered by telephone companies (telcos) and designed to transmit voice and nonvoice (e.g., computer data, fax, video) communications on the same network.

Interexchange Carrier (IEC) Any company that provides long-distance telephone and telecommunications services such as AT&T, Sprint, British Telecom (BT), and MCI WorldCom.

Interior Gateway Protocol (IGP) Any intradomain Internet protocol used to exchange routing information within an *autonomous system*. Examples include *RIP, RIP-2, OSPF, IGRP,* and *Enhanced IGRP (EIGRP)*.

Interior Gateway Routing Protocol (IGRP) A routing protocol developed by Cisco to address some of the problems associated with routing in large heterogeneous networks.

Intermediate System to Intermediate System (IS-IS) An intradomain routing protocol designed by *OSI* to run within an *AS* (called a "routing domain" in the OSI world). IS-IS uses a *link-state routing algorithm* to calculate least-cost paths and is similar in operation to *OSPF*. The formal title of this protocol is Intermediate System to Intermediate System Intra-Domain Routing Exchange Protocol.

Intermodulation Noise Electrical *noise* that occurs when two frequencies interact to produce a phantom signal at a different frequency. Occurs in *frequency-division multiplexed* channels.

International Organization for Standardization (ISO) An international organization that develops and promotes networking standards worldwide.

International Telecommunications Union (ITU) A global standards organization. ITU is the former *CCITT*.

Internet When used as a noun and spelled with a lowercase *i*, "internet" is an abbreviation for *internetwork*, which refers to a collection of interconnected networks that functions as a single network. When used as a proper noun and spelled with an uppercase *I*, "Internet" refers to the world's largest internetwork, which consists of hundreds of thousands of interconnected networks worldwide and based on a specific set of network standards (*TCP/IP*).

Internet Architecture Board (IAB) An organization that is part of the *Internet Society* responsible for the overall planning and designing of the Internet. Responsibilities include setting Internet standards, managing the publication of RFC documents, and resolving technical issues. Assigned to the IAB are the *Internet Engineering Task Force* and the *Internet Research Task Force*. Formerly known as the Internet Activities Board.

Internet Assigned Numbers Authority (IANA) An organization that has authority over all number spaces used in the Internet including *IP addresses*. IANA control will soon be transferred to the *Internet Corporation for Assigned Names and Numbers (ICANN)*.

Internet Control Message Protocol (ICMP) An *IP datagram* that carries messages about the communications environment of the *Internet*.

Internet Corporation for Assigned Names and Numbers (ICANN) A private, nonprofit corporation with international representation expressly formed to assume the responsibilities currently being performed by IANA and other government organizations that provide domain name service.

Internet Engineering Task Force (IETF) An organization that is part of the *Internet Architecture Board* and primarily concerned with addressing short- or medium-term Internet engineering issues. Relies on the Internet Engineering Steering Group (IESG) to prioritize and coordinate activities.

Internet Protocol (IP) A layer-connectionless protocol. IP receives data bits from the lower layer, assembles these bits into packets, called IP *datagrams*, and selects the "best" route based on some metric to route the packets between nodes. This is the IP of *TCP/IP*.

Internet Registry (IR) A formal hierarchical system used for assigning *IP addresses*. From top to bottom, this hierarchy consists of *IANA*, regional Internet registries (RIRs), and local Internet registries (LIRs), and works as follows: IANA allocates blocks of IP address space to RIRs; RIRs allocate blocks of IP address space to their LIRs; LIRs then assign addresses to either end users or ISPs.

Internet Research Task Force (IRTF) An organization that is part of the *Internet Architecture Board* and primarily concerned with addressing long-term research projects. Relies on the Internet Research Steering Group (IRSG) to prioritize and coordinate activities.

Internet Service Provider (ISP) A company that provides its customers with access to the Internet.

Internet Society (ISOC) An international organization comprised of volunteers who promote the Internet as a medium for global communication and collaboration. ISOC is considered the ultimate authoritative organization of the Internet.

Internet2 A collaborative project of the University Corporation for Advanced Internet Development (UCAID), which comprises over 100 U.S. universities, government organizations, and private sector firms. Internet2's mission is to develop advanced Internet technologies and applications that support the research endeavors of colleges and universities. Internet2 members use the *vBNS* to test and advance their research.

Internetwork A collection of interconnected networks that function as a single network. The individual networks comprising an internetwork are called *subnetworks*.

Interoperability The degree in which products (software and hardware) developed by different vendors are able to communicate successfully (i.e., interoperate) with each other over a network.

Intranet An internal network implementation of traditional Internet applications within a company or an institution.

Inverse Multiplexing The reverse of *multiplexing*. Instead of partitioning a single communication medium into several channels, an inverse *multiplexer* combines several "smaller" channels (i.e., low-speed circuits) into a single high-speed circuit. This technique is also sometimes generically called *line aggregation*.

IP See *Internet Protocol*.

IP Address A network address assigned to a node's network interface and used to uniquely identify (locate) the node within the Internet. Two versions are currently implemented: *IPv4* and *IPv6*.

IPSec See *IP Security.*

IP Security (IPSec) A suite of network security protocols that operates at layer 3 and provides address authentication, data encryption, and automated key exchanges between sender and receiver nodes.

IP Telephony The integration voice and data in which digitized voice signals are converted to *IP* packets and then transmitted over the *Internet*.

IPv4 An acronym for *Internet protocol version 4*.

IPv4 Address An *IP address* based on *IPv4*. These addresses consist of 32 bits (0 through 31) partitioned into four groups of eight bits each (called *octets*) and organized into five classes (A through E) based on the values of bits 0 through 3.

IPv6 An acronym for *Internet protocol version 6*, which is an evolutionary replacement to IPv4. IPv6 maintains most IPv4 functions, relegates certain functions that either were not working or were rarely used in IPv4 as optional, and adds new functionality that is missing from IPv4. Sometimes called IPng (for next generation).

IPv6 Address An *IP address* based on *IPv6*. An IPv6 address consists of 128 bits and is 4 billion $\times$ 4 billion times the size of the IPv4 address space (2^{96} vs. 2^{32}). Unlike IPv4 addresses, IPv6 addresses use a colon as their delimiter (instead of a "dot" notation), and they are written as eight 16-bit integers expressed in hexadecimal form.

IR See *Infrared* or *Internet Registry.*

IRTF See *Internet Research Task Force.*

IS-95 Any *CDMA*-based *cellular communications network* that operates at 800 MHz. IS-95A provides 14.4 kbps data transfer rates and IS-95B, which is referred to as a 2.5 generation technology, increases data transfer rates to 64 kbps. IS-95 is sometimes called cdmaOne.

IS-136 Any *TDMA*-based *cellular communications network* that operates at either 800 MHz or 1900 MHz. Two examples include *D-AMPS* and *GMS*.

ISDN See *Integrated Services Digital Network.*

ISDN-like Digital Subscriber Line (IDSL) A *DSL* variant that provides symmetrical service at a maximum of 144 kbps each way. Uses *ISDN* hardware.

IS-IS See *Intermediate System to Intermediate System.*

ISO See *International Organization for Standardization*.

ISOC See *Internet Society.*

Isochronous Communications The delivery of time-sensitive data such as voice or video transmissions. Networks that are capable of delivering isochronous service (e.g., *ATM*) preallocate a specific amount of bandwidth over regular intervals to ensure that the transmission is not interrupted.

IsoEthernet Short for *Isochronous Ethernet*, an IEEE standard (*IEEE 802.9a*) that is designed to support time-sensitive applications such as videoconferencing and telephony. IsoEthernet runs both conventional 10-Mbps *Ethernet* and *ISDN B channels* over the same network. The Ethernet channel is used for normal data networking needs; the ISDN *B* channels are used for time-sensitive applications.

ISP See *Internet Service Provider.*

ITU See *International Telecommunications Union*.

IXC See *Interexchange Carrier.*

Jabber An *oversized* Ethernet/802.3 *frame* and an invalid *CRC checksum*.

Jitter An irregular variation in the timing between a sender's and receiver's respective clocks or an irregular variation in the shape of a signal. In phase jitter, a signal will be out of phase, and in amplitude jitter, a signal's amplitude will vary over time.

Jumbo Frame A proprietary-based Ethernet frame that extends Ethernet's 1500-byte data field to 9000 bytes. Jumbo frame technology enables data transfer rates to approach gigabit speeds on a Gigabit Ethernet LAN, which is currently limited in performance due to the relatively small data field. For example, a 900,000-byte message requires 600 frames that support a 1500-byte data field, but only 100 jumbo frames. Thus, the amount of processing overhead for Gigabit Ethernet is six times more than the jumbo frame Gigabit Ethernet's overhead for the same message.

Kerberos A client/server network security authentication system, developed at MIT and based on *DES* encryption. It is an *Internet* standard that uses a three-pronged approach for authentication: a database that contains users' rights, an authentication server, and a ticket-granting server. Kerberos is named after Cerberus, the three-headed dog in Greek mythology that guarded the gates to Hades.

LAN See *Local Area Network.*

LANE See *LAN Emulation*.

LAN Emulation (LANE) An *ATM* protocol that specifies a technology that enables ATM to emulate Ethernet/802.3 or *token ring* networks. In ATM's protocol hierarchy, LANE is above *AAL5* in the *ATM adaptation layer*. The LANE protocol defines a service interface for the *network layer* that functions identically to the one used by Ethernet/802.3 and token ring LANs. Data that cross this interface are encapsulated in the appropriate *MAC sublayer* format.

LAP-D See *Link Access Protocol-D Channel*.

LAPM See *Link Access Procedure for Modems*.

LATA See *Local Access and Transport Area*.

Latency The amount of delay a network device introduces when data frames pass through it. It is the amount of time a frame spends "inside" a network device. For example, switch latency is usually measured from the instant the first bit of a frame enters the device to the time this bit leaves the outbound (i.e., destination) port.

Layer 2 Forward (L2F) A protocol that provides tunneling between an *ISP's* dialup server and the network.

Layer 2 Tunneling Protocol (L2TP) A method for tunneling PPP sessions across a network. It combines *PPTP* and *L2F*.

Layer-3 Switch A layer-2 switch that is capable of examining layer-3 header information, which is then used to filter network protocols or broadcasts. Also refers to a router that is capable of performing router table lookups and packet forwarding at hardware speeds via application-specific integrated circuit (ASIC) chips.

Layer-4 Switch A *router* that is capable of examining upper layer (layer-4 through layer-7) information to make routing decisions. It is more appropriate to refer to layer-4 switches as either layer-2 or layer-3 application switches because application information from upper layers is being used for routing decisions.

LEC See *Local Exchange Carrier.*

Lightwave Wireless A line-of-sight laser-based connection facility that allows long-distance light-based wireless networking without the need to install cable.

Line-of-Sight A type of wireless transmission that requires the transmitter and receiver to be able to "see" each other; that is, they must be in each other's "line-of-sight."

Line Set Used by the National ISDN Users' Forum to describe the number of multiplexed *B* and *D* channels and the type of *ISDN* service supported.

Link Access Protocol-D Channel (LAP-D) An *ITU* standard on which the *ISDN D* channel is based.

Link Access Procedure for Modems (LAPM) A *modem* protocol that uses *CRC* and ARQ for *error control*. CRC is used for *error detection*; ARQ prevents the modem from accepting any more data until the defective frame has been retransmitted successfully. *V.42*'s default is LAPM. Thus, if a connection is being initialized between two V.42-compliant modems, they will use LAPM for error control. If one of the modems is not V.42-compliant, then the modems will negotiate to use *MNP 1–4*.

Link-State Algorithm A routing algorithm in which routers send each other information about the links they have established to other routers via a link-state advertisement (LSA), which contains the names and various cost metrics of a router's neighbors. LSAs are flooded throughout an entire router's domain. Thus, rather than storing actual paths (which is the case with *distance-vector algorithms*), link-state algorithms store the information needed to generate such paths. An example of a link-state algorithm is Dijkstra's shortest path algorithm, which iterates on length of a path to determine the shortest route.

Lobe A *token ring* node, as defined in the IBM world.

Lobe Length The cable length between *token ring* nodes.

Local Access and Transport Area (LATA) A specific geographical region in which an *LEC* provides local telephone and telecommunications services in the United States. There are 195 LATAs. Services that cross LATA boundaries are provided by *IECs*.

Local Area Network (LAN) A network that interconnects computing resources within a moderately sized geographical area. This can include a room, several rooms within a building, or several buildings on a campus. A LAN's range usually is no more than 10 km in radius.

Local Exchange Carrier (LEC) A telecommunications provider that provides service within a prescribed geographical area. See also *CLEC* and *ILEC*.

Local Loop The circuit that connects the telephone central office or exchange (sometimes called *POP*) with a customer's location. In frame relay, this circuit is called the *port connection* or *access line*. Formally called *digital subscriber loop*.

Logical Link Control (LLC) Sublayer The top sublayer of the data link layer that provides framing, flow control, and error control. Defined in *IEEE 802.2*.

Loop A network configuration in which nodes are connected via dedicated wiring instead of through a centralized hub (as is the case of a *star* design). Loops can be either *simple* (only one connection between any two nodes), *partial* (some nodes are interconnected by more than one link), or *complete* (every node has a connection to every other node). A loop is also referred to as a *meshed* design.

Low-Earth Orbit (LEO) Satellite A satellite placed in orbit at an altitude of 300 to 1200 miles above the earth. Depending on their orbit, a constellation of up to 48 LEO satellites is needed for global coverage.

L2F See *Layer 2 Forward*.

L2TP See *Layer 2 Tunneling Protocol*.

MAC See *Media Access Control Sublayer*.

Manchester Encoding A data transmission encoding scheme that differs from standard digital transmission schemes. Instead of "high" equaling 1 and "low" equaling 0, a timing interval is used to measure high-to-low transitions. Furthermore, instead of a timed transmission period being "all high" or "all low" for either 1 or 0, a 1 is sent as a half-time-period low followed by a half-time-period high, and a 0 is sent as a half-time-period high followed by a half-time-period low. Consequently, the end of the last bit transmitted is easily determined immediately following the transmission of the last bit.

MAE See *Metropolitan Area Exchange*.

MAN See *Metropolitan Area Network*.

MAU See *Media Attachment Unit* or *Multistation Access Unit*.

Media The plural of *medium*.

Media Access Control (MAC) Sublayer The bottom half of the *data link layer* that provides media access management protocols for accessing a shared medium. Example *MAC sublayer* protocols include IEEE 802.3 (*Ethernet*) and IEEE 802.5 (*token ring*).

Media Attachment Unit (MAU) Another term for a *transceiver*.

Media Converter A layer-1 device that enables different network media to be connected to one another.

Medium The physical environment used to connect networked devices. See also *Media*.

Medium-Earth Orbit (MEO) Satellite A satellite placed in orbit at an altitude of 6000 to 12,000 miles above the earth. A constellation of 20 MEO satellites is needed for global coverage.

Meshed Design The interconnectivity among multiple nodes or sites. In a fully meshed design, every node or site is connected with every other node or site. In a partially meshed design, only some nodes or sites are interconnected.

Metric A generic term used in *routing* to represent different quantities such as distance, number of router *hops*, and *bandwidth*.

Metro-Area Satellites A specially equipped jet that flies 50,000 feet above cities to provide wireless, broadband networking service to a metropolitan area.

Metropolitan Area Exchange (MAE) An *Internet* interconnect site similar to a *NAP*. A NAP is funded by the National Science Foundation; a MAE is not. There are currently two MAE points, one each on the east and west coasts of the United States and known as MAE East and MAE West.

Metropolitan Area Network (MAN) A network that interconnects computing resources that span a metropolitan area such as buildings located throughout a local county or city. MANs generally refer to networks that span a larger geographical area than *LANs* but a smaller geographical area than *WANs*.

Microcom Networking Protocol (MNP) Defines various levels of *error correction* and compression for *modems*.

Micron One micrometer (one-millionth of a meter) and abbreviated by the symbol μm. Used in specifying the size of fiber-optic cable.

Microwave An *RF* transmission method that uses high-frequency waves and operates at a higher frequency in the electromagnetic spectrum (usually above 900 MHz). Microwave transmissions are considered a *line-of-sight* medium.

MIME See *Multipurpose Internet Mail Extensions*.

MNP See *Microcom Networking Protocol*.

MNP 1-4 The first four *MNP* levels used for hardware error control. All four levels are incorporated into *V.42*.

MNP 5 The fifth level of *MNP* that incorporates the *MNP 1-4*. Also uses a data compression algorithm that compresses data by a factor of 2 to 1.

MNP 6 The sixth level of *MNP* that supports V.22 *bis* and V.29.

MNP 7 The seventh level of *MNP* that improves *MNP 5*'s data compression algorithm to a 3 to 1 compression factor.

MNP 8 The eighth level of *MNP* that extends *MNP 7*; enables half-duplex devices to operate in full-duplex mode.

MNP 9 The ninth level of *MNP* that is used in a variety of circuits.

MNP 10 The tenth level of *MNP* that is used in cellular modems and in situations where line quality is poor.

Mobile IP An IETF standard that makes it possible for *cellular communications network* users to roam within or outside their company's network but still maintain their home-network-based IP address.

Modem An acronym for *modulator/demodulator*. A modem transforms (modulates) a computer's digital signal into analog form at the sending side so the signal can be carried across a standard telephone line. On the receiving side, a modem demodulates the signal—it reconverts the transmitted analog signal from the phone line to digital form before it is passed to the computer.

Modulation A method in which a characteristic of an electromagnetic wave is altered. Types of modulation include *amplitude modulation*, *frequency modulation*, *phase modulation*, and *pulse-code modulation*.

Molecular Electronics The concept in which the current development of switches, transistors, and logic gates can be designed from individual molecules.

Moving Picture Experts Group (MPEG) A series of compression standards used for compressing multimedia data.

MPEG See *Moving Picture Experts Group*.

Multicast A data transmission that is destined to a group of recipients.

Multidrop Design A network configuration in which each system node is connected to a common cable plant and assigned a specific number that is used to communicate with the system and also to establish priority of when a system will be communicated with from a master control system. Primarily used in factories.

Multilink PPP (MP) An *IP* protocol that combines multiple physical links (i.e., telephone lines) into a single high-capacity channel. Unlike *BONDING*, which is implemented in hardware, MP is achieved via software. MP is also applicable to analog dialup connections.

