THIRD EDITION

COMPUTER NETWORKS

ANDREW S. TANENBAUM

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Computer Networks

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Computer Networks

Third Edition

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To Suzanne, Barbara, Marvin, and Little Bram

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PREFACE

This book is now in its third edition. Each edition has corresponded to a different phase in the way computer networks were used. When the first edition appeared in 1980, networks were an academic curiosity. When the second edition appeared in 1988, networks were used by universities and large businesses. When the third edition appeared in 1996, computer networks, especially the worldwide Internet, had become a daily reality for millions of people.

Furthermore, the networking hardware and software have completely changed since the second edition appeared. In 1988, nearly all networks were based on copper wire. Now, many are based on fiber optics or wireless communication. Proprietary networks, such as SNA, have become far less important than public networks, especially the Internet. The OSI protocols have quietly vanished, and the TCP/IP protocol suite has become dominant. In fact, so much has changed, the book has almost been rewritten from scratch.

Although Chap. 1 has the same introductory function as it did in the second edition, the contents have been completely revised and brought up to date. For example, instead of basing the book on the seven-layer OSI model, a five-layer hybrid model (shown in Fig. 1-21) is now used and introduced in Chap. 1. While not exactly identical to the TCP/IP model, it is much closer to the TCP/IP model in spirit than it is to the OSI model used in the second edition. Also, the new running examples used throughout the book—the Internet and ATM networks— are introduced here, along with some gigabit networks and other popular networks.

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In Chap. 2, the focus has moved from copper wire to fiber optics and wireless communication, since these are the technologies of the future. The telephone system has become almost entirely digital in the past decade, so the material on it has been largely rewritten, with new material on broadband ISDN added. The material on cellular radio has been greatly expanded, and new material on low-orbit satellites has been added to the chapter.

The order of discussion of the data link layer and the MAC sublayer has been reversed, since experience with students shows that they understand the MAC sublayer better after they have studied the data link layer. The example protocols there have been kept, as they have proven very popular, but they have been rewritten in C. New material on the Internet and ATM data link layers has been added.

The MAC sublayer principles of Chap. 4. have been revised to reflect new protocols, including wavelength division multiplexing, wireless LANs, and digital radio. The discussion of bridges has been revised, and new material has been added on high-speed LANs.

Most of the routing algorithms of Chap. 5 have been replaced by more modern ones, including distance vector and link state routing. The sections on congestion control have been completely redone, and material on the running examples, the Internet and ATM is all new.

Chap. 6 is still about the transport layer, but here, too, major changes have occurred, primarily, the addition of a large amount of new material about the Internet, ATM, and network performance.

Chap. 7, on the application layer, is now the longest chapter in the book. The material on network security has been doubled in length, and new material has been added on DNS, SNMP, email, USENET, the World Wide Web, HTML, Java, multimedia, video on demand, and the MBone.

Of the 395 figures in the third edition, 276 (70 percent) are completely new and some of the others have been revised. Of the 371 references to the literature, 282 (76 percent) are to books and papers that have appeared since the second edition was published. Of these, over 100 are to works published in 1995 and 1996 alone. All in all, probably 75 percent of the entire book is brand new, and parts of the remaining 25 percent have been heavily revised. Since this is effectively a new book, the cover was redesigned to avoid confusion with the second edition.

Computer books are full of acronyms. This one is no exception. By the time you are finished reading this one, all of the following should ring a bell: AAL, AMPS, ARP, ASN, ATM, BGP, CDMA, CDPD, CSMA, DQDB, DNS, FAQ, FDM, FTP, FTTC, FTTH, GSM, HDLC, HEC, HIPPI, IAB, ICMP, IDEA, IETF, IPv6, ISO, ITU, LATA, MAC, MACA, MAN, MIB, MIME, NAP, NNTP, NSA, NSAP, OSI, OSPF, PCM, PCN, PCS, PEM, PGP, PPP, PSTN, PTT, PVC, QAM, RARP, RFC, RSA, SABME, SAP, SAR, SDH, SDLC, SHA, SMI, SNA, SNMP, SNRME, SPX, TCP, UDP, VHF, VLF, VSAT, WARC, WDM, WWV, and WWW. But don't worry. Each one will be carefully defined before it is used.

To help instructors using this book as a text for course, the author has prepared three teaching aids:

- A problem solutions manual.
- PostScript files containing all the figures (for making overhead sheets).
- A simulator (written in C) for the example protocols of Chap. 3.

The solutions manual is available from Prentice Hall (but only to instructors). The file with the figures and the simulator are available via the World Wide Web. To get them, please see the author's home page: *http://www.cs.vu.nl/~ast/* .

The book was typeset in Times Roman using Troff, which, after all these years, is still the only way to go. While Troff is not as trendy as WYSIWYG systems, the reader is invited to compare the typesetting quality of this book with books produced by WYSIWYG systems. My only concession to PCs and desktop publishing is that for the first time, the art was produced using Adobe Illustrator, instead of being drawn on paper. Also for the first time, the book was produced entirely electronically. The PostScript output from Troff was sent over the Internet to the printer, where the film for making the offset plates was produced. No intermediate paper copy was printed and photographed, as is normally done.

Many people helped me during the course of the third edition. I would especially like to thank Chase Bailey, Saniya Ben Hassen, Nathaniel Borenstein, Ron Cocchi, Dave Crocker, Wiebren de Jonge, Carl Ellison, M. Rasit Eskicioglu, John Evans, Mario Gerla, Mike Goguen, Paul Green, Dick Grune, Wayne Hathaway, Franz Hauck, Jack Holtzman, Gerard Holzmann, Philip Homburg, Peter Honeyman, Raj Jain, Dave Johnson, Charlie Kaufman, Vinay Kumar, Jorg Liebeherr, Paul Mockapetris, Carol Orange, Craig Partridge, Charlie Perkins, Thomas Powell, Greg Sharp, Anne Steegstra, George Swallow, Mark Taylor, Peter van der Linden, Hans van Staveren, Maarten van Steen, Kees Verstoep, Stephen Walters, Michael Weintraub, Joseph Wilkes, and Stephen Wolff. Special thanks go to Radia Perlman for many helpful suggestions. My students have also helped in many ways. I would like to single out Martijn Bot, Wilbert de Graaf, Flavio del Pomo, and Arnold de Wit for their assistance.

My editor at Prentice Hall, Mary Franz, provided me with more reading material than I had consumed in the previous 10 years. She was also helpful in numerous other ways, small, medium, large, and jumbo. My production editor, Camille Trentacoste, taught me about people of snow, 8-up flats, fax [sic], and other important items, while performing yeoperson's service with a Picky Author and a tight schedule.

Finally, we come to the most important people. Suzanne, Barbara, Marvin, and even little Bram, have been through this routine before. They endure it with infinite patience and good grace. Thank you.

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INTRODUCTION

Each of the past three centuries has been dominated by a single technology. The 18th Century was the time of the great mechanical systems accompanying the Industrial Revolution. The 19th Century was the age of the steam engine. During the 20th Century, the key technology has been information gathering, processing, and distribution. Among other developments, we have seen the installation of worldwide telephone networks, the invention of radio and television, the birth and unprecedented growth of the computer industry, and the launching of communication satellites.

Due to rapid technological progress, these areas are rapidly converging, and the differences between collecting, transporting, storing, and processing information are quickly disappearing. Organizations with hundreds of offices spread over a wide geographical area routinely expect to be able to examine the current status of even their most remote outpost at the push of a button. As our ability to gather, process, and distribute information grows, the demand for even more sophisticated information processing grows even faster.

Although the computer industry is young compared to other industries (e.g., automobiles and air transportation), computers have made spectacular progress in a short time. During the first two decades of their existence, computer systems were highly centralized, usually within a single large room. Not infrequently, this room had glass walls, through which visitors could gawk at the great electronic wonder inside. A medium-size company or university might have had one or two

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computers, while large institutions had at most a few dozen. The idea that within 20 years equally powerful computers smaller than postage stamps would be mass produced by the millions was pure science fiction.

The merging of computers and communications has had a profound influence on the way computer systems are organized. The concept of the "computer center" as a room with a large computer to which users bring their work for processing is now totally obsolete. The old model of a single computer serving all of the organization's computational needs has been replaced by one in which a large number of separate but interconnected computers do the job. These systems are called **computer networks**. The design and organization of these networks are the subjects of this book.

Throughout the book we will use the term "computer network" to mean an *interconnected* collection of *autonomous* computers. Two computers are said to be interconnected if they are able to exchange information. The connection need not be via a copper wire; fiber optics, microwaves, and communication satellites can also be used. By requiring the computers to be autonomous, we wish to exclude from our definition systems in which there is a clear master/slave relation. If one computer can forcibly start, stop, or control another one, the computers are not autonomous. A system with one control unit and many slaves is not a network; nor is a large computer with remote printers and terminals.

There is considerable confusion in the literature between a computer network and a **distributed system**. The key distinction is that in a distributed system, the existence of multiple autonomous computers is transparent (i.e., not visible) to the user. He[†] can type a command to run a program, and it runs. It is up to the operating system to select the best processor, find and transport all the input files to that processor, and put the results in the appropriate place.

In other words, the user of a distributed system is not aware that there are multiple processors; it looks like a virtual uniprocessor. Allocation of jobs to processors and files to disks, movement of files between where they are stored and where they are needed, and all other system functions must be automatic.

With a network, users must *explicitly* log onto one machine, *explicitly* submit jobs remotely, *explicitly* move files around and generally handle all the network management personally. With a distributed system, nothing has to be done explicitly; it is all automatically done by the system without the users' knowledge.

In effect, a distributed system is a software system built on top of a network. The software gives it a high degree of cohesiveness and transparency. Thus the distinction between a network and a distributed system lies with the software (especially the operating system), rather than with the hardware.

Nevertheless, there is considerable overlap between the two subjects. For example, both distributed systems and computer networks need to move files around. The difference lies in who invokes the movement, the system or the user.

† "He" should be read as "he or she" throughout this book.

Although this book primarily focuses on networks, many of the topics are also important in distributed systems. For more information about distributed systems, see (Coulouris et al., 1994; Mullender, 1993; and Tanenbaum, 1995).

1.1. USES OF COMPUTER NETWORKS

Before we start to examine the technical issues in detail, it is worth devoting some time to pointing out why people are interested in computer networks and what they can be used for.

1.1.1. Networks for Companies

Many organizations have a substantial number of computers in operation, often located far apart. For example, a company with many factories may have a computer at each location to keep track of inventories, monitor productivity, and do the local payroll. Initially, each of these computers may have worked in isolation from the others, but at some point, management may have decided to connect them to be able to extract and correlate information about the entire company.