Multimedia Networking Involves transmitting information as a combination of full-motion video, sound, graphics, text, and animation across a network.

Multimode Fiber A type of fiber-optic cable with a core diameter ranging from 50 to 100 μm. In multimode fiber, different rays of light bounce along the fiber at different angles as they travel through the core. This results in some degree of signal distortion at the receiving end. Multimode fiber can be of two types: *graded-index* or *step-index*.

Multiplexer A device that does *multiplexing*. Also called a *mux* for short.

Multiplexing A technique used to place multiple signals on a single communications channel. Multiplexing partitions a channel into many separate channels, each capable of transmitting its own independent signal, thereby enabling many different transmissions over a single medium.

Multipurpose Internet Mail Extensions (MIME) An extension to the *Simple Mail Transport Protocol (SMTP)* that extends the concept of e-mail by providing support for different data types and for complex message bodies. MIME enables users to exchange nontext files such as graphics, video, or audio in addition to plaintext files.

Multistation Access Unit (MAU) A token ring hub.

Mux Abbreviation for *multiplexer*.

NADH See *North American Digital Hierarchy*.

NAP See *Network Access Point*.

National Information Infrastructure (NII) A federal policy initiative to facilitate and accelerate the development and utilization of the nation's information infrastructure. The perception of the NII is one of a "seamless web" of telecommunications networks consisting of computers, specialized databases, radios, telephones, televisions, and satellites. The NII is expected to provide consumers with convenient and instantaneous access to nearly any kind of information ranging from research results to medical and educational material to entertainment.

NC See *Network Computer.*

Near-End Crosstalk (NEXT) See *Crosstalk.*

netstat A UNIX program that generates a local host's routing table. Similar output can be generated on a Windows NT system using the command *route print*.

Network Access Point (NAP) An *Internet* traffic exchange point that provides centralized Internet access to Internet service providers. A NAP serves as a critical regional "switching station" where all different network backbone providers meet and exchange traffic on each other's backbone.

Network Appliance Any one of several powerful computing devices designed to support a single dedicated application such as Web browsing or e-mail. In some cases, network appliances do not have keyboards or monitors.

Network Architecture A formal, logical structure that defines how network devices and software interact and function; defines communication protocols, message formats, and standards required for interoperability.

Network Computer (NC) An inexpensive ($500 or less) network access device with functionality that allows some applications to be run, but not as complete as what would typically be found on a PC or a workstation of some sort. NCs are stripped-down systems that use the network to access their applications dynamically.

Network Convergence The integration of different network technologies and applications, which when combined, create powerful new entities and products. One example of network convergence is IP telephony, which integrates voice and data onto an IP network. Network convergence, however, involves more than simply combining voice and data.

Network Diameter The overall length between a network's two remotest nodes.

Network Ethics Specific standards of moral conduct by network users for the responsible use of network devices and resources.

Network Interface Card (NIC) A layer-2 device that performs standard *data link layer* functions, including organizing data into frames, transferring frames between the ends of a communication channel, and managing the link by providing error control, initialization, control termination, and flow control. A NIC is also known as a *LAN* adapter, network adapter, network card, and network board. When used in Ethernet/802.e networks, a NIC is called an Ethernet card or adapter.

Network Operating System (NOS) Software that is installed on a system to make it network capable. Examples include IBM's LAN Server, Banyan's VINES, and Novell's NetWare (also known as IntranetWare). A NOS is independent of a computer's native operating system—it is loaded "on top" of the computer's operating system and provides the computer with networking capability based on a particular protocol. If an operating system provides built-in network support (e.g., Microsoft's Windows NT and Sun's Solaris), then the OS is called a *networkable* operating system.

Network Protocol A formal specification that defines the vocabulary and rules of data communication. See also *Protocol*.

Network Protocol Suite A set of related and interoperating network protocols. An example is the *TCP/IP* protocol suite, which consists of protocols for e-mail, Web service, file transfers, and routing.

Network Security The proper safeguarding of everything associated with a network, including data, media, and equipment. It involves administrative functions, such as *threat assessment*, technical tools and facilities such as cryptographic products, and network access control products such as *firewalls*. It also involves making certain that network resources are used in accordance with a prescribed policy and only by people who are authorized to use these resources.

Network Service Access Point (NSAP) An *OSI* addressing mechanism used by private *ATM* networks. NSAPs are 20-byte addresses and include a 13-byte prefix that can be used to identify a specific location, including a country, region, or end system.

Network Standards A formal set of rules, developed by and agreed upon by various organizations, defining hardware interfaces, communication protocols, and network architectures. Several standards exist, including *de jure, de facto, proprietary,* and *consortia*.

Network Termination Unit (NTU) A device that terminates *E-1* circuits. An NTU provides broadly similar *CSU/DSU* functionality.

Network Topology The basic design of a computer network that details how key network components such as nodes and links are interconnected.

NEXT See *Crosstalk.*

Next Generation Internet (NGI) A high-speed networking initiative that forges collaborative partnerships between the private and public sectors.

NGI See *Next Generation Internet.*

NIC See *Network Interface Card.*

NII See *National Information Infrastructure.*

Node Another name for a *device*. Usually used to identify computers that are network hosts, workstations, or servers.

Noise Any undesirable, extraneous signal in a transmission medium. There are generally two forms of noise—*ambient* and *impulse*. Noise degrades the quality and performance of a communications channel and is one of the most common causes of transmission errors in computer networks.

Nonpersistent CSMA A *CSMA*-based protocol in which a node continually waits a random period of time whenever it detects a busy channel. Once it senses an idle channel, it may then transmit data.

Nonreturn to Zero, Invert On Ones (NRZI) An encoding technique used for converting digital data into digital signals. In NRZI, encoding is based on transitions from one voltage state to another (i.e., from a low-to-high state or from a high-to-low state). Specifically, data are coded 0 if no transitions occur and 1 at the beginning of a transition.

North American Digital Hierarchy (NADH) A multiplexed *T1* structure used in North America that combines multiple T1 lines into higher-rated T-carrier circuits. For example, a *T2* circuit consists of four multiplexed T1 circuits and has an aggregate bandwidth of 6.312 Mbps; a *T3* link consists of 28 multiplexed T1 circuits with an aggregate bandwidth of 44.736 Mbps; and a *T4* channel consists of 168 multiplexed T1 circuits and is rated at 274.176 Mbps.

NOS See *Network Operating System.*

NRZI See *Nonreturn to Zero, Invert on Ones*.

NSAP See *Network Service Access Point*.

nslookup A UNIX and Microsoft NT program used to acquire the *IP address* of a *domain name*. This program can also be used for IP address resolution, which translates a numerical IP address to its corresponding domain name.

NTU See *Network Termination Unit.*

Nyquist's Theorem A sampling theorem that states that the maximum signaling rate of a noiseless channel is twice the number of samples. Thus, if we have w cycles per second (i.e., hertz), then we can have $2w$ signal states. Generalizing this concept, if a noiseless communications channel uses N values per signaling state, then the channel's maximum data transmission capacity in bits per second is given as $2w\text{Log}_2N$, where w = the number of cycles expressed in hertz and N = the number of discrete signaling states used.

1-persistent CSMA A CSMA-based protocol in which a node continuously monitors a shared channel until it is idle and then seizes the channel and begins transmitting data. The "one" in 1-persistent represents the probability that a single waiting node will be able to transmit data once it detects an idle channel ($p = 1$).

OC See *Optical Carrier.*

Omnidirectional Antenna An antenna used in *cellular communications networks* that provides 360° of coverage; thus, the signal spreads in all directions during transmission.

Open Shortest Path First (OSPF) An *interior gateway protocol* based on a *link-state algorithm*. Designed for large, heterogeneous *IP* networks.

Optical Carrier (OC) A *fiber-optic* digital transmission hierarchy used for *SONET.* OC rates range from OC-1, which is the equivalent of 28 DS-1 channels (51.84 Mbps), to OC-192, which is the equivalent of 5376 DS-1 channels (9.953 Gbps). OC rates are the optical equivalent of *STS* rates.

OSI An acronym for open systems interconnection. See *OSI Model.*

OSI Model A network architecture for developing network protocol standards. The OSI model formally defines and codifies the concept of *layered* network architecture. It uses well-defined operationally descriptive layers that describe what happens at each stage in the processing of data for transmission. The OSI model consists of the following seven layers, which are numbered in descending order: application (7), presentation (6), session (5), transport (4), network (3), data link (2), and physical (1).

OSPF See *Open Shortest Path First*.

Oversized Frame An Ethernet/802.3 frame with more than 1518 bytes but a valid *CRC checksum.*

Oversubscription In *frame relay*, when the capacity of a frame relay connection into the frame relay network is less than the total bandwidth guaranteed by the provider. More specifically, the *port speed* is less than the aggregate *CIR*.

Packet The smallest unit of information that is transferred across a packet-switched network. In *TCP/IP*, a packet is called a *datagram*.

Packet-Filtering Firewall A *router* or a dedicated device that filters network access at the network layer by examining packet addresses (source and destination).

Packet-Switched Network A network design that enables nodes to share a communications channel via a *virtual circuit*. Messages are partitioned into smaller messages called *packets*, which may contain only a few hundred bytes of data, accompanied by addressing information. Packets are sent to the destination node one at a time, at any time, and not necessarily in a specific order. The network hardware delivers the packets through the virtual circuit to the specified destination node, which is responsible for reassembling them in the correct order.

Packet-Switching A switching technique in which messages are partitioned into smaller units called packets, which contain addressing information as well as sequence numbers. Packets are then sent to the destination node one at a time, at any time, and not necessarily in a specific order. Furthermore, the channel used to transmit a packet is occupied only for the duration of the packet's transmission. Two general packet-switching strategies are virtual circuit packet-switching and datagram packet-switching. In *virtual circuit packet-switching*, a nondedicated, shared path is established between sending and receiving nodes; all packets traverse this path in sequence via a store-and-forward transmission in which complete packets are first stored on an intermediate node and then forwarded to a successive node along the path en route to the destination. In *datagram packet-switching*, packets are transmitted independently of each other and not necessarily along the same path. Thus, packets can arrive out of sequence.

PAN See *Personal Area Network.*

PAR See *Positive Acknowledgment with Retransmission.*

Parallel Communication A data transmission method in which the bits representing a character of data are transmitted simultaneously on separate channels. Also called *parallel transmission.*

Parity The use of an extra bit (called a *parity bit* or a *redundancy bit*) to detect single-bit errors in data transmissions. Parity can be specified as even, odd, or none. Even parity means that there must be an even number of 1-bits in each bit string; odd parity means that there must be an odd number of 1-bits in each bit string; and no parity means that parity is ignored. The extra bit (i.e., the parity bit) is forced to either 0 or 1 to make the total number of bits even or odd.

Partitioning A network configuration strategy that involves dividing a LAN into several separate (but still interconnected) network segments. Also called *segmentation*.

PBX See *Private Branch Exchange*.

PC Card A layer-2 plug-in adapter used in portable or laptop computers. Three different "types" are available. Type 1 cards are 3.3 millimeters thick and enhance the memory capabilities of a device; Type II cards are 5 mm thick and are used for modems and network adapters for both Ethernet and token ring; Type III cards are 10.5 mm thick and are generally either miniature hard disks or wireless *NICs*; and Type IV cards, when produced, will be approximately 16 mm thick and support hard disk drives that have a capacity greater than what is currently available from Type III cards. PC cards were formerly known as *PCMCIA cards*.

PCM See *Pulse Code Modulation.*

PCMCIA Card A layer-2 device that was originally designed to serve as memory cards for microcomputers. These cards are now known as *PC cards*. PCMCIA stands for Personal Computer Memory Card International Association.

Peer-to-Peer A model or paradigm on which some network communications and applications are based. In a peer-to-peer environment, each networked host runs both the client and server parts of an application.

Period The reciprocal of the frequency. It is the amount of time it takes to complete a single cycle, that is, seconds per cycle.

Permanent Virtual Circuit (PVC) A communications channel that provides a logical connection between two sites instead of a physical one. In a *connection-oriented* protocol such as *frame relay*, PVCs appear as *private links* because a circuit must be established between end nodes prior to data communications. The difference is PVCs are virtual circuits, not dedicated ones, and hence bandwidth is shared among multiple sites by *multiplexing* techniques. Thus, PVCs provide nondedicated connections through a shared medium, which enables data from multiple sites to be transmitted over the same link concurrently.

Personal Area Network (PAN) A home-based computer network.

PGP See *Pretty Good Privacy.*

Phase A characteristic of a sine wave that specifies the point a wave has advanced within its cycle. From a mathematical perspective, phase is generally expressed in degrees or radians. For example, a 90° (or $\pi/2$ radians) phase shift means that one-fourth of the wave has been shifted, which indicates that it has advanced one-fourth of the period.

Phase Modulation (PM) A *modulation* technique in which the phase angle, which is the phase difference between two signals, is manipulated in some manner. This manipulation effectively delays the natural flow of the waveform temporarily.

Phase-Shift Keying (PSK) Similar to *phase modulation* except phase-shift keying is the more appropriate term when used in the context of converting digital data (or signals) into analog signals at the sending end and then back to digital form at the receiving end. PSK modifies the phase angle of the carrier wave based on the digital data being transmitted. The changes in phase angle are what convey the data in a phase modulated signal. In its simplest implementation, one phase represents a binary 0 and a second phase represents a binary 1.

Physical Layer The lowest layer (layer 1) of the *OSI model*. The *physical layer* translates *frames* received from the *data link layer* (layer 2) into electrical, optical, or electromagnetic signals representing 0 and 1 values, or bits. Abbreviated PHY in the documentation.

ping A UNIX and Microsoft NT program used to test the communication path between source and destination nodes. Ping is an *ICMP*-based application and is an acronym for "packet Internet groper."

Pinout The electrical signals associated with pins and connector. Also called pin assignment.

PKI See *Public Key Infrastructure*.

Plain Old Telephone System (POTS) The conventional telephone network. See also *Public Switched Telephone Network.*

Plaintext An uncoded message; a message in its original, meaningful (uncoded) form.

Plastic Fiber A type of fiber-optic cable in which the fibers (i.e., conductors) are constructed of plastic instead of glass.

Plenum Cable Any type of cable that contains an outer sheath or "jacket" that is composed of a Teflon coating. Plenum cable is used for cable "runs" through a return air system. In case the cable burns during a fire, both PVC and Teflon give off nasty toxic gases when burning. Teflon, however, is fire retardant and takes much longer to get to a burning point.

PM See *Phase Modulation*.

Point of Presence (POP) A telco's central office or switching station.

Point-to-Point Network A network design in which only adjacent nodes (nodes that are next to each other and only one hop away) can communicate with one another.

Point-to-Point Tunneling Protocol (PPTP) A protocol that provides encryption and authentication for remote dialup and LAN-to-LAN connections. PPTP establishes two types of connections: A control session for establishing and maintaining a secure tunnel from sender to receiver, and a data session for the actual data transmission.

POP See *Point of Presence* or *Post Office Protocol*.

Port Connection The *local loop* in *frame relay*. Also called *access line*.

Port Speed In *frame relay*, the data transmission rate in bits per second of the *local loop*.

Positive Acknowledgment with Retransmission (PAR) An error-control strategy used in TCP in which a receiver acknowledges all successfully transmitted frames. If the sender does not receive a positive acknowledgment from the receiver within a certain time frame, then it will assume the data segment did not arrive, or it arrived damaged, and retransmit the data.

Post Office Protocol (POP) A TCP/IP application protocol used for transferring e-mail from a server to a desktop PC client.

POTS See *Plain Old Telephone System*.

PPTP See *Point-to-Point Tunneling Protocol*.

Pretty Good Privacy (PGP) A *public key* application developed by Phil Zimmerman for e-mail security.

PRI See *Primary Rate Interface*.

Primary Rate Interface (PRI) An *ISDN* primary access channel. The ANSI ISDN PRI standard, which is used in the United States, comprises twenty-three 64-kbps channels for voice or data—called *B* channels—and one 64-kbps control channel—called a *D* channel. This format, known as $23B + D$, is based on the North American DS-1 service and provides 1.544 Mbps. The ITU-T ISDN PRI standard supports a $30B + 2D$ format that is based on the European E-1 standard and provides 2.048 Mbps.

Private Branch Exchange (PBX) A telephone exchange used within an organization to provide internal telephone extensions and access to the public telephone network; it is the modern-day equivalent of what used to be called a switchboard.

Private Link A communications channel that provides a private, dedicated link between two sites. Also commonly referred to as standard leased line.

Private Switch One application of an Ethernet switch. A private switch supports only one *MAC* address per port, which provides each node with its own dedicated segment. This eliminates contention for the cable, thereby liberating the end nodes from performing collision detection.

Probabilistic Network An artificial intelligence-network convergence strategy in which numbers are assigned to degrees of uncertainty about each bit of data or possible outcome so that a device can construct rules about how it can manipulate the numbers.

Promiscuous Mode A state in which an Ethernet interface can be placed so that it can capture every frame that is transmitted on the network. For example, an *Ethernet NIC* set in promiscuous mode collects all messages placed on the medium regardless of their destination address.

Propagation Delay The time a signal takes getting from one point in a circuit to another.

Proprietary Standards Network standards that are developed in a manufacturer-specific manner. Their specifications are not in the public domain and are only used and accepted by a specific vendor.

Protocol An accepted or established set of procedures, rules, or formal specifications governing specific behavior or language. When applied to networks, a *network protocol* is a formal specification that defines the vocabulary and rules of data communication.

Proxy Server A device or product that provides network protection at the application level by using custom programs for each protected application. These custom-written application programs act as both a client and server and effectively serve as proxies to the actual applications. Also called application gateway firewall or proxy gateway.

PSK See *Phase-Shift Keying.*

PSTN See *Public Switched Telephone Network.*

Public Key A special code, available in the public domain, that can be used to code and decode messages.

Public Key Infrastructure (PKI) A network security infrastructure that involves the process of issuing, delivering (i.e., publishing), managing, and revoking public keys to effect secure transmissions across a public network such as the Internet.

Public Switched Telephone Network (PSTN) The traditional analog-based telephone system used in the United States that was originally designed for voice transmissions.

Pulse-Code Modulation (PCM) A coding technique used to convert analog signals to digital signals and vice versa.