Put in slightly more general form, the issue here is **resource sharing**, and the goal is to make all programs, equipment, and especially data available to anyone on the network without regard to the physical location of the resource and the user. In other words, the mere fact that a user happens to be 1000 km away from his data should not prevent him from using the data as though they were local. This goal may be summarized by saying that it is an attempt to end the "tyranny of geography."

A second goal is to provide **high reliability** by having alternative sources of supply. For example, all files could be replicated on two or three machines, so if one of them is unavailable (due to a hardware failure), the other copies could be used. In addition, the presence of multiple CPUs means that if one goes down, the others may be able to take over its work, although at reduced performance. For military, banking, air traffic control, nuclear reactor safety, and many other applications, the ability to continue operating in the face of hardware problems is of utmost importance.

Another goal is **saving money**. Small computers have a much better price/performance ratio than large ones. Mainframes (room-size computers) are roughly a factor of ten faster than personal computers, but they cost a thousand times more. This imbalance has caused many systems designers to build systems consisting of personal computers, one per user, with data kept on one or more shared **file server** machines. In this model, the users are called **clients**, and the whole arrangement is called the **client-server model**. It is illustrated in Fig. 1-1.

In the client-server model, communication generally takes the form of a request message from the client to the server asking for some work to be done.

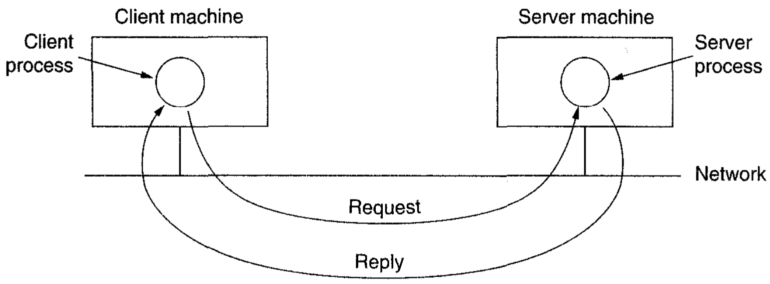

Fig. 1-1. The client-server model.

The server then does the work and sends back the reply. Usually, there are many clients using a small number of servers.

Another networking goal is scalability, the ability to increase system performance gradually as the workload grows just by adding more processors. With centralized mainframes, when the system is full, it must be replaced by a larger one, usually at great expense and even greater disruption to the users. With the client-server model, new clients and new servers can be added as needed.

Yet another goal of setting up a computer network has little to do with technology at all. A computer network can provide a powerful **communication medium** among widely separated employees. Using a network, it is easy for two or more people who live far apart to write a report together. When one worker makes a change to an on-line document, the others can see the change immediately, instead of waiting several days for a letter. Such a speedup makes cooperation among far-flung groups of people easy where it previously had been impossible. In the long run, the use of networks to enhance human-to-human communication will probably prove more important than technical goals such as improved reliability.

1.1.2. Networks for People

The motivations given above for building computer networks are all essentially economic and technological in nature. If sufficiently large and powerful mainframes were available at acceptable prices, most companies would simply choose to keep all their data on them and give employees terminals connected to them. In the 1970s and early 1980s, most companies operated this way. Computer networks only became popular when networks of personal computers offered a huge price/performance advantage over mainframes.

Starting in the 1990s, computer networks began to start delivering services to private individuals at home. These services and the motivations for using them are quite different than the "corporate efficiency" model described in the previous section. Below we will sketch three of the more exciting ones that are starting to happen:

1. Access to remote information.
2. Person-to-person communication.
3. Interactive entertainment.

Access to remote information will come in many forms. One area in which it is already happening is access to financial institutions. Many people pay their bills, manage their bank accounts, and handle their investments electronically. Home shopping is also becoming popular, with the ability to inspect the on-line catalogs of thousands of companies. Some of these catalogs will soon provide the ability to get an instant video on any product by just clicking on the product's name.

Newspapers will go on-line and be personalized. It will be possible to tell the newspaper that you want everything about corrupt politicians, big fires, scandals involving celebrities, and epidemics, but no football, thank you. At night while you sleep, the newspaper will be downloaded to your computer's disk or printed on your laser printer. On a small scale, this service already exists. The next step beyond newspapers (plus magazines and scientific journals) is the on-line digital library. Depending on the cost, size, and weight of book-sized notebook computers, printed books may become obsolete. Skeptics should take note of the effect the printing press had on the medieval illuminated manuscript.

Another application that falls in this category is access to information systems like the current World Wide Web, which contains information about the arts, business, cooking, government, health, history, hobbies, recreation, science, sports, travel, and too many other topics to even mention.

All of the above applications involve interactions between a person and a remote database. The second broad category of network use will be person-to-person interactions, basically the 21st Century's answer to the 19th Century's telephone. Electronic mail or **email** is already widely used by millions of people and will soon routinely contain audio and video as well as text. Smell in messages will take a bit longer to perfect.

Real-time email will allow remote users to communicate with no delay, possibly seeing and hearing each other as well. This technology makes it possible to have virtual meetings, called **videoconference**, among far-flung people. It is sometimes said that transportation and communication are having a race, and whichever wins will make the other obsolete. Virtual meetings could be used for remote school, getting medical opinions from distant specialists, and numerous other applications.

Worldwide newsgroups, with discussions on every conceivable topic are already commonplace among a select group of people, and this will grow to include the population at large. These discussions, in which one person posts a message and all the other subscribers to the newsgroup can read it, run the gamut from humorous to impassioned.