PVC See *Permanent Virtual Circuit*. Also an abbreviation for polyvinyl chloride.

PVC Cable Any type of cable that contains an outer sheath or "jacket" that is composed of polyvinyl chloride (PVC). Also called *nonplenum cable*.

QoS See *Quality of Service.*

Quadrature Amplitude Modulation (QAM) A modulation technique used by *modems* that involves the combined effects of *phase modulation* and *amplitude modulation*. QAM uses eight phase changes and two amplitudes to create 16 different signal changes.

Quality of Service (QoS) Parameters associated with data prioritization that specify such things as the amount of bandwidth a priority data transmission requires as well as the maximum amount of latency the transmission can tolerate in order for the transmission to be meaningful. QoS is needed for transmitting real-time voice and video traffic.

RA See *Routing Arbiter.*

Radio A general term used in wireless communications that specifies the 3 kHz to 300 GHz frequency range of the electromagnetic spectrum.

Radio Frequencies (RF) Any transmission method that uses electromagnetic waveforms.

Radio Transmission Any wireless technique that uses *RF* to transmit information.

RADSL See *Rate-Adaptive Digital Subscriber Line.*

Random Access Protocol A network protocol that governs how nodes are to act in those instances where accessing a shared medium at will on a first-come first-served basis is permitted. Also called *Contention Protocol*.

Rate-Adaptive Digital Subscriber Line (RADSL) A nonstandard *DSL* variant that provides transmission rates similar to *ADSL*. Transmission rates can be adjusted automatically based on distance and line quality.

RBOC See *Regional Bell Operating Company.*

Real-Time Transport Control Protocol (RTCP) A companion protocol to *RTP* that provides monitoring and feedback support. Working in concert with RTP, RTCP monitors the *quality of service* each destination node is receiving and conveys this information to the sender.

Real-Time Transport Protocol (RTP) An IETF standard that provides an end-to-end connection-oriented protocol to support delay-sensitive data types such as audio and video.

Redundancy Bits Extra bits incorporated into a data frame that provide error-correction information. A data set composed of both user data and redundancy bits is called a *codeword*. Also called *check bits*.

Regional Bell Operating Company (RBOC) A regional telephone company in the United States formed after the AT&T breakup in 1984.

Reliable Service A type of service that requires a sending node to acknowledge receipt of data. This is called an acknowledged datagram service.

Repeater A layer-1 device that provides both physical and electrical connections. Its function is to regenerate and propagate signals—it receives signals from one cable segment, regenerates, retimes, and amplifies them, and then transmits these "revitalized" signals to another cable segment. Repeaters extend the diameter of Ethernet/802.3 networks but are considered part of the same collision domain.

Request for Comments (RFC) The working notes of the Internet research and development community. RFCs provide network researchers and designers with a medium for documenting and sharing new ideas, network protocol concepts, and other technically related information. They contain meeting notes from Internet organizations, describe various Internet protocols and experiments, and detail standards specifications. All Internet standards are published as RFCs (not all RFCs are Internet standards, though).

Resource Reservation Protocol (RSVP) A layer-3 protocol developed by *IETF* to provide a mechanism to control network latency for specific applications. This is done by prioritizing data and allocating sufficient bandwidth for data transmission. RSVP can be thought of as an IP-based *QoS* protocol.

RF See *Radio Frequencies.*

RFC See *Request for Comments*.

Ring Design A network design that is based on a broadcast topology in which nodes are connected to a physical ring and data messages are transferred around the ring in either a clockwise or counterclockwise manner (or both).

RIP See *Routing Information Protocol*.

RIP-2 An updated version of *RIP*, formally known as RIP Version 2. New features include authentication, interpretation of *IGP* and *BGP* routes, *subnet mask* support, and multicasting support.

Risk Analysis The assessment of how much a loss is going to cost a company.

RJ A specific series of connectors defined in the Universal Service Order Code (USOC) definitions of telephone circuits. RJ is telephone lingo for "registered jack."

RJ-11 A four-wire modular connector used for telephones.

RJ-45 An eight-wire modular connector used in 10BASE-T LANs.

Router A layer-3 device that is responsible for determining the appropriate path a packet takes to reach its destination. Commonly referred to as *gateway*.

Routing A layer-3 function that directs data packets from source to destination.

Routing Arbiter (RA) A project that facilitates the exchange of network traffic among various independent Internet backbones. Special servers that contain routing information databases of network routes are maintained so that the transfer of traffic among the various backbone providers meeting at a *NAP* is facilitated.

Routing Information Protocol (RIP) A distance-vector algorithm that determines the best route by using a *hops* metric. RIP was at one time the *de facto standard* for IP routing.

Routing Protocol A specific *protocol* that determines the route a packet should take from source to destination. Routing protocols are a function of network protocols. For example, if your network protocol is *TCP/IP*, then several routing protocol options are available including *RIP*, *RIP-2*, and *OSPF*. If your network protocol is OSI's CNLP, then your routing protocol is IS-IS. Routing protocols determine the "best" path a packet should take when it travels through a network from source to destination and maintain routing tables that contain information about the network's topology. Routing protocols rely on routing algorithms to calculate the least-cost path from source to destination.

Routing Table A data structure that contains, among others, the destination address of a node or network, known router addresses, and the network interface associated with a particular router address. When a router receives a packet, it looks at the packet's destination address to identify the destination network, searches its routing table for an entry corresponding to this destination, and then forwards the packet to the next router via the appropriate interface.

RSA A *public key* encryption algorithm for encoding data. The abbreviation stands for Rivest, Shamir, and Adleman, the last names of the three individuals who designed it.

RSVP See *Resource Reservation Protocol*.

RTCP See *Real-Time Transport Control Protocol*.

RTP See *Real-Time Transport Protocol*.

Runt Frame An Ethernet/802.3 frame that has at least 8 bytes but fewer than 64 bytes and has a valid CRC checksum.

SAN See *Storage Area Network*.

SAP See *Service Access Point*.

SAS See *Single-Attachment Station*.

Satellite Communication System An *RF*-based broadcast network design involving earth ground stations and orbiting communication satellites. Data transmissions from a land-based antenna to the satellite (called the uplink) are generally point-to-point, but all nodes that are part of the network are able to receive the satellite's transmissions (called the downlink).

SC Connector A TIA/EIA-568A standard connector for fiber-optic cable; also called a *568SC connector*.

SCSI Acronym for small computer systems interface (pronounced "scuzzy"), which is a high-speed parallel interface used to connect a computer to peripheral devices such as scanners, printers, CD-ROM drives, and mass storage devices such as ZIP drives via a single computer port. Up to eight SCSI devices can be interconnected in a daisy-chained configuration.

SDH See *Synchronous Digital Hierarchy.*

SDR See *Software-Defined Radio*.

SDSL See *Symmetric Digital Subscriber Line*.

Segmentation See *Partitioning*.

Serial Communication A data transmission method in which the bits representing a character of data are transmitted in sequence, one bit at a time, over a single communications channel. Also referred to as *serial transmission*.

Server A networked device that provides resources to *client* machines. Examples include print servers, mail servers, file servers, and Web servers. Servers are shared by more than one user; clients have only a single user.

Service Access Point (SAP) Specified within the *OSI model*, a SAP refers to a location through which a process operating at one layer provides services to a process operating at the layer above it.

Service Profile Identification (SPID) Numbers assigned by the telcos and used to identify the various processes of an *ISDN* device. (Used only in North America.)

Session Initiation Protocol (SIP) An IETF protocol suite for *VoIP* that was developed at Columbia University to resolve some of *H.323*'s deficiencies, including the use of a fixed port number and H.323's complexity.

Shannon's Limit A mathematical theorem that describes a model for determining the maximum data rate of a noisy analog communications channel. Shannon's limit is given by the following formula: Maximum Data Rate (MDR) = $H \log_2(1 + (S/N))$, where MDR is given in bits per second, H = bandwidth in hertz, and (S/N) is a measure of the *signal- to-noise ratio*. Named for the mathematician who derived it, Claude Shannon.

Shielded Twisted-Pair (STP) Twisted-pair cable in which individual wire pairs are shielded (i.e., protected from *noise*).

Short Message Services (SMS) A feature of the de facto *GSM* cellular standard that is based on the store-and-forward transmission concept. Messages up to 160 characters long are sent to an SMS center where the message is stored and then transmitted to the recipient. The SMS center also transmits a delivery confirmation message to the sender. An SMS message can include any alphanumeric characters as well as nontext data such as a binary-formatted message, and can be transmitted and received simultaneously with a GSM-based cell call.

Signal Quality Error (SQE) A signal generated by a transceiver and read by the controller of the host to which the transceiver is connected. In V2.0 Ethernet, SQE is called *heartbeat* and is generated periodically to inform the host's controller that the transceiver is "alive." In IEEE 802.3, SQE is only generated when a real signal quality error occurs.

Signal-to-Noise Ratio (SNR) A measure of signal quality expressed in decibels (dB). It is the ratio of signal strength to background noise on a cable. More specifically, SNR is the ratio between the desired signal and the unwanted noise in a communications medium.

Simple Mail Transport Protocol (SMTP) A TCP application protocol for exchanging electronic mail.

Simplex Communication A data transmission method in which data may flow in only one direction; one device assumes the role of sender and the other assumes the role of receiver. These roles are fixed and cannot be reversed. An example of a simplex communication is a television transmission.

Single-Attachment Station (SAS) An FDDI node that is connected to only the primary pair of fibers and can be isolated from the network in the case of some types of failure. An SAS is also called a *Class B* node.

Single-Mode Fiber A type of fiber-optic cable with a core diameter ranging from 7 to 9 µm. In single-mode fiber, only a single ray of light, called the *axial ray*, can pass. Thus, a light wave entering the fiber exits with very little distortion, even at very long distances and very high data rates.

SIP See *SMDS Interface Protocol* or *Session Initiation Protocol*.

Sliding Window Protocol A *flow-control* strategy that enables a sender to transmit frames continuously without having to wait for acknowledgements to these frames from the receiver. The sliding window concept is implemented by requiring the sender to sequentially number each data frame it sends and by having the sender and receiver maintain information about the number of frames they can respectively send or receive.

SMA Connector A fiber-optic cable connector that meets military specifications.

Smart Card A type of "credit card" with embedded integrated circuits that stores information in electronic form and is used for authentication. Similar to a *digital certificate*.

SMDS See *Switched Multimegabit Data Service*.

SMDS Interface Protocol (SIP) *SMDS* protocol that consists of three different levels: SIP Level 3, SIP Level 2, and SIP Level 1. These three protocol levels are similar in function to the first three layers of the *OSI* model but represent SMDS's MAC sublayer and hence operate at the data link layer.

SMS See *Short Message Service*.

SMTP See *Simple Mail Transport Protocol*.

SNR See *Signal-to-Noise Ratio.*

Software-Defined Radio (SDR) A future-based technology that will transform a *cellular communications network* into a software-based system that will enable users to download software directly into their handsets, giving them additional functionality.

SOHO An abbreviation for *small office/home office*.

SONET See *Synchronous Optical Network.*

Spanning Tree A single path between source and destination nodes that does not include any loops. It is a loop-free subset of a network's topology. The spanning tree algorithm, specified in IEEE 802.1d, describes how bridges (and switches) can communicate to avoid network loops.

SPID See *Service Profile Identification*.

Split-Horizon A strategy employed by *RIP* to ensure that a router never sends routing information back in the direction from which it came. Used to prevent routing loops.

Split-Horizon with Poisoned Reverse A modified *split-horizon* strategy in which routing information provided by a neighbor is included in updates sent back to that neighbor. Such routes are assigned a cost factor of infinity, which makes the network unreachable.

Spread Spectrum A radio technology that refers to a security technique. Spread spectrum transmission camouflages data by mixing signals with a pseudonoise (PN) pattern and transmitting the real signal with the PN pattern. The transmission signal is spread over a range of the frequencies in the radio spectrum.

SQE See *Signal Quality Error.*

Stackable Repeater Hub Individual repeater units "stacked" one on top of another. Instead of using a common shared backplane, stackable hubs use a "pseudo-backplane" based on a common connector interface. An external cable interconnects the individual hubs in a daisy-chained manner. Once interconnected, the entire chain of hubs becomes a single logical unit that functions as a single repeater.

Stacking Height The maximum number of stackable repeater hubs permitted.

Standby Monitor A station (i.e., lobe) on a token ring network that oversees the *active monitor*. Except for the active monitor, all token ring lobes are standby monitors.

Star A network configuration characterized by the presence of a central processing hub, which serves as a wire center for connecting nodes. All data must pass through the hub in order for nodes to communicate with each other.

Stateful Firewall A device or product that monitors all transactions between two systems and is capable of (a) identifying a specific condition in the transaction between two applications, (b) predicting what should transpire next in the transaction, and (c) detecting when normal operational "states" of the connection are being violated.

Static Route A fixed route that is entered into a router's *routing table* either manually or via a software configuration program.

Statistical Multiplexing A *multiplexing* technique that allocates part of a channel's capacity only to those nodes that require it (i.e., have data to transmit). Based on the premise that, statistically, not all devices necessarily require a portion of the channel at exactly the same time.

ST Connector Similar to a *BNC connector* but used with *fiber-optic cable*.

Step-Index Multimode Fiber A type of multimode fiber in which light pulses are guided along the cable from source to destination by reflecting off the cladding.

STM See *Synchronous Transport Module*.

Stop-and-Wait Protocol A *flow-control* strategy that requires the sender to transmit one frame and then wait for the receiver to acknowledge receipt of this frame before any additional frames are transmitted.

Storage Area Network (SAN) A network dedicated exclusively for storing data. Usually involves *Fibre Channel* technology.

Store-and-Forward A method used by bridges and switches in which the contents of an entire frame are captured by the device before a decision is made to filter or forward the frame. A store-and-forward network switch is also called a *buffering switch*. A network based on this principle is called a *store-and-forward network*.

STP (Shielded Twisted-Pair) See *Twisted-Pair Cable*.

STS See *Synchronous Transport Signal*.

Subnet Mask A special network address used to identify a specific subnetwork. Using a unique bit combination, a mask partitions an address into a network ID and a host ID.

Subnetting The partitioning of a network address space into separate, autonomous *subnetworks*. Key to subnetting is a network's *subnet mask*.

Subnetwork A network segment. Commonly abbreviated as *subnet*.

SVC See *Switched Virtual Circuit*.

Switch A network device that filters or forwards data based on specific information. A layer-2 switch (e.g., an Ethernet switch) filters or forwards frames from one node to another using MAC-level (i.e., hardware) addresses; a layer-3 switch filters or forwards packets based on network addresses; and a layer-4 (or higher) switch filters or forwards messages based on specific application protocols. Forwarding rates are usually done at wire speed and via "private" connections (i.e., no other node "sees" the traffic). Switches partition Ethernet/802.3 networks into multiple collision domains.

Switched Ethernet An Ethernet/802.3 LAN that is based on network switches instead of repeaters or bridges. A switched Ethernet/802.3 LAN isolates network traffic between sending and receiving nodes from all other connected nodes. It also transforms traditional Ethernet/802.3 from a broadcast technology to a point-to-point technology.

Switched Multimegabit Data Service (SMDS) A cell-based, connectionless, high-speed, public, packet-switched, broadband, metropolitan area data network.

Switched Virtual Circuit (SVC) A circuit between source and destination nodes that is established on the fly and then removed after data communications have ended. SVCs are logical dynamic connections instead of logical permanent connections, as with *PVCs*. Thus, SVCs provide switched, on-demand connectivity.

Symmetric Cryptography See *Cryptography.*

Symmetric Digital Subscriber Line (SDSL) A *DSL* variant in which traffic is transmitted at the same rate in each direction. Maximum transmission rate is 768 kbps. Uses single-wire pair. Telephone service not supported. Suitable for videoconferencing.

Synchronous Communication A data communication method that requires sending and receiving nodes to monitor each other's transmissions so that the receiving node always knows when a new character is being sent. In this instance, the sending and receiving nodes are "in sync" with each other.

Synchronous Digital Hierarchy (SDH) An *ITU physical layer* standard that provides an international specification for high-speed digital transmission via optical fiber. SDH incorporates *SONET* and uses the *STM* signal hierarchy as its basic building block. SDH is essentially the same as SONET. At OC-3 rates and higher, the two are virtually identical.

Synchronous Optical Network (SONET) An ANSI *physical layer* standard that provides an international specification for high-speed digital transmission via optical fiber. At the source interface, signals are converted from electrical to optical form. They are then converted back to electrical form at the destination interface. The basic building block of the SONET signal hierarchy is *STS-1* (51.84 Mbps). See also *Synchronous Digital Hierarchy.*

Synchronous Transport Module (STM) Represents a digital transmission carrier system used for *SDH*. STM rates range from STM-1, which is equivalent to OC-3 (155.52 Mbps), to STM-64, which is equivalent to OC-192 (9.953 Gbps).

Synchronous Transport Signal (STS) A digital transmission hierarchy used for *SONET.* STS rates range from STS-1, which is the equivalent of 28 DS-1 channels (51.84 Mbps), to STS-

192, which is the equivalent of 5376 DS-1 channels (9.953 Gbps). STS rates are the electrical equivalent of *OC* rates.

T-1 The multiplexing of 24 separate voice channels, each rated at 64 kbps, plus one 8-kbps framing channel into a single wideband digital signal rated at 1.544 Mbps.

T-2 A multiplexed circuit that combines 4 *T1* circuits and has an aggregate bandwidth of 6.312 Mbps.

T-3 A multiplexed circuit that combines 28 *T1* circuits and has an aggregate bandwidth of 44.736 Mbps.

T-4 A multiplexed circuit that combines 168 *T1* circuits and has an aggregate bandwidth of 274.176 Mbps.

TCP See *Transmission Control Protocol*.

TCP/IP An acronym for *Transmission Control Protocol/Internet Protocol*. A formal network protocol suite based on its two namesake subprotocols, *TCP* and *IP*.

TDM See *Time Division Multiplexing.*

TDMA See *Time Division Multiple Access*.

TE See *Terminal Equipment*.

Telco An acronym for *telephone company.*

Terminal Adapter (TA) A device that connects noncompatible ISDN devices to an ISDN network. If a TA is used for an ISDN dialup connection, then it can be thought of as a modem. If a TA is used to connect a device to a LAN, then it can be thought of as a network interface card. It should be noted that, although a TA is frequently referred to as an ISDN modem or digital modem in the context of an ISDN dialup connection, this reference is incorrect. By definition, a modem performs analog-to-digital and digital-to-analog conversions. Since ISDN is completely digital, no such conversions are necessary; thus, the expressions "ISDN modem" or "digital modem" are incongruous.

Terminal Equipment (TE) A specific communication device that connects to an *ISDN* network. Two TEs are referenced in the specification: *TE1* refers to an ISDN-compatible device (e.g., digital telephone or a computer with a built-in ISDN port), and *TE2* refers to a noncompatible ISDN device (e.g., an analog telephone or a computer without a built-in ISDN port).

Terminator A layer-1 device that prevents signal reflections by providing electrical resistance at the end of a cable to "absorb" signals to keep them from bouncing back and being heard again by the devices connected to the cable.

Thick Ethernet IEEE 802.3 10BASE5, which uses "thick" coaxial cable (outer diameter between 0.375 inch and 0.405 inch) as its physical medium.

Thin Ethernet IEEE 802.3 10BASE2, which uses "thin" coaxial cable (outer diameter between 0.175 inch and 0.195 inch) as its physical medium.

Threat Assessment An activity that involves determining how much security is necessary for proper control of system and network assets. Threat assessment is guided by answering the overriding question, "What assets are critical to the operation of my network and who do I think would want access to them?"

Throughput A realistic measure of the amount of data transmitted between two nodes in a given time period. It is a function of hardware/software speed, CPU power, overhead, and many other items. Compared to *bandwidth*, throughput is what the channel really achieves, whereas bandwidth is what is theoretically possible.

Time Division Multiple Access (TDMA) A *cellular communications network* access technology that partitions a cellular channel into three time slots, combines the corresponding digitized conversations into a single digital stream and then transports this stream on a single channel using the same frequency.

Time Division Multiplexing (TDM) A multiplexing technique that assigns to each node connected to a channel an identification number and a small amount of time in which to transmit. TDM-based transmissions are serially sequenced.

Token A special frame on a token ring or token bus network. Possession of the token permits a node to transmit data.

Token Bus A local area network technology based on a token passing protocol for media access. Defined in IEEE 802.4. A token bus network is characterized as a logical ring on a physical bus—physically, the network resembles a bus topology, but logically, the network is arranged as a ring with respect to passing the token from node to node.

Token Passing Protocol A network protocol that requires nodes to possess a special frame, called a *token*, prior to transmitting data. Token passing schemes are both contention-free and collision-free.

Token Ring A local area network technology based on a token passing protocol for media access control. Defined by IEEE 802.5. A token ring LAN is implemented either as a logical ring using a physical ring topology or as a logical ring structure arranged in a physical star configuration.

traceroute A UNIX program that depicts the gateways a packet transverses. A corresponding Microsoft NT command is called *tracert*.

Transceiver A service used in Ethernet/802.3 networks to connect nodes to the physical medium. Transceivers serve as both the physical connection and the electrical interface between a node and the physical medium, enabling the node to communicate with the medium. Transceivers transmit and receive signals simultaneously.

Transmission Control Protocol (TCP) A layer-4 connection-oriented protocol that performs several functions, including: providing for reliable transmission of data by furnishing end-to-end error detection and correction; guaranteeing that data are transferred across a network accurately and in the proper sequence; retransmitting any data not received by the destination node; and guaranteeing against data duplication between sending and receiving nodes. It is the TCP of *TCP/IP*.

Tree A network configuration in which nodes are connected to one another in a hierarchical fashion. A root node or hub is connected to second-level nodes or hubs; second-level devices are connected to third-level devices, which in turn are connected to fourth-level devices, and so forth.

Triple DES A variant of *DES* that uses three DES operations instead of one.

Tunneling See *Encapsulation*.

Twisted-Pair Cable A type of copper cable that uses at least two insulated copper wires that have been twisted together. There are two basic types: *unshielded twisted-pair (UTP)* and *shielded twisted-pair (STP)*.

UDP See *User Datagram Protocol*.

UDSL See *Universal Digital Subscriber Line*.

UMTS See *Universal Mobile Telephone Standard*.

UNI See *User-to-Network Interface*.

Unicast A data transmission that is destined to a single recipient.

Uniform Resource Identifier (URI) A conceptual entity used with *HTTP* to represent a universal set of names and addresses of all resources regardless of their location. URIs enable specific resources to be identified throughout the World Wide Web. They contain the name, location, or any other defining attribute of the specified resource. One example of a URI is a *URL*.

Uniform Resource Locator (URL) A specific type of *URI* that is used to identify a resource's location throughout the World Wide Web by specifying the access method needed to acquire the resource. The general form of this URL is http://*host*:*port*/*url-path*, where *host* is the fully qualified domain name of an Internet-connected node, *port* is the port number to which the TCP connection is made, and *url-path* provides the details of how the specified resource is accessed relative to the scheme being used.

Universal Digital Subscriber Line (UDSL) A *DSL* variant that provides symmetrical service at 2 Mbps each way.

Universal Mobile Telephone Standard (UMTS) A future-based *cellular communications network* that is based on a combination of *TDMA* and *CDMA* access technologies, supports data transfer rates of 2 Mbps, and operates at 2 GHz.

Unreliable Service A network service type that requires no acknowledgment of receipt of data from the receiving node to the sending node. This is called a *datagram service*.

Unshielded Twisted-Pair (UTP) Twisted-pair cable in which individual wire pairs are not shielded (i.e., protected from *noise*).

URI See *Uniform Resource Identifier*.

URL See *Uniform Resource Locator*.

User Agent An entity such as a Web browser through which HTTP client request messages are made to an HTTP server.

User Datagram Protocol (UDP) A connectionless protocol providing an unreliable datagram service. UDP does not furnish any end-to-end error detection or correction, and it does not retransmit any data it did not receive.

User-to-Network Interface (UNI) An end node's port where the *local loop* terminates at a customer's site.

Utilization A network performance measure that specifies the amount of time a LAN spends successfully transmitting data. *Average utilization* means that over some period of time (e.g., 10 hours), on average, a certain percentage of the LAN's capacity is used for successfully transmitting data. *Peak utilization* means that, at a specific moment in time, a certain percentage of the LAN's capacity was utilized.

UTP See *Unshielded Twisted-Pair.*

V.22 *bis* ITU-T standard for 2400-bps full-duplex modems; cycles to 1200 bps/600 bps.

V.29 ITU-T standard for 9600-bps facsimile service.

V.32 ITU-T standard for 9600-bps modems; cycles to 4800 bps when line quality degrades, and cycles forward when line quality improves.

V.32 *bis* ITU-T standard that extends *V.32* to 7200, 12,000, and 14,400 bps; cycles to lower rate when line quality degrades and cycles forward when line quality improves.

V.32 *ter* Pseudostandard that extends *V.32 bis* to 19,200 bps and 21,600 bps.

V.34 ITU-T standard for 28,800 bps modems. (*Note:* V.34 modems upgraded with special software can achieve data rates of 31,200 bps or 33,600 bps.)

V.FAST Proprietary pseudostandard from Hayes and Rockwell for modems transmitting at data rates up to 28,800 bps; served as a migration path for *V.34*.

V.42 ITU-T standard for modem *error correction*. Uses *LAPM* as the primary error-correcting protocol, with *MNP* Classes 1 through 4 as an alternative.

V.42 *bis* ITU-T standard that enhances *V.42* by incorporating the British Telecom Lempel Ziv data *compression* technique to V.42 *error correction*. Most *V.32, V.32 bis,* and *V.34* compliant modems come with V.42 or V.42 *bis* or MNP.

V.44 ITU-T standard that provides a 6:1 compression ratio. When compared to *V.42 bis*, download speeds can increase from 20% to 60%.

V.90 ITU-T standard for 57,600-bps modems (commonly called "56K modems") in which asymmetric data rates apply (i.e., the send and receive rates are different). Depending on telephone line conditions, upstream rates (send) are restricted to 33,600 bps, and downstream rates (receive) are restricted to 57,600 bps. V.90 modems are designed for connections that are digital at one end and have only two analog-digital conversions each way.

V.92 ITU-T standard that enhances *V.90* by increasing upstream rates to 48,000 bps, by decreasing connection times by one-half, and by supporting call-waiting.

vBNS+ See *Very High Performance Backbone Network Service*.

VCC See *Virtual Channel Circuit.*

VCI See *Virtual Channel Identifier.*

VDSL See *Very High-speed Digital Subscriber Line*.

Vectored Antenna An antenna used in *cellular communications networks* that subdivides the coverage area of a *cell*. A common subdivision is 120º, which means that a cell is partitioned into three vectors of 120º each. Each vectored region independently manages various frequency-related issues such as channel allocation and signal strength.

very High-Performance Backbone Network Service + (vBNS+) A nationwide network that supports high-performance, high-bandwidth applications. The vBNS+ began in 1995 as the vBNS, which was a restricted access collaborative networking project between the National Science Foundation and MCI WorldCom. Today, the vBNS+ operates as a new advanced networking service and is open to the entire U.S. higher education and research community. It has an OC-48 packet over *SONET* backbone that runs parallel with the original vBNS OC-12 *ATM* backbone. In addition to the advanced networking services it offers, the vBNS+ is also a development platform for next generation *Internet* applications.

Very High-Speed Digital Subscriber Line (VDSL) A *DSL* variant that provide asymmetric service over fiber. Downstream rates range from 13 to 52 Mbps; upstream rates range from 1.5 to 2.3 Mbps. Suitable for Internet/intranet access, video-on-demand, database access, remote LAN access, and high-definition TV.

Virtual Channel Connection (VCC) A virtual circuit that provides a logical connection between an *ATM* source and destination. Data can only be transmitted in one direction via a VCC. A VCC is denoted by a *virtual channel identifier (VCI)*, which is included as part of the ATM cell header. Multiple virtual channels that share the same connection can be packaged into a single virtual path.

Virtual Channel Identifier (VCI) A parameter used to identify *ATM virtual channels*. VCI information is carried within an ATM cell header.

Virtual Circuit A nondedicated connection through a shared medium that gives the high-level user the appearance of a dedicated, direct connection from the source node to the destination node.

Virtual Circuit Packet-Switching See *Packet-Switching*.

Virtual Local Area Network (VLAN) A network consisting of nodes that are not physically connected to the same medium. Instead, they are connected in a virtual sense using specially designed software that groups several ports in a switch into a single workgroup. Nodes connected to these ports are considered part of a workgroup, and network traffic from any node/port is (usually) limited to only those nodes or ports assigned to the workgroup

Virtual Path Connection (VPC) A semipermanent connection that provides a logical collection of *ATM virtual channels* that have the same endpoints. More specifically, a VPC carries a group of virtual channels, all of which have the same endpoints. Virtual paths enable any connection that uses the same network path from source to destination to be bundled into a single unit. A *virtual path identifier (VPI)* denotes a virtual path and is included in a cell's header. A virtual path can also provide a form of traffic control by logically (not physically) partitioning network traffic based on the type of data being carried and associated *quality of service*.

Virtual Path Identifier (VPI) A parameter used to identify an *ATM virtual path*. VPI information is carried within an ATM cell header.

Virtual Private Network (VPN) An *IP* connection between two sites over a public IP network that has its payload traffic encrypted so that only source and destination nodes can decrypt the traffic packets. A VPN enables a publicly accessible network to be used for highly confidential, dynamic, and secure data transmissions.

VLAN See *Virtual Local Area Network.*

VOFR See *Voice Over Frame Relay.*

VOIP See *Voice Over IP.*

Voice Over Frame Relay (VOFR) A technology that enables voice signals to be transmitted over a *frame relay* network.

Voice Over IP (VOIP) A technology that enables users to place telephone calls across the Internet.

VPC See *Virtual Path Connection.*

VPI See *Virtual Path Identifier.*

VPN See *Virtual Private Network.*

WAN See *Wide Area Network.*

WAP See *Wireless Application Protocol.*

Wavelength A measure of the length of a wave. It is the distance an electrical or light signal travels in one complete cycle.

Wavelength Division Multiplexing (WDM) A *multiplexing* method used with fiber-optic cables. Involves the simultaneous transmission of light sources over a single fiber-optic channel. Light sources of different wavelengths are combined by a WDM multiplexer and transmitted over a single line. When the signals arrive, a WDM demultiplexer separates them and transmits them to their respective destination receivers.

WCDMA See *Wideband Code Division Multiple Access*.

WDM See *Wavelength Division Multiplexing.*

Wearable Computing Coined at the Massachusetts Institute of Technology's Media Lab in the 1970s, the term today indicates a network convergence strategy in which computer technology is woven into clothing fabric. To date, a team has created a tie that doubles as a mobile telephone, with a tiny microphone lodged into the knot and a keypad woven into the silk.

Web-Based Networking A relatively new communications model that has emerged as a result of the Internet and which fosters the notion of *the network is the computer*, a phrase coined in the late 1980s by Sun Microsystems' president, Scott McNealy. "The network is the computer" implies that by making resources available to users via a network, the network essentially becomes the single, most powerful computer accessible. Thus, the network gives users access to more computing power than their desktop models.

Well-Known Port Number A port address that identifies the specific process or application a user accesses on a host. For example, *SMTP* is assigned well-known port number 25.

Wide Area Network (WAN) A network that interconnects computing resources that are widely separated geographically (usually over 100 km). This includes towns, cities, states, and countries. A WAN generally spans an area greater than 5 miles (8 km) and can be thought of as consisting of a collection of *LANs*.

Wideband Code Division Multiple Access (WCDMA) An enhancement to *CDMA* that provides a 2-Mbps peak data rate, operates on a 5 MHz channel, and supports both voice and data. WCDMA is sometimes referred to as Broadband CDMA (BCDMA).

Wire A general term for the physical layer of a network. The three main physical attributes of wire are *conductor, insulation,* and *outer jacket*. Wire also has three important electrical characteristics that can directly affect the quality of the signal transmitted across it: *capacitance, impedance,* and *attenuation*. Signal quality is affected most by the combination of attenuation and capacitance. The two primary forms of wire are copper and fiber. Also called *cable*.

Wireless Application Protocol (WAP) The de facto standard that enables wireless communications devices such as pagers, PDAs, and Web-enabled cell phones to provide Internet communications as well as advanced telephony services. WAP uses a special markup language called the Wireless Markup Language (WML).

Wireless Communications A type of communications in which signals travel through space instead of through a physical cable. There are two general types of wireless communication: *radio transmission* and *infrared transmission*.

Wireless LAN (WLAN) A *LAN* consisting of nodes that rely on *wireless communication* techniques for transmitting or receiving data. Specified by *IEEE 802.11.*

Wireless Local Loop (WLL) A wireless alternative to the traditional copper-based local loop. A WLL provides a wireless residential community network that completely bypasses the existing local loop cable infrastructure.

Wire Speed A unit of measure used to describe a device's maximum (i.e., fastest) filtering and forwarding rates. In 10-Mbps Ethernet/802.3, wire speed is equal to 14,880 frames per second. This is frequently reported as 14,880 packets per second.

WLAN See *Wireless LAN.*

WLL See *Wireless Local Loop*.

Workgroup Switch One application of an Ethernet switch. A workgroup switch partitions a single shared medium into multiple shared media and supports more than MAC address per port. Also called *segment switches*.

Workstation A computer system that has its own operating system and is connected to a network. A workstation can be a personal computer such as a Macintosh or Intel-based PC, a graphics workstation such as those manufactured by Sun Microsystems, a super-minicomputer such as IBM's AS/400, a super-microcomputer such as DEC's Alpha, or a mainframe such as an IBM ES-9000. Also called *host, server, desktop,* or *client.*

X.25 An early networking protocol that defines the interface between a public packet-switching data network and the device used to access this network.

XA-SMDS See *Exchange Access SMDS***.**

This Page Intentionally Left Blank

Bibliography

Chapter 1: Elements of Computer Networking

Aiken, R. J., and J. S. Cavallini. 1994. Standards—When Is It Too Much of a Good Thing? *Connexions: The Interoperability Report*, 8(8): 18–33.

Bradner, S. 1996. The Internet Standards Process—Revision 3. *RFC 2026*, October.

Brooks, J. 2000. A Primer on Components Necessary for Storage Area Networks. *Cisco World*, July, 31.

Comer, D. E. 1991. *Internetworking with TCP/IP Volume 1: Principles, Protocols, and Architecture.* Upper Saddle River, NJ: Prentice-Hall.

Conrad, J. W. 1988. Communications Standards Organizations. In *Handbook of Communication Systems Management,* ed. J. W. Conrad, 227–236. Boston: Auerbach Publishers.

Gallo, M., and W. Hancock. 2002. *Computer Communications and Networking Technologies.* Pacific Grove, CA: Brooks/Cole.

———, and R. Nenno. 1985. *Computers and Society with Basic and Pascal.* Boston: PWS.

Goldworm, B. 2000. The Difference Between SANs and NAs. [Electronic document: 2 October]; available from http://www.nwfusion.com/newsletters/stor/2000/1002stor1.html.

Hancock, B. 1989. *Network Concepts and Architectures.* Wellesley, MA: QED Information Sciences.

Kaufman, F. 1977. *Distributed Processing.* Newark: Coopers & Lybrand.

Martin, R. 1996. Intranets Taking over Communication Business. *Cisco World,* November, 12–13.

Milne, J. 1998. State of the NOS. *Network Computing,* 1 February, 50–77.

Molta, D. 1996. NC: Network Computer or No Contest. *Networking Computing,* 1 December, 153–154.

Schaeffer, H. 1987. *Data Center Operations: A Guide to Effective Planning, Processing, and Performance*. 2nd ed. Upper Saddle River, NJ: Prentice-Hall.

Stamper, D. A. 1991. *Business Data Communications*. Redwood City, CA: Benjamin/Cummings.

Stevens, W. R. 1990. *UNIX Network Programming.* Upper Saddle River, NJ: Prentice-Hall.

Tanenbaum, A. S. 1996. *Computer Networks*. 3rd ed. Upper Saddle River, NJ: Prentice-Hall.

Vaughn-Nichols, S. J. 1996. The NC Follies: A Network Computer Is a Small Idea. *Internet World,* December, 72–73.