Our third category is entertainment, which is a huge and growing industry. The killer application here (the one that may drive all the rest) is video on demand. A decade or so hence, it may be possible to select any movie or television program ever made, in any country, and have it displayed on your screen instantly. New films may become interactive, where the user is occasionally prompted for the story direction (should MacBeth murder Duncan or just bide his time?) with alternative scenarios provided for all cases. Live television may also become interactive, with the audience participating in quiz shows, choosing among contestants, and so on.

On the other hand, maybe the killer application will not be video on demand. Maybe it will be game playing. Already we have multiperson real-time simulation games, like hide-and-seek in a virtual dungeon, and flight simulators with the players on one team trying to shoot down the players on the opposing team. If done with goggles and 3-dimensional real-time, photographic-quality moving images, we have a kind of worldwide shared virtual reality.

In short, the ability to merge information, communication, and entertainment will surely give rise to a massive new industry based on computer networking.

1.1.3. Social Issues

The widespread introduction of networking will introduce new social, ethical, political problems (Laudon, 1995). Let us just briefly mention a few of them; a thorough study would require a full book, at least. A popular feature of many networks are newsgroups or bulletin boards where people can exchange messages with like-minded individuals. As long as the subjects are restricted to technical topics or hobbies like gardening, not too many problems will arise.

The trouble comes when newsgroups are set up on topics that people actually care about, like politics, religion, or sex. Views posted to such groups may be deeply offensive to some people. Furthermore, messages need not be limited to text. High-resolution color photographs and even short video clips can now easily be transmitted over computer networks. Some people take a live-and-let-live view, but others feel that posting certain material (e.g., child pornography) is simply unacceptable. Thus the debate rages.

People have sued network operators, claiming that they are responsible for the contents of what they carry, just as newspapers and magazines are. The inevitable response is that a network is like a telephone company or the post office and cannot be expected to police what its users say. Stronger yet, having network operators censor messages would probably cause them to delete everything with even the slightest possibility of their being sued, and thus violate their users' rights to free speech. It is probably safe to say that this debate will go on for a while.

Another fun area is employee rights versus employer rights. Many people read and write email at work. Some employers have claimed the right to read and possibly censor employee messages, including messages sent from a home terminal after work. Not all employees agree with this (Sipior and Ward, 1995).

Even if employers have power over employees, does this relationship also govern universities and students? How about high schools and students? In 1994, Carnegie-Mellon University decided to turn off the incoming message stream for several newsgroups dealing with sex because the university felt the material was inappropriate for minors (i.e., those few students under 18). The fallout from this event will take years to settle.

Computer networks offer the potential for sending anonymous messages. In some situations, this capability may be desirable. For example, it provides a way for students, soldiers, employees, and citizens to blow the whistle on illegal behavior on the part of professors, officers, superiors, and politicians without fear of reprisals. On the other hand, in the United States and most other democracies, the law specifically permits an accused person the right to confront and challenge his accuser in court. Anonymous accusations cannot be used as evidence.

In short, computer networks, like the printing press 500 years ago, allow ordinary citizens to distribute their views in different ways and to different audiences than were previously possible. This new-found freedom brings with it many unsolved social, political, and moral issues. The solution to these problems is left as an exercise for the reader.

1.2. NETWORK HARDWARE

It is now time to turn our attention from the applications and social aspects of networking to the technical issues involved in network design. There is no generally accepted taxonomy into which all computer networks fit, but two dimensions stand out as important: transmission technology and scale. We will now examine each of these in turn.

Broadly speaking, there are two types of transmission technology:

1. Broadcast networks.
2. Point-to-point networks.

Broadcast networks have a single communication channel that is shared by all the machines on the network. Short messages, called **packets** in certain contexts, sent by any machine are received by all the others. An address field within the packet specifies for whom it is intended. Upon receiving a packet, a machine checks the address field. If the packet is intended for itself, it processes the packet; if the packet is intended for some other machine, it is just ignored.

As an analogy, consider someone standing at the end of a corridor with many rooms off it and shouting "Watson, come here. I want you." Although the packet may actually be received (heard) by many people, only Watson responds. The others just ignore it. Another example is an airport announcement asking all flight 644 passengers to report to gate 12.

Broadcast systems generally also allow the possibility of addressing a packet to *all* destinations by using a special code in the address field. When a packet with this code is transmitted, it is received and processed by every machine on the network. This mode of operation is called **broadcasting**. Some broadcast systems also support transmission to a subset of the machines, something known as **multicasting**. One possible scheme is to reserve one bit to indicate multicasting. The remaining $n - 1$ address bits can hold a group number. Each machine can "subscribe" to any or all of the groups. When a packet is sent to a certain group, it is delivered to all machines subscribing to that group.

In contrast, **point-to-point** networks consist of many connections between individual pairs of machines. To go from the source to the destination, a packet on this type of network may have to first visit one or more intermediate machines. Often multiple routes, of different lengths are possible, so routing algorithms play an important role in point-to-point networks. As a general rule (although there are many exceptions), smaller, geographically localized networks tend to use broadcasting, whereas larger networks usually are point-to-point.

Interprocessor distance	Processors located in same	Example
0.1 m	Circuit board	Data flow machine
1 m	System	Multicomputer
10 m	Room	
100 m	Building	Local area network
1 km	Campus	
10 km	City	Metropolitan area network
100 km	Country	
1,000 km	Continent	Wide area network
10,000 km	Planet	The internet

Fig. 1-2. Classification of interconnected processors by scale.

An alternative criterion for classifying networks is their scale. In Fig. 1-2 we give a classification of multiple processor systems arranged by their physical size. At the top are **data flow machines**, highly parallel computers with many functional units all working on the same program. Next come the **multicomputers**, systems that communicate by sending messages over very short, very fast buses. Beyond the multicomputers are the true networks, computers that communicate by exchanging messages over longer cables. These can be divided into local, metropolitan, and wide area networks. Finally, the connection of two or more networks is called an internetwork. The worldwide Internet is a well-known example of an internetwork. Distance is important as a classification metric because different techniques are used at different scales. In this book we will be concerned with only the true networks and their interconnection. Below we give a brief introduction to the subject of network hardware.

1.2.1. Local Area Networks

Local area networks, generally called **LANs**, are privately-owned networks within a single building or campus of up to a few kilometers in size. They are widely used to connect personal computers and workstations in company offices and factories to share resources (e.g., printers) and exchange information. LANs are distinguished from other kinds of networks by three characteristics: (1) their size, (2) their transmission technology, and (3) their topology.

LANs are restricted in size, which means that the worst-case transmission time is bounded and known in advance. Knowing this bound makes it possible to use certain kinds of designs that would not otherwise be possible. It also simplifies network management.

LANs often use a transmission technology consisting of a single cable to which all the machines are attached, like the telephone company party lines once used in rural areas. Traditional LANs run at speeds of 10 to 100 Mbps, have low delay (tens of microseconds), and make very few errors. Newer LANs may operate at higher speeds, up to hundreds of megabits/sec. In this book, we will adhere to tradition and measure line speeds in megabits/sec (Mbps), not megabytes/sec (MB/sec). A megabit is 1,000,000 bits, not 1,048,576 (2^{20}) bits.

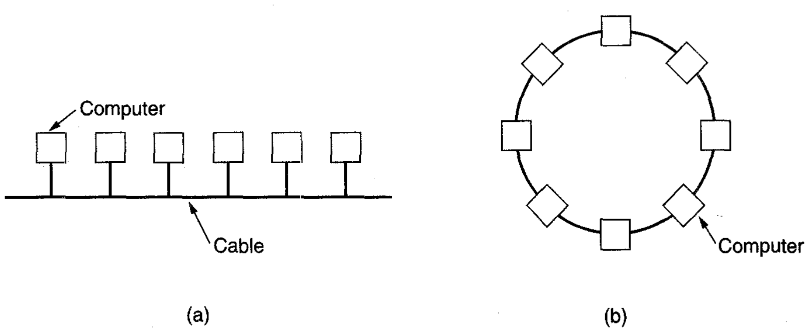

Fig. 1-3. Two broadcast networks. (a) Bus. (b) Ring.

Various topologies are possible for broadcast LANs. Figure 1-3 shows two of them. In a bus (i.e., a linear cable) network, at any instant one machine is the master and is allowed to transmit. All other machines are required to refrain from sending. An arbitration mechanism is needed to resolve conflicts when two or more machines want to transmit simultaneously. The arbitration mechanism may be centralized or distributed. IEEE 802.3, popularly called **Ethernet**™, for example, is a bus-based broadcast network with decentralized control operating at 10 or 100 Mbps. Computers on an Ethernet can transmit whenever they want to; if two or more packets collide, each computer just waits a random time and tries again later.

A second type of broadcast system is the ring. In a ring, each bit propagates around on its own, not waiting for the rest of the packet to which it belongs. Typically, each bit circumnavigates the entire ring in the time it takes to transmit a few bits, often before the complete packet has even been transmitted. Like all other broadcast systems, some rule is needed for arbitrating simultaneous accesses to the ring. Various methods are in use and will be discussed later in this book. IEEE 802.5 (the IBM token ring), is a popular ring-based LAN operating at 4 and 16 Mbps.

Broadcast networks can be further divided into static and dynamic, depending on how the channel is allocated. A typical static allocation would be to divide up time into discrete intervals and run a round robin algorithm, allowing each machine to broadcast only when its time slot comes up. Static allocation wastes channel capacity when a machine has nothing to say during its allocated slot, so most systems attempt to allocate the channel dynamically (i.e., on demand).

Dynamic allocation methods for a common channel are either centralized or decentralized. In the centralized channel allocation method, there is a single entity, for example a bus arbitration unit, which determines who goes next. It might do this by accepting requests and making a decision according to some internal algorithm. In the decentralized channel allocation method, there is no central entity; each machine must decide for itself whether or not to transmit. You might think that this always leads to chaos, but it does not. Later we will study many algorithms designed to bring order out of the potential chaos.

The other kind of LAN is built using point-to-point lines. Individual lines connect a specific machine with another specific machine. Such a LAN is really a miniature wide area network. We will look at these later.

1.2.2. Metropolitan Area Networks

A **metropolitan area network**, or **MAN** (plural: MANs, not MEN) is basically a bigger version of a LAN and normally uses similar technology. It might cover a group of nearby corporate offices or a city and might be either private or public. A MAN can support both data and voice, and might even be related to the local cable television network. A MAN just has one or two cables and does not contain switching elements, which shunt packets over one of several potential output lines. Not having to switch simplifies the design.

The main reason for even distinguishing MANs as a special category is that a standard has been adopted for them, and this standard is now being implemented. It is called **DQDB** (**Distributed Queue Dual Bus**) or for people who prefer numbers to letters, 802.6 (the number of the IEEE standard that defines it). DQDB consists of two unidirectional buses (cables) to which all the computers are connected, as shown in Fig. 1-4. Each bus has a head-end, a device that initiates transmission activity. Traffic that is destined for a computer to the right of the sender uses the upper bus. Traffic to the left uses the lower one.