Wittmann, A. 1997. Pssst, Buddy! Wanna Buy a Network Computer? *Network Computing,* 15 January, 117.

Woods, L. D. 1988. Management Issues in Distributed Data Processing. In *Handbook of Communication Systems Management*, ed. J. W. Conrad, 47–57. Boston: Auerbach Publishers.

Chapter 2: Network Topologies, Architectures, and the OSI Model

Abrams, M. D. 1982a. *Computer Networking Tutorial*. Rockville, MD: Computer Network Associates.

———. 1982b. *Data Communications Tutorial*. Rockville, MD: Computer Network Associates.

Ambort, D. 1990. Standards from Where? *LANtimes*, July, 61–64.

Clark, E. 1997. WDM Expands Fiber's Horizons. *LAN Magazine,* March, 67–71.

Conrad, J. W. 1988a. Open Systems Interconnection. In *Handbook of Communication Systems Management*, ed. J. W. Conrad. 237–251. Boston: Auerbach Publishers.

———. 1988b. Application of Data Communications Protocol Standards. In *Handbook of Communication Systems Management*, ed. J. W. Conrad. 253–266. Boston: Auerbach Publishers.

———. 1988c. Standards and Architecture for Local Area Networks. In *Handbook of Communication Systems Management*, ed. J. W. Conrad. 291–297. Boston: Auerbach Publishers.

Dyson, P. 1995. *The Network Press Dictionary of Networking.* 2nd ed. San Francisco: Sybex.

Feibel, W. 1995. *The Network Press Encyclopedia of Networking*. 2nd ed. San Francisco: Sybex.

Fisher, S. 1990. OSI Across the Water. *LANtimes*, July, 70.

Gallo, M., and W. Hancock. 2002. *Computer Communications and Networking Technologies*. Pacific Grove, CA: Brooks/Cole.

Hancock, B. 1989. *Network Concepts and Architectures*. Wellesley, MA: QED Information Sciences.

Higgins, K. J. 1994. How Evolution of TCP/IP Eclipsed OSI. *Open Systems Today*, 9 May, 54.

Malamud, C. 1991. TCP/IP—A Dependable Networking Infrastructure. *Networking Computing*, April, 84–86.

Martin, J., and K. Chapman. 1989. *Local Area Networks: Architectures and Implementations*. Upper Saddle River, NJ: Prentice-Hall.

McCloghrie, K., and M. T. Rose. 1994. Back to Basics: The Internet Transport Layer. *Connexions: The Interoperability Report*, June, 2–9.

Miller, M. A., 1992. *Troubleshooting TCP/IP: Analyzing the Protocols of the Internet*. San Mateo, CA: M&T Books.

Noor, A. I. 1988. Comparisons of Internetworking Architectures. In *Handbook of Communication Systems Management*, ed. J. W. Conrad. 267–276. Boston: Auerbach Publishers.

Postel, J. 1981a. Internet Protocol. *RFC 791*, September.

———. 1981b. Transmission Control Protocol. *RFC 793*, September.

Quarterman, J. S., and J. C. Hoskins. 1986. Notable Computer Networks. *Communications of the* ACM, 10, 932–971.

Spencer, L. T. 1990. Defining the OSI Model. *LANtimes*, September, 59.

Tanenbaum, A. S. 1996. *Computer Networks*. 3rd. ed. Upper Saddle River, NJ: Prentice-Hall.

Chapter 3: The Internet and TCP/IP

Albitz, P., and C. Liu. 1992. *DNS and BIND*. Sebastopol, CA: O'Reilly & Associates.

Auerbach, K., and C. Wellens. 1995. Internet Evolution or Revolution? *LAN*, October, 60–65.

Baker, S. 1993a. The Evolving Internet Backbone. *UNIX Review*, September, 15–21.

———. 1993b. Joining the Internet Club. *UNIX Review*, October, 23–33.

Banta, G. 1996. Internet Pipe Schemes. *Internet World,* October, 62–70.

Berners-Lee, T. 1994. Universal Resource Identifiers in WWW: A Unifying Syntax for the Expression of Names and Addresses of Objects on the Network as Used in the World-Wide Web. *RFC 1630,* June.

Berners-Lee, T., L. Masinter, and M. McCahill. 1994. Universal Resource Locators (URL). *RFC 1738*, December.

Berners-Lee, T., R. Fielding, and L. Masinter, 1998. Universal Resource Identifiers (URI): Generic Syntax. *RFC 2396*, August.

Borenstein, N. and N. Freed. 1993. MIME (Multipurpose Internet Mail Extensions) Part One: Mechanisms for Specifying and Describing the Format of Internet Message Bodies. *RFC 1521,* September.

Bradner, S. 1996. The Internet Standards Process—Revision 3. *RFC 2026*, October.

Bradner, S., and A. Mankin. 1995. The Recommendation for the IP Next Generation Protocol. *RFC 1752*, January.

Brown, R. 1993. TCP/IP: Reliable Glue for Enterprise Networks. *Networking Management*, June, 76–80.

Callon R. 1987. A Proposal for a Next Generation Internet Protocol. *Proposal to X3S3*, December.

Collinson, P. 1993. IP. *SunExpert Magazine*, October, 34–40.

Comer, D. E. 1991. *Internetworking with TCP/IP Volume 1. Principles, Protocols, and Architecture*. Upper Saddle River, NJ: Prentice-Hall.

Cooper, L. F. 1996. Controlled Chaos at Work—The Commercialization of the Internet. *Communications Week*, 1 April, issue 603.

Cray, A. 1998. Voice over IP: Here's How. *Data Communications*, April, 44–58.

Dutcher, W. 1997. IP Addressing: Playing the Numbers. *Data Communications*, 21 March, 69–74.

Farrow, R., and R. Power. 1998. Internet Security: Trouble in the Stack. *Network Magazine*, May, 95–96.

Fenton, B. 1997. Death of the Internet. *Popular Mechanics*, January, 41–44.

Fielding, R., J. Gettys, J. Mogul, H. Frystyk, L. Masinter, P. Leach, and T. Berners-Lee. 1999. Hypertext Transfer Protocol—HTTP/1.1. *RFC 2616*, June.

Freed, N., and N. Borenstein. 1996. Multipurpose Internet Mail Extensions (MIME) Part One: Format of Internet Message Bodies. *RFC 2045*, November.

Gallo, M., and W. Hancock. 2002. *Computer Communications and Networking Technologies*. Pacific Grove, CA: Brooks/Cole.

Gellens, R., C. Newman, and L. Lundblade. 1998. POP3 Extension Mechanism. *RFC 2449*, November.

Hall, E. 1999. Advanced TCP Options. *Networking Computing*, 8 February, 110–114.

Hall, E., and D. Willis. 1998. VOIP in the Enterprise. *Networking Computing*, 1 October, 41–62.

Held, G. 1997. IP for the Next Generation. *Network*, July, 65–70.

Hinden, R. M. 1996. IP Next Generation Overview. *Communications of the ACM,* 39 (6): 61–71.

Hinden, R., B. Carpenter, and L. Masinter. 1999. Format for Literal IPv6 Addresses in URL's. *RFC 2732*, December.

Hoffman, P., L. Masinter, and J. Zawinski. 1998. The mailto URL Scheme. *RFC 2368*, July.

Horak, R. 2000. *Communications Systems & Networks*. Foster City, CA: M&T.

Hudgins-Bonafield, C. 1995. How Will the Internet Grow? *Network Computing*, 1 March, 80–91.

———. 1996. The Prospect of IP Renumbering. *Network Computing*, 1 June, 84–92.

Huitema, C. 1994. The H Ratio for Address Assignment Efficiency. *RFC 1715*, November.

Hunt, C. 1992. *TCP/IP Network Administration*. Sebastopol, CA: O'Reilly & Associates, Inc.

Hurley, H. 1998. Value-Added ISPs. *Network Magazine*, 35–38.

Karvé, A. 1997. IP Multicast Streams to Life. *Network Magazine*, October, 53–58.

———. 1998. IP Telephony Finds Its Voice. *Network Magazine*, January, 57–61.

Kleinrock, L. 1992. Technology Issues in the Design of the NREN. In *Building Information Infrastructure*, ed. Brian Kahin. 174–198. New York: McGraw-Hill Primis.

Krol, E. 1993. FYI on "What Is the Internet?" *RFC 1462*, May.

Lange, L. 1996a. Internet Inc.—Bandwidth Becomes Big Business. *CMP Media, Inc.*, 1 June, issue 48.

Mandelbaum, R., and P. Mandelbaum. 1992. The Strategic Future of the Mid-level Networks. In *Building Information Infrastructure*, ed. Brian Kahin. 59–118. New York: McGraw-Hill Primis.

McCloghrie, K., and M. T. Rose. 1994. Back to Basics: The Internet Transport Layer. *Connexions: The Interoperability Report*, August, 2–9.

Meehan, T. 1997. Six Great Myths of IPv6. *Data Communications,* November, 140.

Miller, M. A. 1992. *Troubleshooting TCP/IP: Analyzing the Protocols of the Internet*. San Mateo, CA: M&T.

Moskowitz, R. G. 1995. Plan Now for the New Internet Protocol. *Network Computing,* 1 May, 144–150.

Myers, J., and M. Rose. 1996. Post Office Protocol—Version 3. *RFC 1939*, May.

Peterson, L., and B. Davie. 1996. *Computer Networks: A Systems Approach*. San Francisco: Morgan Kaufmann.

Postel, J. 1981a. Internet Protocol. *RFC 791*, September.

———. 1981b. Transmission Control Protocol. *RFC 793*, September.

———. 1982. Simple Mail Transfer Protocol. *RFC 821*, August.

Postel, J., and J. Reynolds. 1983. Telnet Protocol Specification. *RFC 854*, May.

———. 1985. File Transfer Protocol (FTP). *RFC 959*, October.

Reynolds, J, and J. Postel. 1994. Assigned Numbers. *RFC 1700*, October.

Shipley, G. 1997. Plumbing 101: Choosing the Right ISP. *Network Computing,* 1 March, 116–118.

Stallings, W. 2000. *Data and Computer Communications.* 6th ed. Upper Saddle River, NJ: Prentice-Hall.

Stevens, W. R. 1990. *UNIX Network Programming*. Upper Saddle River, NJ: Prentice-Hall.

Stone, M. 1991. Guide to TCP/IP Network Addressing. *LAN Technology*, April, 41–45.

Tanenbaum, A. S. 1996. *Computer Networks.* 3rd ed. Upper Saddle River, NJ: Prentice-Hall.

"The National Information Infrastructure: Agenda for Action." 1993. 58 Federal Regulation 49,025, 21 September.

Tristram, C. 1995. The Trouble with TCP/IP. *Open Computing*, February, 52–54.

Vaha-Sipila, A. 2000. URLs for Telephone Calls. *RFC 2806*, April.

Willis, D. 1996. Interchange Carriers As ISPs: Long-Distance Runaround. *Network Computing,* 1 September, 104–115.

———, and J. Milne. 1996. Domestic Internet Service Providers: Adapt or Perish. *Network Computing,* 1 June, 100–108.

Chapter 4: Physical Layer Concepts

Angel, J. 2000. Look Ma, No Cables. [Electronic document: 6 November]; available from http://www.networkmagazine.com/article/NMG20001106S0004.

Belanger, P. 2000. Wi-Fi Certification Helps Wireless LANs Come of Age. *Network World*, 21 August [Electronic document]; available from http://www.nwfusion.com/columnists/2000/0821belanger.html.

Breidenbach, S. 1990. Motorola Develops Wireless Network. *LANtimes,* 5 November, 15–16.

Chiquoine, W. A. 1997. Enterprise Cable Management. *Network Magazine,* July, 114–119.

Clark, E. 1998. Network Cabling's New High-wire Act. *Network Magazine,* March, 74–79.

Codenoll Technology Corporation. 1993. *The Fiber Optic LAN Handbook.* 5th ed. Yonkers, NY: Codenoll Technology Corporation.

Cray, A. 1997. Wiring for Speed, Playing for Time. *Data Communications,* April, 75–80.

Data.com. 2000. LANS Unplugged: Wireless LANs and IEEE 802.11. [Electronic document: 11 April]; available from http://www.networkmagazine.com/article/PIT20000410S0047.

Dyson, P. 1995. *The Network Press Dictionary of Networking*. 2nd ed. San Francisco: Sybex.

Feibel, W. 1995. *The Network Press Encyclopedia of Networking*. 2nd ed. San Francisco: Sybex.

Frank, A. 1994. Fiber Optics for Networks. *LAN Magazine*, November, 25–26.

———. 2000. HomeRF Fights for Its Place in the Sun. [Electronic document: 19 July]; available from http://www.networkmagazine.com/article/DCM20000719S0002.

———. 2000. FCC Spreads Joy for Frequency-Hoppers. [Electronic document: 1 September]; available from http://www.commweb.com/article/COM20001002S0003/3

———. 2000. Wireless Networking Looks to 5 GHz. [Electronic document: 3 October]; available from http://www.commweb.com/article/COM20000901S0007.

Gallo, M., and W. Hancock. 2002. *Computer Communications and Networking Technologies.* Pacific Grove, CA: Brooks/Cole.

Grier, J. 1995. The LAN Unleashed. *LAN Magazine,* September, 107–112.

Hancock, B. 1996. *Advanced Ethernet/802.3 Management and Performance.* Boston: Digital Press.

Henderson, T. 1995. Railroading the Category 5 Spec. *LAN Magazine,* October, 119–124.

Hudgins-Bonafield, C. 1998. Networking in the 21st Century: The Sky's the Limit. *Network Computing,* 15 March, 70–94.

Hume, B., and C. Ogden. 1993. Taming the Wire. *LAN Magazine,* September, 91–100.

Johnson, J. T. 1994. Wireless Data: Welcome to the Enterprise. *Data Communications,* 21 March, 42–55.

Jones, J. 1995. Cable Ready. *LAN Magazine,* September, 87–93.

Karvé, A. 1997a. Lesson 112: 802.11 and Spread Spectrum. *Network Magazine,* December, 25–26.

———. 1997b. The Wide World of Wireless. *Network Magazine,* December, 42–48.

———. 1999. Wave Division Multiplexing. Network Magazine, 1 August [Electronic document]; available from http://www.networkmagazine.com/article/NMG20000727S0010.

Keough, L. 1992. Premises Wiring: The Quiet Revolution. *Data Communications*, November, 103–115.

Kilarski, D. 1997. Satellite Networks: Data Takes to the Skies. *Network Magazine*, December, 52– 58.

Kim, D. 1994. Cable Ready. *LAN Magazine,* January, 83–90.

LAN Technology. 1990. Wireless Networking Gains Momentum: Vendors Exploit Different Technologies. *LAN Technology,* December, 25–28.

Leeds, F., and J. Chorey. 1991a. Cutting Cable Confusion: The Facts about Coax. *LAN Technology*, July, 31–49.

———. 1991b. Round up Your Cable Woes. *LAN Technology*, October, 39–56.

———. 1991b. Twisted-Pair Wiring Made Simple. *LAN Technology*, April, 49–61.

Makris, J. 1998. The Copper Stopper? *Data Communications*, March, 63–71.

Mara, F. 1996. Rewiring the Workplace. *LAN,* May, 101–107.

———. 1997. Testing High-performance Copper Cabling. *Network Magazine,* July, 127–131.

McMullen, M. 1995. Making Light of Data. *LAN Magazine*, September, 99–106.

Miller, K. 1994. Cellular Essentials for Wireless Data Transmission. *Data Communications,* 21 March, 61–67.

Molta, D. 1996. How Far Is It to 802.11 Wireless LANs? *Network Computing*, 1 June, 126–129.

———, and J. Linder. 1995. The High Wireless Act. *Network Computing*, 1 July, 82–100.

Morse, S. 1991. Sorting Out the Spaghetti: Ethernet Wiring Made Simple. *Network Computing*, October, 88–93.

Myers, T. 1997. Ensuring Proper Cable Installation. *Network Magazine*, July, 121–125.

Richardson, R. 1997. Home-style Wiring. *Network*, June, 95–99.

Sanders, R. 1992. Mapping the Wiring Maze. *LAN Technology,* 15 October, 27–36.

Saunders, S. 1992. Premises Wiring Gets the Standard Treatment. *Data Communications*, November, 105.

———. 1994. Bad Vibrations Beset Category 5 UTP Users. *Data Communications*, June, 49–53.

Schnaidt, P. 1994. Cellular Hero. *LAN Magazine*, April, 38–44.

Snider, M. 2001. Wireless World has Fewer Wide-Open Spaces. *USA Today*, 31 July, 3D.

Stallings, W. 2000. *Data and Computer Communications*. 6th ed. Upper Saddle River, NJ: Prentice-Hall.

Strizich, M. 1993. Networks Unplugged. *LAN Magazine*, December, 53–64.

Tanenbaum, A. S. 1996. *Computer Networks.* 3rd ed. Upper Saddle River, NJ: Prentice-Hall.

Tolly, K., and D. Newman. 1993. Wireless Internetworking. *Data Communications*, 21 November, 60–72.

Wittman, A. 1994. Will Wireless Win the War? *Networking Computing*, 1 June, 58–70.

———. 1997. Earthquake or a New IEEE Standard? *Network Computing*, 15 August, 37.

Chapter 5: Data Link Layer Concepts and IEEE Standards

Baker, F. 2000. QoS: Establishing Business Priorities. *Packet: Cisco Systems Users Magazine*, Fourth Quarter, 61–67.

Campbell, R. 1994. The Last Word on Thick Ethernet. *LAN Technology*, January, 26–27.

Codenoll Technology Corporation. 1993. *The Fiber Optic LAN Handbook*. 5th ed. Yonkers, NY: Codenoll Technology Corporation.

Dyson, P. 1995. *The Network Press Dictionary of Networking*. 2nd ed. San Francisco: Sybex.

Feibel, W. 1995. *The Network Press Encyclopedia of Networking*. 2nd ed. San Francisco: Sybex.

Gallo, M., and W. Hancock. 2002. *Computer Communications and Networking Technologies*. Pacific Grove, CA: Brooks/Cole.

Grigonis, R. 2001. Working the QoS Puzzle. *ComputerTelephony.com,* January, 68–82.

Hancock, B. 1988. *Designing and Implementing Ethernet Networks*. Wellesley, MA: QED Information Sciences, Inc.