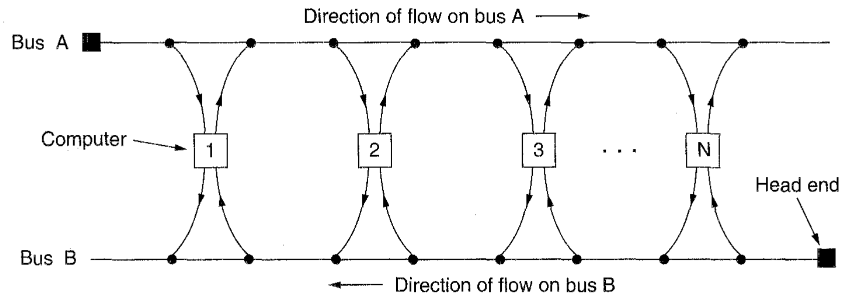

Fig. 1-4. Architecture of the DQDB metropolitan area network.

A key aspect of a MAN is that there is a broadcast medium (for 802.6, two cables) to which all the computers are attached. This greatly simplifies the design compared to other kinds of networks. We will discuss DQDB in more detail in Chap. 4.

1.2.3. Wide Area Networks

A **wide area network**, or **WAN**, spans a large geographical area, often a country or continent. It contains a collection of machines intended for running user (i.e., application) programs. We will follow traditional usage and call these machines **hosts**. The term **end system** is sometimes also used in the literature. The hosts are connected by a **communication subnet**, or just **subnet** for short. The job of the subnet is to carry messages from host to host, just as the telephone system carries words from speaker to listener. By separating the pure communication aspects of the network (the subnet) from the application aspects (the hosts), the complete network design is greatly simplified.

In most wide area networks, the subnet consists of two distinct components: transmission lines and switching elements. Transmission lines (also called **circuits**, **channels**, or **trunks**) move bits between machines.

The switching elements are specialized computers used to connect two or more transmission lines. When data arrive on an incoming line, the switching element must choose an outgoing line to forward them on. Unfortunately, there is no standard terminology used to name these computers. They are variously called **packet switching nodes**, **intermediate systems**, and **data switching exchanges**, among other things. As a generic term for the switching computers, we will use the word **router**, but the reader should be aware that no consensus on terminology exists here. In this model, shown in Fig. 1-5, each host is generally connected to a LAN on which a router is present, although in some cases a host can be connected directly to a router. The collection of communication lines and routers (but not the hosts) form the subnet.

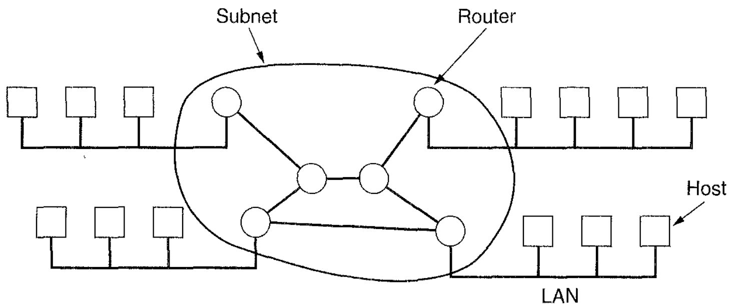

Fig. 1-5. Relation between hosts and the subnet.

An aside about the term "subnet" is worth making. Originally, its only meaning was the collection of routers and communication lines that moved packets from the source host to the destination host. However, some years later, it also acquired a second meaning in conjunction with network addressing (which we will discuss in Chap. 5). Hence the term has a certain ambiguity about it. Unfortunately, no widely-used alternative exists for its initial meaning, so with some hesitation we will use it in both senses. From the context, it will always be clear which is meant.

In most WANs, the network contains numerous cables or telephone lines, each one connecting a pair of routers. If two routers that do not share a cable nevertheless wish to communicate, they must do this indirectly, via other routers. When a packet is sent from one router to another via one or more intermediate routers, the packet is received at each intermediate router in its entirety, stored there until the required output line is free, and then forwarded. A subnet using this principle is called a **point-to-point**, **store-and-forward**, or **packet-switched** subnet. Nearly all wide area networks (except those using satellites) have store-and-forward subnets. When the packets are small and all the same size, they are often called **cells**.

When a point-to-point subnet is used, an important design issue is what the router interconnection topology should look like. Figure 1-6 shows several possible topologies. Local networks that were designed as such usually have a symmetric topology. In contrast, wide area networks typically have irregular topologies.