———. 1996. *Advanced Ethernet/802.3 Management and Performance*. Boston: Digital Press.

Hunt, C. 1995. *Networking Personal Computers*. Sebastopol, CA: O'Reilly & Associates.

Martin, J. 1989. *Local Area Networks: Architectures and Implementations*. Upper Saddle River, NJ: Prentice-Hall.

Metcalfe, R., and D. Boggs. 1976. Ethernet: Distributed Packet Switching for Local Computer Networks. *Communications of the ACM*, 19(7), 395–404.

Peterson, L., and B. Davie. 1996. *Computer Networks: A Systems Approach*. San Francisco: Morgan Kaufmann.

Stallings, W. 1997. *Local and Metropolitan Area Networks*. 5th ed. Upper Saddle River, NJ: Prentice-Hall.

———. 2000. *Data and Computer Communications*. 6th ed. Upper Saddle River, NJ: Prentice-Hall.

Tanenbaum, A. S. 1996. *Computer Networks*. 3rd ed. Upper Saddle River, NJ: Prentice-Hall.

The AG Group. 1993. *Etherpeek Network Analysis Software User Manual Version 2.0*. Walnut Creek, CA: The AG Group.

Chapter 6: Network Hardware Components (Layers 1 and 2)

Baker, F. 1995. Switching Gears. *LAN Interoperability: A Quarterly Supplement to LAN Magazine*, May, 45–52.

Bay Networks. 1996. *Fundamentals of Switching: A Guide to Workgroup Networking*. Santa Clara, CA: Bay Networks.

Boardman, B. 1996. Good Things Come in Small Packages: Combo PC Cards. *Network Computing*, 1 April, 108–116.

Breyer, R., and S. Riley. 1996. *Switched and Fast Ethernet*. 2nd ed. Emeryville, CA: Ziff-Davis.

Chipcom. 1993. *Network Switching Solutions*. Southborough, MA: Chipcom.

Dyson, P. 1995. *The Network Press Dictionary of Networking*. 2nd ed. San Francisco: Sybex.

Feibel, W. 1995. *The Network Press Encyclopedia of Networking*. 2nd ed. San Francisco: Sybex.

Feltman, C. 1994. Sizzling Switches. *LAN Magazine*, February, 115–128.

Gallo, M., and W. Hancock. 2002. *Computer Communications and Networking Technologies*. Pacific Grove, CA: Brooks/Cole.

Gerber, B. 1994. More PCMCIA Slots, Please. *Network Computing*, 1 August, 161–162.

Gervais, J. 2000. Smart NICs Preserve CPU Cycles. [Electronic document: 11 December]; available from http://www.nwfusion.com/news/tech/2000/1211tech.html.

Higgins, K. J. 1994. Switching Hubs: Rising Network Stars. *Open Systems Today*, 20 June, 50–58.

Kohlhepp, R. J. 1996. Stacking Managed Hubs in Your Favor. *Network Computing*, October, 112–115.

Krivda, C. 1995. Another Swipe at Switching. *LAN*, October, 129–134.

Magidson, S. 1994. PCMCIA: Finally, Standards Bring Compatible Products. *Network Computing*, 15 February, 124–127.

Majkiewicz, J. 1993. Switching—Everybody's Doing It. *SunExpert*, September, 48–63.

McMullen, M. 1993. Using PCMCIA Cards and Wireless Connectivity, PDAs May Become Handy. *LAN Magazine*, December, 40–48.

Phipps, M. 1999. Understanding the Basics: A Guide to LAN Switch Architecture and Performance. *Packet: Cisco Systems Users Magazine*, Fourth Quarter, 54–57.

Schnaidt, P. 1994. Switch Hunt. *LAN Magazine*, June, 75–84.

Sharer, R. 1995. A Switch in Time. *LAN Magazine*, May, 109–114.

Steinke, S. 1995. Ethernet Switching. *LAN Magazine*, March, 25–26.

Tristram, C. 1994. PCMCIA on the Desktop. *Open Computing*, December, 81–82.

Ungermann-Bass Networks. 1995. *LAN Switching Buyer's Guide*. Santa Clara, CA: Ungermann-Bass Networks.

Wittman, A. 1994. The Switch. *Network Computing*, 1 September, 62–72.

Wobus, J. 1993. When to Avoid Adding Ethernet Hubs. *Network Computing*, January, 154–157.

Wong, H. H. 1996. Increase Ethernet Performance with Switched Technology. *Technical Support*, June, 37–43.

Chapter 7: WANs, Internetworking, and Network Layer Concepts and Components

Anderson, P., and G. James. 1998. Switching at Layer 4. *Network World*, 20 July.

Ascend Communications, Inc. 1998. *Virtual Private Networks for the Enterprise: A Resource Guide for Information Technology and Network Managers Worldwide*. Alameda, CA: Ascend Communications.

Aziz, Z. 2000. Going Connectionless. *Packet: Cisco Systems Users Magazine*, Fourth Quarter, 78–82.

Baker, F. 1994. OSPF Fundamentals. *LAN Magazine*, December, 71–78.

Baker, S. 1994. Attaining Gigabit Speed. *UNIX Review*, 17–26.

Callon, R., V. Haimo, and M. Lepp. 1989. Routing in an Internetwork Environment. *Connexions: The Interoperability Report*, 3(8): 2–7.

Coltun, R. 1989. OSPF: An Internet Routing Protocol. *Connexions: The Interoperability Report*, 3(8): 19–25.

Conover, J. 1998. Slicing Through IP Switching. *Network Computing*, 15 March, 50–67.

Davie, B., P. Doolan, and Y. Rekhter. 1998. *Switching in IP Networks: IP Switching, Tag Switching, and Related Technologies*. San Francisco: Morgan Kaufmann.

Davie, B. 1999. Multiprotocol Label Switching. *Packet: Cisco Systems Users Magazine*, Second Quarter, 23–27.

Dern, D. P. 1990. Standards for Interior Gateway Routing Protocols. *Connexions: The Interoperability Report*, 4(7): 2–10.

Dickson, K. 1999. The ABCs of VPNs. *Packet: Cisco Systems Users Magazine*, Fourth Quarter, 50–53.

Enger, B. 1989. Adopting a Gateway. *Connexions: The Interoperability Report*, 3(8): 32–37.

Fitzgerald, S., and L. Bicknell. 1998. Troubleshooting High-Speed WANs. *Network Magazine*, 15 January, 87–91.

———, and E. Greenberg. 1994. Ready for Takeoff. *LAN Magazine*, May, 97–104.

Gallo, M., and W. Hancock. 2002. *Computer Communications and Networking Technologies*. Pacific Grove, CA: Brooks/Cole.

Garcia-Luna-Aceves, J. J. 1989. Loop-free Internet Routing and Related Issues. *Connexions: The Interoperability Report*, 3(8): 8–18.

Gaw, S. 1998. Building the Universal Backbone. *Network Magazine*, 15 January, 74–79.

Geier, J. 1996. Choose Your Own Route. *LAN*, September, 103–108.

Grimes, R. 2001. VPN Standards Mean More Choices, Less Cost. *Cisco World*, May, 17–18.

Hagens, R. 1989. Components of OSI: ES-IS Routing. *Connexions: The Interoperability Report*, 3(8): 46–51.

Hall, E. 1998. Implementing Prioritization on IP Networks. *Network Computing*, 15 August, 76–79.

Hantula, D. 2000. Managing Mixed Environments. *Packet: Cisco Systems Users Magazine*, Fourth Quarter, 74–77.

Heywood, P. 1998. A Switch in Plans. *Data Communications*, July, 25–26.
IPSec Tactics. 2000. *Packet: Cisco Systems Users Magazine*, First Quarter, 81–82.
Karvé, A. 1997. IP Switching Gets Focused. *Network Magazine*, June, 81–85.
Lepp, M. 1989. The IETF Open Routing Working Group. *Connexions: The Interoperability Report*, 3(8): 26–31.
Lewis, C. 1996a. Alternatives to RIP in a Large Internetwork. *Network Computing*, 1 October, 126–131.
———. 1996b. Should RIP Finally Rest in Peace? *Network Computing*, 1 September, 124–128.
Marsan, C. D. 2001. Experts Call MPLS Bad for 'Net. *Network World*, 6 August Electronic document]; available from http://www.nwfusion.com/news/2001/0806mpls.html.
McQuillan, J. 1997. Routers and Switches Converge. *Data Communications*, 21 October, 120–124.
Moy, J. 1990. OSPF: Next Generation Routing Comes to TCP/IP Networks. *LAN Technology,* April, 71–79.
MPLS and IPSec. 2001. *Packet: Cisco Systems Users Magazine*, First Quarter, 37–43.
Newman, D., T. Giorgia, and F. Yavari-Issalou. 1998. VPNs: Safety First, but What About Speed? *Data Communications,* July, 59–67.
Osborne, E. 2000. Express Delivery. *Packet: Cisco Systems Users Magazine*, Fourth Quarter, 83–87.
Periera, R. 2000. IPSec Makes Policy, Interoperability Strides. *Packet: Cisco Systems Users Magazine*, First Quarter, 69.
Peterson, L., and B. Davie. 1996. *Computer Networks: A Systems Approach*. San Francisco: Morgan Kaufmann.
Roberts, E. 1997. IP on Speed. *Data Communications*, March, 84–96.
Scholl, F. 1997. Upgrading Your Backbone. *Network Magazine,* May, 95–99.
Staats, R. 2001. Making A Case for MPLS VPNs Over Traditional VPNs. *Cisco World*, April, 12–13.
Stallings, W. 1996. Make Way for the Hot WANs. *LAN,* January, 48–54.
———. 2000. *Data and Computer Communications*. 6th ed. Upper Saddle River, NJ: Prentice-Hall.
Steinke, S. 1996a. Getting Data over the Telephone Line: Lesson 92: CSUs and DSUs. *LAN,* April, 27–28.
———. 1996b. The Internet as Your WAN. *LAN*, October, 47–52.
———. 1997. Lesson 105: Switching vs. Routing. *Network Magazine*, May, 27–28.
———. 1998. What Is a Switch Today? *Network Magazine*, January, 136.
Tanenbaum, A. S. 1996. *Computer Networks.* 3rd ed. Upper Saddle River, NJ: Prentice-Hall.
The Inner Workings of MPLS. 2000. *Packet: Cisco Systems Users Magazine*, First Quarter, 83–87.
Trudel, B. 2000. Tunnel Vision. *Packet: Cisco Systems Users Magazine,* Fourth Quarter, 68–73.
Tsuchiya, P. 1989. Components of OSI: Routing (An Overview). *Connexions: The Interoperability Report*, 3(8): 38–45.
Welcher, P. J. 2000. Introduction to MPLS: An Evolution from Tag Switching. *Cisco World*, September, 42–44.
Zines, E. 2000. VPN Mania. *Packet: Cisco Systems Users Magazine*, First Quarter, 56–61.

Chapter 8: Ethernet, Ethernet, and More Ethernet

10G: It's Here. 2001. *Packet: Cisco Systems Users Magazine*, Second Quarter, 42–47.

Axner, D. 2000. Gigabit Ethernet Goes On MAN Hunt. *Network World*, 4 December [Electronic document]; available from http://www.nwfusion.com/edge/research//2000/1204edgefeat.html.

Baldwin, C. 1994. *High Speed Networking Technologies: Options and Implications.* Southborough, MA: Chipcom Corporation.

Bolles, G. 1991. A Guided Tour of 10BASE-T. *Network Computing* (fall): 16–22.

Breyer, R., and S. Riley. 1996. *PC Week Switched and Fast Ethernet.* 2nd ed. Emeryville, CA: Ziff-Davis.

Chae, L. 1995. Tutorial—Lesson 88: Fast Ethernet. *LAN Magazine,* December, 27–28.

Clark, E. 1998a. Gigabit Ethernet Goes High Gear. *Network Magazine,* February, 61–65.

———. 1998b. The Gigabit Gambit. *Network Magazine,* September, 48–53.

Cohen, R. 1996a. A Comparison of the Performance of 100BASE-TX and 100VG Networks. *Connexions: The Interoperability Report*, 10(12): 16–24.

———. 1996b. 100VG and 100BASE-T Tutorial. *Connexions: The Interoperability Report*, 10(11): 25–27.

Conover, J. 1997. The Road to Fast Ethernet Networks. *Network Computing,* 15 February, 146–149.

———. 1998. Building a Better Ethernet Infrastructure. *Network Computing,* 1 October, 88–94.

Costa, L. F. 1994. *Planning and Designing High Speed Networks Using 100VG-AnyLAN.* Upper Saddle River, NJ: Prentice-Hall.

Dornan, A. 2000. Emerging Technology: Ethernet Makes a Run for the WAN. *Network Magazine*, 5 October [Electronic document]; available from http://www.networkmagazine.com/article/NMG20001004S0002.

Ethernet on the Move. 2000. *Packet: Cisco Systems Users Magazine*, Fourth Quarter, 115.

Extreme Networks. 2001. Looking at the World Through Ethernet-Colored Glasses: The Rosy Promise of 10 Gigabit Ethernet. [Electronic document]; available from http://www.extremenetworks.com.

Feltman, C. 1996a. Ethernet Earns Its Wings. *LAN Interoperability: Quarterly Supplement,* May, 14–20.

———. 1996b. Virtual LAN, Real Performance. *LAN,* January, 67–72.

Flynn, D., and T. Perkinson. 1994. Fast Ethernet vs. AnyLAN: Degrees of Separation. *Data Communications*, May, 158.

Fogle, D. 1996. Tutorial—Lesson 90: Ethernet Frame Types. *LAN Magazine,* February, 27–28.

Frank, A. 1995. Flocking to a Faster Network. *LAN Magazine*, August, 67–77.

———. 1996. Multimedia LANs and WANs. *LAN Magazine,* July, 81–86.

———, and D. Fogle. 1996. 100VG-AnyLAN's High-Speed Hopes. *LAN Magazine*, January, 128–133.

Fratto, M. 1997. Unlocking Virtual Private Networks. *Network Computing,* 1 November, 53–78.

Gallo, M., and W. Hancock. 2002. *Computer Communications and Networking Technologies.* Pacific Grove, CA: Brooks/Cole.

Geier, J. 1996. Up Close and Personal. *LAN Interoperability: Quarterly Supplement,* May, 39–45.

Gigabit Ethernet Alliance. 1996. *Gigabit Ethernet: White Paper,* August. Gigabit Ethernet Alliance.

———. 1999. Running 1000BASE-T: Gigabit Ethernet over Copper. *Gigabit Ethernet: White Paper,* September. Gigabit Ethernet Alliance.
Gigabit Ethernet Meets ATM. 2000. *Packet: Cisco Systems Users Magazine*, Second Quarter, 20–21.
Gohn, B. 1997a. Policy-based Services for Ethernet Networks. In Special Supplement by 3Com Corporation. *LAN,* January, 8–9.
———. 1997b. Switching and Routing—And Everything in Between. In Special Supplement by 3Com Corporation. *LAN,* January, 10–12.
Goralski, W., and G. Kessler. 1996. Changing Channels. *LAN Interoperability: Quarterly Supplement,* May, 6–12.
Greenstein, I. 1990. Fiber-optic LANs Improve in Price and Performance. *Network Management,* June, 84–89.
Hancock, B. 1988. Designing and Implementing Ethernet Networks, Wellesley, MA: QED Information Sciences.
Hancock, B. 1992. Advanced Ethernet/802.3 Management & Performance. Boston: Digital Press.
Hall, E. 1998. Bringing Prioritization Services to Ethernet. *Network Computing,* 1 August, 94–100.
Held, G. 1996. The LAN Guessing Game. *LAN Magazine,* May, 93–100.
———. 1999. Avoiding High-Speed LAN Potholes. *Network Magazine,* January, 68–71.
Henderson, T. 1995. Flooring It with Ethernet. *LAN Magazine*, March, 55–62.
———. 1998. Gigabit Ethernet Blueprint. *Network Magazine,* September, 42–46.
Hindin, E. 1998. Say What? QoS in English. *Network World,* 17 August.
Johnson, J. 1995. The Need for Speed. *InformationWeek*, 23 October, 36–48.
Karvé, A. 1996. Multimedia Takes the Stage. *LAN Magazine,* May, 123–127.
———. 1997. Ethernet's Next Frontier. *LAN,* January, 40–47.
———. 1998. Pushing Beyond Gigabit Ethernet. *Network Magazine,* September, 54–58.
———. 1999. Gigabit Ethernet Gets a New Look. *Network Magazine,* August, 36–42.
Kessler, G. 1993. Changing Channels. *LAN Magazine,* December, 69–84.
Krivda, C. 1995. Another Swipe at Switching. *LAN,* October, 129–134.
Lippis, N. 1996. Gigabit Ethernet Starts to Sizzle. *Data Communications,* November, 31–32.
Lounsbury, A. 1997. Gigabit Ethernet: The Difference Is in the Details. *Data Communications,* May, 75–80.
Makris, J. 1998. The Copper Stopper? *Data Communications,* March, 63–73.
Mandeville, R., and D. Newman. 1997. VLANs: Real Virtues. *Data Communications,* May, 83–91.
———, and D. Shah. 1998. Gigabit Ethernet Gets It Done. *Data Communications,* February, 66–81.
Metcalfe, R., and D. Boggs. 1976. Ethernet: Distributed Packet Switching for Local Computer Networks. *Communications of the ACM*, 19(7): 395–404.
Minoli, D. 1993. Isochronous Ethernet: Poised for Launch. *Network Computing,* August, 156–161.
Molta, D. 1996. The Ten Greatest Networking Myths. *Networking Computing,* 1 September, 121–122.
Pieper, K., and R. Fowler. 1997. The Next Step in the Ethernet Evolution. In Special Supplement by 3Com Corporation. *LAN,* January, 1–2.
Richardson, R. 1997. VPNs: Just Between Us. *LAN,* February, 99–103.
Podzimek, C. Beware of Collisions: An Ethernet Primer. Cisco World, August, 23–29.