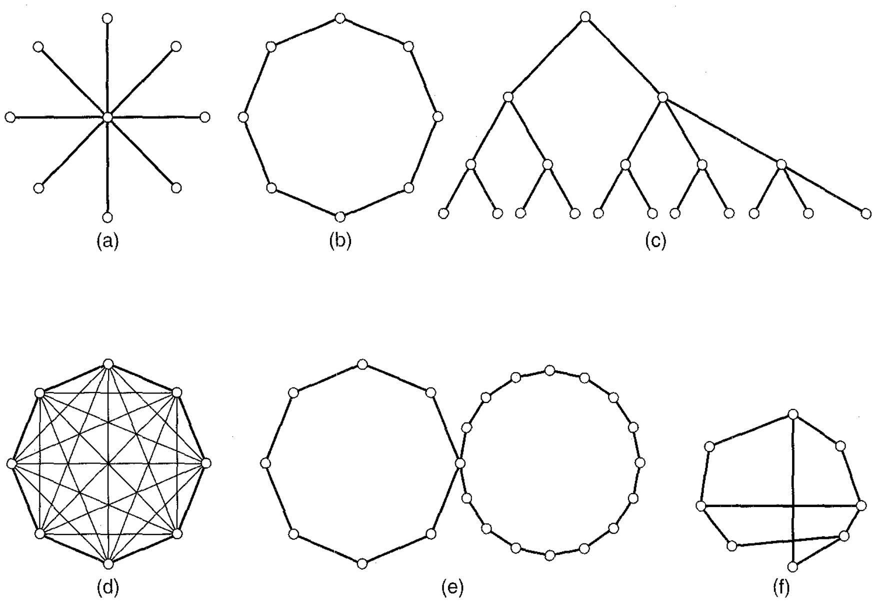

Fig. 1-6. Some possible topologies for a point-to-point subnet. (a) Star. (b) Ring. (c) Tree. (d) Complete. (e) Intersecting rings. (f) Irregular.

A second possibility for a WAN is a satellite or ground radio system. Each router has an antenna through which it can send and receive. All routers can hear the output *from* the satellite, and in some cases they can also hear the upward transmissions of their fellow routers *to* the satellite as well. Sometimes the routers are connected to a substantial point-to-point subnet, with only some of them having a satellite antenna. Satellite networks are inherently broadcast and are most useful when the broadcast property is important.

1.2.4. Wireless Networks

Mobile computers, such as notebook computers and personal digital assistants (PDAs), are the fastest-growing segment of the computer industry. Many of the owners of these computers have desktop machines on LANs and WANs back at the office and want to be connected to their home base even when away from home or en route. Since having a wired connection is impossible in cars and airplanes, there is a lot of interest in wireless networks. In this section we will

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briefly introduce this topic. (Note: by section, we mean those portions of the book with a three-part number such as 1.2.4.)

Actually, digital wireless communication is not a new idea. As early as 1901, the Italian physicist Guglielmo Marconi demonstrated a ship-to-shore wireless telegraph using Morse Code (dots and dashes are binary, after all). Modern digital wireless systems have better performance, but the basic idea is the same. Additional information about these systems can be found in (Garg and Wilkes, 1996; and Pahlavan et al., 1995).

Wireless networks have many uses. A common one is the portable office. People on the road often want to use their portable electronic equipment to send and receive telephone calls, faxes, and electronic mail, read remote files, login on remote machines, and so on, and do this from anywhere on land, sea, or air.

Wireless networks are of great value to fleets of trucks, taxis, buses, and repairpersons for keeping in contact with home. Another use is for rescue workers at disaster sites (fires, floods, earthquakes, etc.) where the telephone system has been destroyed. Computers there can send messages, keep records, and so on.

Finally, wireless networks are important to the military. If you have to be able to fight a war anywhere on earth on short notice, counting on using the local networking infrastructure is probably not a good idea. It is better to bring your own.

Although wireless networking and mobile computing are often related, they are not identical, as Fig. 1-7 shows. Portable computers are sometimes wired. For example, if a traveler plugs a portable computer into the telephone jack in a hotel, we have mobility without a wireless network. Another example is someone carrying a portable computer along as he inspects a train for technical problems. Here a long cord can trail along behind (vacuum cleaner model).

Wireless	Mobile	Applications
No	No	Stationary workstations in offices
No	Yes	Using a portable in a hotel; train maintenance
Yes	No	LANs in older, unwired buildings
Yes	Yes	Portable office; PDA for store inventory

Fig. 1-7. Combinations of wireless networks and mobile computing.

On the other hand, some wireless computers are not portable. An important example here is a company that owns an older building that does not have network cabling installed and wants to connect its computers. Installing a wireless LAN may require little more than buying a small box with some electronics and setting up some antennas. This solution may be cheaper than wiring the building.

Although wireless LANs are easy to install, they also have some disadvantages. Typically they have a capacity of 1–2 Mbps, which is much slower than

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wired LANs. The error rates are often much higher, too, and the transmissions from different computers can interfere with one another.

But of course, there are also the true mobile, wireless applications, ranging from the portable office to people walking around a store with a PDA doing inventory. At many busy airports, car rental return clerks work out in the parking lot with wireless portable computers. They type in the license plate number of returning cars, and their portable, which has a built-in printer, calls the main computer, gets the rental information, and prints out the bill on the spot. True mobile computing is discussed further in (Forman and Zahorjan, 1994).

Wireless networks come in many forms. Some universities are already installing antennas all over campus to allow students to sit under the trees and consult the library's card catalog. Here the computers communicate directly with the wireless LAN in digital form. Another possibility is using a cellular (i.e., portable) telephone with a traditional analog modem. Direct digital cellular service, called **CDPD** (**Cellular Digital Packet Data**) is becoming available in many cities. We will study it in Chap. 4.

Finally, it is possible to have different combinations of wired and wireless networking. For example, in Fig. 1-8(a), we depict an airplane with a number of people using modems and seat-back telephones to call the office. Each call is independent of the other ones. A much more efficient option, however, is the flying LAN of Fig. 1-8(b). Here each seat comes equipped with an Ethernet connector into which passengers can plug their computers. A single router on the aircraft maintains a radio link with some router on the ground, changing routers as it flies along. This configuration is just a traditional LAN, except that its connection to the outside world happens to be a radio link instead of a hardwired line.