Roberts, E. 1996. Weighed Down by Doubts: Is the Proposed High-Speed Spec Too Good to Be True? *Data Communications,* November, 55–58.
———. 1997. Gigabit Ethernet: Fat Pipe or Pipe Bomb? *Data Communications,* May, 58–72.
Saunders, S. 1994. Full-Duplex Ethernet: More Niche Than Necessity? *Data Communications,* March, 87–92.
Schnaidt, P. 1994a. Plug in at 100. *LAN Magazine,* March, 71–79.
———. 1994b. Tutorial—Lesson 73: Which Fast LAN? *LAN Magazine,* September, 25–26.
———. 1996. Is Gigabit Ethernet the Next Miracle Cure? *Network Computing,* 15 October, 33.
Scholl, F. 1993. 10BASE-F Stretches the Ethernet Backbone. *Data Communications,* October, 103–108.
Shimada, K. 1994. Fast Talk About Fast Ethernet. *Data Communications,* 21 March, 21–22.
Shipley, B. 1996. Ethernet's Endurance Contest. *LAN Magazine,* May, 67–72.
Skorupa, J., and G. Prodan. 1997. Battle of the Backbones: ATM vs. Gigabit Ethernet. *Data Communications,* April, 87–89.
Snyder, J. 1996. Ethernet in the Fast Lane. *MacWorld*, March, 128–133.
Stallings, W. 2000. *Data and Computer Communications.* 6th ed. Upper Saddle River, NJ: Prentice-Hall.
Steinke, S. 1995a. Ethernet Switching. *LAN Magazine,* March, 25–26.
———. 1995b. The State of the Ethernet. *LAN Magazine,* March, 44–52.
Tolley, B. 1997a. Standards Emerging for Gigabit Ethernet. In Special Supplement by 3Com Corporation. *LAN,* January, 3–4.
———. 1997b. The Lowdown on High Speed: Gigabit Ethernet and ATM. In Special Supplement by 3Com Corporation. *LAN,* January, 5–7.
Tolly, K. 1994. Full Speed Ahead for Full-Duplex Ethernet. *Data Communications,* March, 39–40.
Ungermann-Bass Networks. 1995. *LAN Switching Buyer's Guide.* Santa Clara, CA: Ungermann-Bass Networks.
Wittmann, A. 1993. Fast Ethernet—The Curtain Rises on a New Debate. *Network Computing,* May, 18–22.
———. 1994. Faster Anyone? HP's 100VG-AnyLAN. *Network Computing,* 1 October, 47–49.
Wobus, J. 1993. When to Avoid Adding Ethernet Hubs. *Network Computing,* January, 154–157.
Wong, H. H. 1996. Increase Ethernet Performance with Switched Technology. *Technical Support,* June, 37–43.
Zeichick, A. 1997. Glossary: IEEE 802. *Network Magazine,* November, 28.

Chapter 9: Token Ring

Baldwin, C. 1994. *High Speed Networking Technologies: Options and Implications.* Southborough, MA: Chipcom Corporation.
Feibel, W. 1995. *The Network Press Encyclopedia of Networking.* 2nd ed. San Francisco: Sybex.
Frank, A. 1995. Flocking to a Faster Network. *LAN Magazine,* August, 67–77.
———. 1996. Multimedia LANs and WANs. *LAN Magazine,* July, 81–86.
Gallo, M., and W. Hancock. 2002. *Computer Communications and Networking Technologies.* Pacific Grove, CA: Brooks/Cole.
Gaw, S. 1996. Token Ring Flips the Switch. *LAN,* July, 67–72.
Goralski, W., and G. Kessler. 1996. Changing Channels. *LAN Interoperability: Quarterly Supplement,* May, 6–12.

Held, G. 1996. The LAN Guessing Game. *LAN Magazine,* May, 93–100.

Karvé, A. 1995. Lord of the Ring. *LAN Magazine,* August, 107–114.

———. 1996. Multimedia Takes the Stage. *LAN Magazine,* May, 123–127.

Krivda, C. 1995. Another Swipe at Switching. *LAN,* October, 129–134.

Lippis, N. 1997. The Token Ring Trap. *Data Communications,* February, 23–24.

Love, R. D. 1998. The Fast Track to High-Speed Token Ring. *Data Communications,* February, 95–100.

Martin, J. 1989. *Local Area Networks: Architectures and Implementations.* Upper Saddle River, NJ: Prentice-Hall.

Roberts, E. 1998. A Slow Start for High-Speed Token Ring. *Data Communications,* February, 85–92.

Stallings, W. 2000. *Data and Computer Communications.* 6th ed. Upper Saddle River, NJ: Prentice-Hall.

Tanenbaum, A. S. 1996. *Computer Networks.* 3rd ed. Upper Saddle River, NJ: Prentice-Hall.

Ungermann-Bass Networks. 1995. *LAN Switching Buyer's Guide.* Santa Clara, CA: Ungermann-Bass Networks.

Chapter 10: Fiber Distributed Data Interface (FDDI)

American National Standard. 1986a. FDDI Token Ring Physical Layer Medium Dependent PMD. *American National Standard,* ASC X3T9.5, February, rev. 7.

———. 1986b. FDDI Token Ring Media Access Control MAC. *American National Standard,* ASC X3T9.5*,* July, rev. 10.

———. 1986c. FDDI Token Ring Station Management SMT. *American National Standard,* ASC X3T9.5, September, rev. 12.

———. 1986d. FDDI Token Ring Physical Layer Protocol PHY. *American National Standard,* ASC X3T9.5, October, rev. 14.

Burr, W. 1986. The FDDI Optical Data Link. *IEEE Communications Magazine,* May.

Cooper. S. 1989. Joining the Next LAN Generation. *UNIX Review,* 72: 48–59.

Daniels, G. 1990. Implementing and Managing FDDI Networks. *Networking Management,* November, 56–61.

Gallo, M., and W. Hancock. 2002. *Computer Communications and Networking Technologies.* Pacific Grove, CA: Brooks/Cole.

Hancock, B. 1989. *Network Concepts and Architectures.* Wellesley, MA: QED Information Sciences.

Joshi, S. 1986. High-performance Networks: A Focus on the Fiber Distributed Data Interface FDDI Standard. *IEEE Micro,* June.

Ross, F. 1986. FDDI—A Tutorial. *IEEE Communications Magazine,* May.

Sterling, D. 1987. *Technicians Guide to Fiber Optics.* Albany, NY: Delmar Publishers.

The Institute of Electrical and Electronics Engineers. 1985a. Logical Link Control. *American National Standard ANSI/IEEE Std. 802.2.* The Institute of Electrical and Electronics Engineers.

———. 1985b. Token Ring Access Method and Physical Layer Specifications. *American National Standard ANSI/IEEE Std. 802.5.* The Institute of Electrical and Electronics Engineers.

Wolter, M. S. 1990. Fiber Distributed Data Interface FDDI—A Tutorial. *Connexions: The Interoperability Report*, 4(10): 16–26.

———. 1991. ANSI x3T9.5 Update: Future FDDI Standards. *Connexions: The Interoperability Report*, 5(10): 21–23.

Chapter 11: Integrated Services Data Network (ISDN)

Alvich-LoPinto, M. 1990. Piecing Together the ISDN Puzzle. *Today's Office,* October, 21–23.

Atkins, J., and M. Norris. 1995. *Total Area Networking: ATM, Frame Relay and SMDS Explained.* New York: Wiley.

Bates, R. J., and D. Gregory. 1998. *Voice and Data Communications Handbook.* New York: McGraw-Hill.

Birenbaum, E. 1994. A Trio of Access Devices Aggregates ISDN. *Data Communications,* June, 147–148.

Brown, D. 1996. Blazing with Inverse Multiplexed ISDN. *Network Computing,* 1 December, 164–167.

Crane, E., and R. Raucci. 1995. Going Digital. *Open Computing,* February, 93–94.

Dortch, M. 1990. UNIX and ISDN: Partners on the Same Line. *UNIXWorld,* 55–59.

Durr, M. 1994. ISDN Reemerges. *LAN Magazine,* January, 103–112.

Fitzgerald, S. 1994. Global Warming: ISDN Heats Up. *LAN Magazine,* June, 50–55.

Fogle, D., and A. Frank. 1996. The Ascent of ISDN. *LAN*, March, 120–125.

Frank, A. 1995. Lesson 82: The ISDN Connection. *LAN Magazine,* June, 21–22.

———. 1998. Lesson 117: Always On/Dynamic ISDN (AO/DI). *Network Magazine,* April, 23–24.

Gallo, M., and W. Hancock. 2002. *Computer Communications and Networking Technologies.* Pacific Grove, CA: Brooks/Cole.

Hurwicz, M. 1996. On-Line from a LAN. *LAN,* May, 85–90.

Johnston-Turner, M. 1991. Evaluating ISDN Options. *Network World,* 18 March, 40.

Leeds, F. 1996. Plugging in to ISDN. *LAN,* April, 93–96.

Levitt, J. 1993. Speed Is Only One Benefit of Installing an ISDN Link. *Open Systems Today,* 23 January, 1.

Miller, K. 1994. ISDN and Internetworking: Made for Each Other? *Data Communications,* February, 23–24.

Moskowitz, R. 1996. Bracing for the Last and Longest Mile Home. *Network Computing,* 1 April, 35–36.

Parnell, T. 1997. *LAN Times Guide to Wide Area Networks.* Berkeley, CA: Osborne McGraw-Hill.

Petrosky, M. 1992. Momentum Builds for ISDN as a Data Pipe. *LAN Technology,* August, 15–17.

Quiat, B. 1994. V.FAST, ISDN or Switched 56: Which Remote Solution is the Right One? *Network Computing,* 1 March, 70–87.

Robertson, B. 1993. Telecommuting via ISDN: Out of Sight, Not Out of Range. *Network Computing,* July, 134–138.

———. 1994. Choose an ISDN Flavor: NIC or Digital Modem. *Network Computing,* 1 September, 174–176.

———, and J. Newman. 1995. ISDN Does It All! *Network Computing,* 1 May, 62–80.

Schnaidt, P. 1993. Lesson 55: An ISDN Issue. *LAN Magazine,* March, 24–25.

Stallings, W. 1991. Broadband ISDN: A Standards Update. *Network World,* 18 March, 49–52.

———. 1997. *Data and Computer Communications.* 5th ed. Upper Saddle River, NJ: Prentice-Hall.

Ubois, J. 1993. The Big Pipe. *SunWorld,* November, 86–91.

Willis, D., and J. Newman. (1996). ISDN: Prime for the Enterprise. *Network Computing,* 15 May, 45–65.

Wrobel, L. 1997a. ISDN: Innovations Subscribers Do Need—Part I. *Technical Support,* April, 41–44.

———. 1997b. ISDN: Innovations Subscribers Do Need—Part II. *Technical Support,* May, 34–38.

Chapter 12: Frame Relay

Atkins, J., and M. Norris. 1995. *Total Area Networking: ATM, Frame Relay and SMDS Explained.* New York: Wiley.

Bates, R. J., and D. Gregory. 1998. *Voice and Data Communications Handbook.* New York: McGraw-Hill.

Brown, D., and D. Willis. 1998. Videoconferencing on Frame Relay Networks. *Network Computing,* 15 September, 47–62.

Clark, E. 1998. Frame Relay Goes Public. *Network Magazine,* April, 68–73.

Feibel, W. 1995. *The Network Press Encyclopedia of Networking.* 2nd ed. San Francisco: Sybex.

Fitzgerald, S., and L. Kraft. 1993. A New Model for Frame Relay. *LAN Magazine,* August, 61–67.

———. 1993. Outfit Your WAN with Frame Relay. *LAN Magazine,* August, 52–59.

———. 1995. Designing Your Frame Relay Network. *Network Computing,* 15 January, 152–154.

Gallo, M., and W. Hancock. 2002. *Computer Communications and Networking Technologies.* Pacific Grove, CA: Brooks/Cole.

Gareiss, R. 1996. Don't Take No for an Answer. *Data Communications,* November, 67–82.

———. 1997. Frame Relay vs. IP: It's Your Move. *Data Communications,* February, 89–96.

———. 1998. Frame Relay. *Data Communications,* June, 76, 78–86.

Harbison, R. W. 1992. Frame Relay: Technology for Our Time. *LAN Technology,* December, 67–78.

Held, G. 1997. Shopping for Frame Relay. *Network Magazine,* December, 83–87.

Krivda, C. 1996. Frame Relay Comes of Age. *LAN,* November, 117–124.

Merritt, J. 1992. Frame Relay Technology: The Right Link for Wide Area Backbones. *Network Computing,* 1 November, 75–76.

Newman, D., T. Giorgis, and F. Yavari-Issalou. 1998. FRADs: Halfway There. *Data Communications,* 21 September, 43–54.

Newman, J. 1997. Frame Relay Makes Its Voice Heard. *Network Computing,* 15 March, 134–138.

Parnell, T. 1997. *LAN Times Guide to Wide Area Networks.* Berkeley, CA: Osborne McGraw-Hill.

Parsons, T., and D. Bahr. 1996. Acing the Frame Relay Race. *LAN,* June, 95–104.

———, and T. Beach. 1996. Rapid Transit: Frame Relay to ATM. *LAN,* August, 115–120.

Reardon, M. 1998. Frame Relay. *Data Communications,* June, 77, 90–97.

Stallings, W. 2000. *Data and Computer Communications.* 6th ed. Upper Saddle River, NJ: Prentice-Hall.

Willis, D. 1997a. Make Room for Frame Relay. *Networking Computing,* 15 February, 73–86.

———. 1997. Framing the Perfect WAN Contract. *Network Computing,* 15 September, 136–140.

Wu, D. 1997b. Client-Server over Frame Relay. *Network Magazine,* October, 105–109.

Chapter 13: Switched Multimegabit Data Service (SMDS)

Atkins, J., and M. Norris. 1995. *Total Area Networking: ATM, Frame Relay and SMDS Explained*. New York: Wiley.

Bates, R. J., and D. Gregory. 1998. *Voice and Data Communications Handbook.* New York: McGraw-Hill.

Bellcore. 1990. Exchange Access SMDS Service Generic Requirements. *Bellcore Technical Advisory*, TA-TSV-001060*,* December.

———. 1991. Generic System Requirements in Support of Switched Multi-megabit Data Service. *Bellcore Technical Reference*, TR-TSV-000772, May.

Data Communications. 1994. SMDS Is Here: Where Are the Customers? *Data Communications,* October, 55–56.

———. 1997. The Top 25 Flops. *Data Communications,* 21 October, 142–143.

Fitzgerald, S. 1993. Packing Them In. *LAN Magazine,* September, 77–86.

Gallo, M., and W. Hancock. 2002. *Computer Communications and Networking Technologies.* Pacific Grove, CA: Brooks/Cole.

Hughes, L., and S. Starliper. 1990. Switched Multimegabit Data Service (SMDS). *Connexions: The Interoperability Report*, 4(10): 34–37.

Krishnaswamy, P., and M. Ulema. 1991. Developments in SMDS. *Connexions: The Interoperability Report*, 5(10): 24–31.

Kumar, B. 1995. *Broadband Communications: A Professional's Guide to ATM, Frame Relay, SMDS, SONET, and BISDN.* New York: McGraw-Hill.

Mendes, G. 1996. SMDS: A WAN Alternative. *LAN Interoperability: A Quarterly Supplement to LAN Magazine,* August, 22–27.

Miller, M. 1994. *Analyzing Broadband Networks: Frame Relay, SMDS, and ATM.* San Mateo, CA: M&T.

Minoli, D. 1991. The New Wide Area Technologies: SMDS and BISDN. *Network Computing,* August, 88–92.

Parnell, T. 1997. *LAN Times Guide to Wide Area Networks.* Berkeley, CA: Osborne McGraw-Hill.

Sharer, R. 1993. The SMDS Express. *LAN Magazine,* July, 51–58.

———. 1994. Lighten Up. *LAN Magazine,* March, 109–112.

The Institute of Electrical and Electronics Engineers. 1990. Distributed Queue Dual Bus (DQDB) Subnetwork of a Metropolitan Area Network (MAN). *IEEE Standard 802.6-1990.* The Institute of Electrical and Electronics Engineers.

Chapter 14: Asynchronous Transfer Mode (ATM)

Askins, J. 1996. Migrating to ATM? Learn Before You Leap. *Data Communications,* December, 79–80.

Atkins, J., and M. Norris. 1995. *Total Area Networking: ATM, Frame Relay and SMDS Explained*. New York: Wiley.

Bates, R. J., and D. Gregory. 1998. *Voice and Data Communications Handbook.* New York: McGraw-Hill.

Chipcom. 1993. *ATM: The Frontier of Networking.* Southborough, MA: Chipcom.

Cholewja, K. 1996. Affordable ATM. *Data Communications,* December, 94–102.

Clark, E. 1997. Real-World ATM. *Network,* July, 42–53.

Conover, J. 1996. ATM Access Is Ready, Willing and Able to Fly on Your Backbone. *Network Computing,* 15 November, 70–95.

Feltman, C. 1993a. ATM Takes Off. *Interoperability,* 19–46.

———. 1993b. ATM: The Grand Unifier. *LAN Magazine*, November, 52–70.

Fritz, J. N. 1998. ATM WAN Services: Picking a Winner. *Data Communications,* 21 April, 77–80.

Gallo, M., and W. Hancock. 2002. *Computer Communications and Networking Technologies.* Pacific Grove, CA: Brooks/Cole.
Gould, J. 1994. ATM's Long, Strange Trip to the Mainstream. *Data Communications,* June, 120–130.
Gould, W. 1997. Fast Ethernet Pairs with ATM. *Network Magazine,* September, 74–77.
Hurwicz, M. 1997. ATM for the Rest of Us. *Network Magazine,* November, 75–79.
King, S. 1994. Switched Virtual Networks. *Data Communications,* September, 66–80.
Mehta, S. 1996. Telcos: Answering the Call for ATM. *LAN,* March, 46–52.
Minoli, D. 1992a. ATM: The Future of Local and Wide Area Networks. *Network Computing,* 15 October, 128–134.
———. 1992b. ATM Protocols: Let's Get Technical. *Network Computing,* 15 November, 156–163.
Newman, D. 1997. ATM: The Last Word. *Data Communications,* April, 105–133.
Parikh, A. 1993. The Challenge of Migrating to ATM. *Networking Management,* June, B11–B15.
Parnell, T. 1997. *LAN Times Guide to Wide Area Networks.* Berkeley, CA: Osborne McGraw-Hill.
Roberts, L. G. 1997. CIF: Affordable ATM, at Last. *Data Communications,* April, 96–102.
Ruiu, D. 1993. ATM at Your Service? *Data Communications,* November, 85–88.
Stallings, W. 2000. *Data and Computer Communications.* 6th ed. Upper Saddle River, NJ: Prentice-Hall.
Taylor, M. 1996. Voice over ATM: A Sound Assessment. *Data Communications,* December, 86–92.
Taylor, S. 1994. ATM and Frame Relay: Back to the Basics. *Data Communications,* April, 23–24.
———. 1994. Making the Switch to High-Speed WAN Services. *Data Communications,* July, 87–94.
Wittman, A. 1993. Local ATM: Is ATM Ready for the Desktop? *Network Computing,* September, 160–166.
Zeichick, A. 1997. Glossary: Asynchronous Transfer Mode. *Network,* July, 28.

Chapter 15: Dialup and Home Networking

Aber, R. 1997. xDSL Supercharges Copper. *Data Communications,* March, 99–105.
Amati Communications Corporation. 1996. *ADSL Field Guide.* San Jose, CA: Amati Communications Corporation.
Angel, J. 1999. High-Speed Internet Access Comes Home. *Network Magazine,* February, 42–47.
Bolles, G. A. 1991. Modem Talk. *Network Computing,* August, 73–81.
———. 1992. Modem Talk: What's My Line. *Network Computing,* 1 October, 85–89.
Burkes, R. E. 1991. Making a Case for Dial-up LAN Links. *LAN Technology,* March, 49–54.
Cholewka, K. 1997. 56K Modems: The New Spin on Speed. *Data Communications,* May, 93–103.
Clark, E. 1996a. Data Meets the Dial Tone. *Interoperability: Supplement to LAN,* February, 20–33.
———. 1996b. PC, Phone Home! *LAN,* October, 125–132.
———. 1999. Pricing the Last Mile. *Network Magazine,* February, 36–41.
Fitzgerald, S. 1996. Reshaping the Digital Landscape. *LAN Interoperability: A Quarterly Supplement to LAN Magazine,* August, 14–20.

Frezza, B. 1996. Cable TV: Giving Us the Broadband Business. *Network Computing,* 15 May, 31–32.

Gallo, M., and W. Hancock. 2002. *Computer Communications and Networking Technologies.* Pacific Grove, CA: Brooks/Cole.

Kalman, S. 1997. The Bright Future of CTI. *LAN,* January, 85–92.

Liebmann, L. 1996. Free Market Theory. *LAN,* July, 60–64.

Maclean, P. 1993. V.FAST and V.32TERBO: Faster Modems, Slower Standards. *Network Computing,* July, 110–130.

Makris. J. 1998. DSL: Don't Be Duped. *Data Communications,* 21 April, 39–52.

New Lease on Old Wires. 2001. *Packet: Cisco Systems Users Magazine*, 29–31.

Newman, D., M. Carter, and H. Holzbaur. 1998. DSL: Worth Its Wait. *Data Communications,* June, 101–112.

Newman, J. 1997. ADSL: Putting a Charge into Your Copper Cable. *Network Computing,* 1 May, 139–142.

Richardson, R. 1996. Tone Dialing, SOHO Style. *LAN,* October, 107–114.

———. 1997. Betting WAN Access Technology. *Network Computing,* 1 July, 43–56.

Steinke, S. 1996. Competition Comes to the WAN. *LAN,* July, 52–58.

———. 1997. Rehab for Copper Wire. *LAN,* February, 57–62.

———. 1999. A Lineup of Local-Loop Contenders. *Network Magazine,* February, 30–35.

Tadjer, R. 1996. 56-kbps Modems Are in Search of a Home. *Network Computing,* 15 November, 24–26.

The Facts on Coax. 2000. *Packet: Cisco Systems Users Magazine*, Third Quarter, 26–27.

Zeichick, A. 1997. LAN Glossary: Computer Telephony. *LAN,* January, 138.

Zgodzinski, D. 1996a. The Cable Chase. *Internet World,* June, 63–66.

———. 1996b. Enter ADSL. *Internet World,* October, 72–75.

Chapter 16: Network Security Issues

Baum, M. 1994. *Federal Certification Authority Liability and Policy.* Cambridge, MA: Independent Monitoring.

Bird, T. 1998. Building VPNs: The 10-Point Plan. *Data Communications,* June, 123–132.

Bishop. M. 1992. Foiling Password Crackers. *UNIXWorld,* March, 85–91.

Bryant, B. 1988. Designing an Authentication System: A Dialogue in Four Scenes. *Massachusetts Institute of Technology,* 8 February.

Carl-Mitchell, S., and J. S. Quarterman. 1992. Building Internet Firewalls. *UNIXWorld,* February, 93–102.

Cheswick, W. R., and S. M. Bellovin. 1994. *Firewalls and Internet Security: Repelling the Wily Hacker.* Reading, MA: Addison-Wesley.

Cormen, T. H., C. E. Leiserson, and R. L. Rivest, 1990. *Introduction to Algorithms*. New York: McGraw-Hill.

Defensive Attitude: Strategies for Dealing with Hackers and Crackers. 20001. *Packet: Cisco Systems Users Magazine*, First Quarter, 71–73.

Dyson, P. 1995. *The Network Press Dictionary of Networking.* 2nd ed. San Francisco: Sybex.

Ellis, J., B. Fraser, and L. Pesante. 1994. Keeping Internet Intruders Away. *UNIX Review,* September, 35–44.

Farrow, R. 1999. Public Key Infrastructure Basics. *Network Magazine,* January, 78–79.

———, and R. Power. 1998. Is Someone Scanning Your Network? *Network Magazine,* September, 83–84.

Feibel, W. 1995. *The Network Press Encyclopedia of Networking.* 2nd ed. San Francisco: Sybex.

Fratto, M. 1997. Unlocking Private Networks. *Networking Computing,* 1 November, 53–76.

———. 1998. IPSec–Compliant VPN Solutions: Virtualizing Your Network. *Network Computing,* 1 August, 72–80.

Gallo, M., and W. Hancock. 2002. *Computer Communications and Networking Technologies.* Pacific Grove, CA: Brooks/Cole.

———, and R. Nenno. 1985. *Computers and Society with Basic and Pascal.* Boston: PWS.

Garfinkel, S. 1991. *Practical UNIX Security.* Sebastopol, CA: O'Reilly & Associates.

———, and G. Spafford. 1991. Building a Network Firewall. *LAN Technology,* December, 59–68.

Hale, R. 2001. The Many Roads to Intrusion Detection. *Cisco World*, June, 18–20.

Hontanon, R. 1999. Managing UNIX Security. *Network Magazine*, January, 52–56.

Hoover, A. 1995. Securing the Enterprise: Firewalls Can Keep You from Getting Burned. *Internet World,* February, 39–43.

Karvé, A. 1997a. Lesson 104: Public Key Cryptography. *LAN,* April, 23–24.

———. 1997b. Public Key Infrastructures. *Network Magazine,* November, 69–73.

———. 1998. Lesson 115: IP Security. *Network Magazine,* February, 27–28.

———. 1999. Making Sense of the PKI Puzzle. *Network Magazine,* October, 56–60.

Landau, S. 2000. Designing Cryptography for the New Century. *Communications of the ACM*, May, 115–120.

Meredith, G. 2001. Play It Safe Sam. *Packet: Cisco Systems Users Magazine*, First Quarter, 67–70.

Morse, S. 1994. Network Security: How to Stay Ahead of Network Explosion. *Network Computing,* 15 February, 54–63.

Moskowitz, R. G. 1994. Firewalls: Building in That Peaceful, Easy Feeling. *Network Computing,* 1 June, 159–161.

Pescatore, J. 2001. Security in the Internet Economy. *Packet: Cisco Systems Users Magazine*, First Quarter, 54–59.

Quarterman, J. S., and S. Carl-Mitchell. 1993. Tutorial: Local Protection for Networked Systems. *UNIXWorld,* July, 64–72.

Ranum, M. 1994. Internet Firewall Protection. *Open Computing,* September, 95–99.

Reavis, J. 2001. Feature: Goodbye DES, Hello AES. *Network World*, 30 July [Electronic document]; available from http://www.nwfusion.com/research/2001/0730feat2.html.

Rose, T. 2001. How to Protect Your Digital Identity. *Cisco World*, June, 24–25.

Schwartau, W. 1995. Beyond the Firewall: New Systems Watch Intruders and Strike Back. *Internet World,* February, 44–48.

Securing the Mobile Enterprise. 2001. *Packet: Cisco Systems Users Magazine*, First Quarter, 13–18.

Simonds, F. 1996. *Network Security.* New York: McGraw-Hill.

Siyan, K., and C. Hare. 1995. *Internet Firewalls and Network Security.* Indianapolis: New Riders Publishing.

Stallings, W. 1993. Make It Real. *LAN Magazine,* September, 105–112.

———. 1995. Getting Cryptic: PGP for You and Me. *Internet World,* February, 34–39.

———. 2000. *Data and Computer Communications.* 6th ed. Upper Saddle River, NJ: Prentice-Hall.

Steinke, S. 1998. Authentication and Cryptography. *Network Magazine,* January, 51–55.

Stephenson, P. 1995. Securing Remote Access. *Network Computing,* 1 February, 130–134.
Strebe, M., C. Perkins, and M. G. Moncur. 1998. *NT Network Security.* San Francisco: Sybex.
Thayer, R. 1997. Bulletproof IP. *Data Communications,* 21 November, 55–60.
Vaughn-Nichols, S. 1995. It's Alive: Clipper's Still Kicking. *Internet World,* February, 62–65.
Wexler, J. 2000. Defense Mechanisms. *Packet: Cisco Systems Users Magazine*, Third Quarter, 52–57.
———. 2001. Security Blanket: Weaving Security Into the Network Blanket. *Packet: Cisco Systems Users Magazine*, First Quarter, 61–65.

Chapter 17: Network Convergence

Bajorek, C. 2001. ABC's of Transitioning to VoIP. *ComputerTelephony*, January, 95–96.
Bates, R. J., and D. Gregory. 1998. *Voice and Data Communications Handbook.* New York: McGraw-Hill.
Brown, D. 1999, Video Conferencing 2000: H,323's Year? *Network Computing*, 6 September, 79–98.
Carden, P. 2000. Building Voice over IP. [Electronic document: 8 May]; available from http://www.nwc.com/netdesign/1109voipfull.html.
Dyson, P. 1995. *The Network Press Dictionary of Networking*. 2nd ed. San Francisco: Sybex.
Essex, D. 1999. IP Telephony: What Does the Future Hold? *Network Magazine*, January, 40–44.
Extreme Networks. 2000a. Combining Voice over IP with Policy-Based Quality of Service. [Electronic document]; available from http://www.extremenetworks.com.
———. 2000b. An Overview of Voice over IP Standards and Protocol. [Electronic document]; available from http://www.extremenetworks.com.
Feibel, W. 1995. *The Network Press Encyclopedia of Networking*. 2nd ed. San Francisco: Sybex.
French, J. 1999. Voice Hitches a Ride with Data. *Packet: Cisco Systems Users Magazine*, Fourth Quarter, 58–61.
Gallo, M., and W. Hancock. 2002. *Computer Communications and Networking Technologies.* Pacific Grove, CA: Brooks/Cole.
Hall, E. 1999. Preparing Your Network for IP Multicasting. *Networking Computing*, 17 May, 121–122.
Hancock, B. 1989. *Network Concepts and Architectures*. Wellesley, MA: QED Information Sciences.
———. 1996. *Advanced Ethernet/802.3 Management and Performance*. Boston: Digital Press.
Held, G. 1988. Modem Modulation: Techniques and Compatibility Issues. In *Handbook of Communication Systems Management*, ed. J. W. Conrad, 419-431. Boston: Auerbach Publishers.
Horak, R. 2000. *Communications Systems & Networks*. 2nd ed. Foster City, CA: M&T Books.
Jessup, T. 1999. Porting the PBX. *Data Communications*, July, 72–76.
Kraskey, T., and J. McEachern. 1999. Next-Generation Network Voice Services. *Network Magazine*, 1 December [Electronic document]; available from http://www.networkmagazine.com/article/NMG20000426S0024.
Kurose, J. F., and K. W. Ross. 2001. *Computer Networking: A Top-Down Approach Featuring the Internet/* Boston: Addison Wesley Longman.
Lawson, S. 2001. IEEE 1394b to Rev Up Multimedia Networks. [Electronic document: 25 May]; available from http://www.nwfusion.com/news/2001/0525ieee.html.
Lippis, N. 1999. Enterprise VOIP: Two to Go. *Data Communications*, July, 19–20.

Metzler, J. 2000. The Merging of Telephony and IP: How and Why to Make the Move. *Packet: Cisco Systems Users Magazine,* Second Quarter, 60–65.

Michael, B. 2000. DSL Solution. *ComputerTelephony*, June, 88–113.

Nolle, T. 1999. Voices on the LAN. *Network Magazine*, August, 34.

Perez, D. 2000. Voice over IP: Starting at the Beginning. *Cisco World*, October, 22–26.

Prepping LANs for Conversion. 2000. Packet: Cisco Systems Users Magazine, Third Quarter, 29–32.

Rosenberg, J. and R. Shockey. 2000. The Session Initiation Protocol (SIP): A Key Component for Internet Telephony. *ComputerTelephony*, June, 124–139.

Scholl, F. 1997. Planning for Multimedia. *Network Magazine*, June, 101–105.

Shockey, R. 2001. ENUM: Phone Numbers Meet the Net. [Electronic document: 5 July]; available from http://www.cconvergence.com/article/CTM20010618S0010.

Stallings, W. 1997. *Local and Metropolitan Area Networks*. 5th ed. Upper Saddle River, NJ: Prentice-Hall.

———. 2000. *Data and Computer Communications*. 6th ed. Upper Saddle River, NJ: Prentice-Hall.

Stamper, D. A. 1991. *Business Data Communications*. 3rd ed. Redwood City, CA: Benjamin/Cummings.

Steinke, S. 1996. Multimedia: The I-Way Drive-In. *LAN*, August, 45–50.

Tanenbaum, A. S. 1996. *Computer Networks*. 3rd ed. Upper Saddle River, NJ: Prentice-Hall.

Willis, D. 1999a. Integrating Voice and Data. *Networking Computing*, 18 October, 50–73.

———. 1999b. Voice Over IP, The Way It Should Be. *Networking Computing*, 11 January, 96–99.

Woods, D. 1999. Translating Menus at the VoIP Cafe. [Electronic document: 27 December]; available from http://www.nwc.com/1026/1026ws1.html.

Zeichick. A. 1998. Tutorial—Lesson 125: SIgnaling System 7. *Network Magazine*, 1 December [Electronic document]; available from http://www.networkmagazine.com/article/NMG20000508S0044.

Chapter 18: Wireless Networking

Angel, J. 1999. Tutorial—Lesson 137: Wireless LANs. [Electronic document: 1 December]; available from http://www.networkmagazine.com/article/NMG20001106S0004.

———. 2000. Look Ma, No Cables. [Electronic document: 6 November]; available from http://www.networkmagazine.com/article/NMG20000517S0164.

Baig, E. C. 2001. The Era of Living Wirelessly. *USA Today*, 26 June, 1E–2E.

Bates, R. J., and D. Gregory. 1998. *Voice and Data Communications Handbook.* New York: McGraw-Hill.

Belanger, P. 2000. Wi-Fi Certification Helps Wireless LANs Come of Age. *Network World*, 21 August [Electronic document]; available from http://www.nwfusion.com/columnists/2000/0821belanger.html.

Cox, J. 2001. Wireless LAN Holes Exposed. *Network World*, 12 February [Electronic document]; available from http://www.nwfusion.com/archive/2001/116952_02-12-2001.html.

———. 2001. Security Hole Exposed in 802.11b Wireless LANs. *Network World*, 6 August [Electronic document]; available from http://www.nwfusion.com/news/2001/0806ieee.html

Data.com. 2000. LANS Unplugged: Wireless LANs and IEEE 802.11. [Electronic document: 11 April]; available from http://www.networkmagazine.com/article/PIT20000410S0047.

Dornan, A. 2001. Wireless Ethernet: Neither Bitten Nor Blue. *Network Magazine*, 11 April [Electronic document]; available from http://www.commweb.com/article/COM20010411S0005.

Frank, A. 2000. HomeRF Fights for Its Place in the Sun. [Electronic document: 19 July]; available from http://www.networkmagazine.com/article/DCM20000719S0002.
———. 2000. FCC Spreads Joy for Frequency-Hoppers. [Electronic document: 1 September]; available from http://www.commweb.com/article/COM20001002S0003/3
———. 2000. Wireless Networking Looks to 5 GHz. [Electronic document: 3 October]; available from http://www.commweb.com/article/COM20000901S0007.
Horak, R. 2000. *Communications Systems & Networks*. 2nd ed. Foster City, CA: M&T Books.
Kruse, S. 2001. Security Poses a Risk in the Wireless World. *Cisco World*, May, 12.
Meredith, G. 2000. Securing the Wireless LAN. *Packet: Cisco Systems Users Magazine*, Third Quarter, 74–77.
Miller, K. 1994. Cellular Essentials for Wireless Data Transmission. *Data Communications*, 21 March, 61–67.
Nedeltchev, P. 2001. Wireless LAN Ready for Prime Time. *Packet: Cisco Systems Users Magazine*, Third Quarter, 91.
Poe, S. 2000. WAP Leading the Way to Mobile Phone Access. *Cisco World*, September, 28–29.
Robb, J. 2000. Wireless Standards: A Tower of Babel. Cisco World, September, 38–39.
Rose, F. Pocket Monster. *Wired*, September, 126–135.
Snider, M. 2001. Wireless World has Fewer Wide-Open Spaces. *USA Today*, 31 July, 3D.
Stallings, W. 2001. IEEE 802.16 for Broadband Wireless. *Network World*, 3 September [Electronic document]; available from http://www.nwfusion.com/news/tech/2001/0903tech.html.
Wexler, J. 2001. Mobility for the Masses. *Packet: Cisco Systems Users Magazine*, First Quarter, 85–95.
Yaukey, J. 2001. Japanese Wireless Sensation to Hit U.S. *USA Today*, 18 April, 1T.

Index