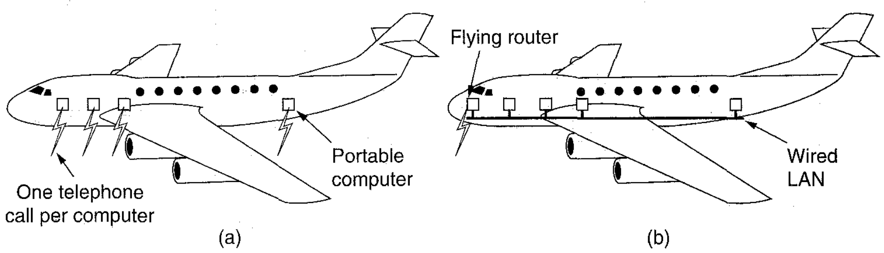

Fig. 1-8. (a) Individual mobile computers. (b) A flying LAN.

While many people believe that wireless portable computers are the wave of the future, at least one dissenting voice has been heard. Bob Metcalfe, the inventor of Ethernet, has written: "Mobile wireless computers are like mobile pipeless bathrooms—portapotties. They will be common on vehicles, and at construction sites, and rock concerts. My advice is to wire up your home and stay there" (Metcalfe, 1995). Will most people follow Metcalfe's advice? Time will tell.

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1.2.5. Internetworks

Many networks exist in the world, often with different hardware and software. People connected to one network often want to communicate with people attached to a different one. This desire requires connecting together different, and frequently incompatible networks, sometimes by using machines called **gateways** to make the connection and provide the necessary translation, both in terms of hardware and software. A collection of interconnected networks is called an **internetwork** or just **internet**.

A common form of internet is a collection of LANs connected by a WAN. In fact, if we were to replace the label "subnet" in Fig. 1-5 by "WAN," nothing else in the figure would have to change. The only real distinction between a subnet and a WAN in this case is whether or not hosts are present. If the system within the closed curve contains only routers, it is a subnet. If it contains both routers and hosts with their own users, it is a WAN.

To avoid confusion, please note that the word "internet" will always be used in this book in a generic sense. In contrast, the **Internet** (note uppercase I) means a specific worldwide internet that is widely used to connect universities, government offices, companies, and of late, private individuals. We will have much to say about both internets and the Internet later in this book.

Subnets, networks, and internetworks are often confused. Subnet makes the most sense in the context of a wide area network, where it refers to the collection of routers and communication lines owned by the network operator, for example, companies like America Online and CompuServe. As an analogy, the telephone system consists of telephone switching offices connected to each other by high-speed lines, and to houses and businesses by low-speed lines. These lines and equipment, owned and managed by the telephone company, form the subnet of the telephone system. The telephones themselves (the hosts in this analogy) are not part of the subnet. The combination of a subnet and its hosts forms a network. In the case of a LAN, the cable and the hosts form the network. There really is no subnet.

An internetwork is formed when distinct networks are connected together. In our view, connecting a LAN and a WAN or connecting two LANs forms an internetwork, but there is little agreement in the industry over terminology in this area.

1.3. NETWORK SOFTWARE

The first computer networks were designed with the hardware as the main concern and the software as an afterthought. This strategy no longer works. Network software is now highly structured. In the following sections we examine the software structuring technique in some detail. The method described here forms the keystone of the entire book and will occur repeatedly later on.

1.3.1. Protocol Hierarchies

To reduce their design complexity, most networks are organized as a series of **layers** or **levels**, each one built upon the one below it. The number of layers, the name of each layer, the contents of each layer, and the function of each layer differ from network to network. However, in all networks, the purpose of each layer is to offer certain services to the higher layers, shielding those layers from the details of how the offered services are actually implemented.

Layer *n* on one machine carries on a conversation with layer *n* on another machine. The rules and conventions used in this conversation are collectively known as the layer *n* **protocol**. Basically, a protocol is an agreement between the communicating parties on how communication is to proceed. As an analogy, when a woman is introduced to a man, she may choose to stick out her hand. He, in turn, may decide either to shake it or kiss it, depending, for example, on whether she is an American lawyer at a business meeting or a European princess at a formal ball. Violating the protocol will make communication more difficult, if not impossible.

A five-layer network is illustrated in Fig. 1-9. The entities comprising the corresponding layers on different machines are called **peers**. In other words, it is the peers that communicate using the protocol.

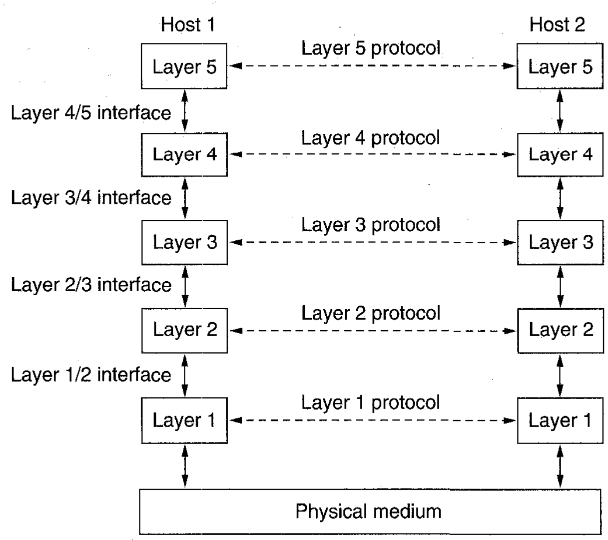

Fig. 1-9. Layers, protocols, and interfaces.

In reality, no data are directly transferred from layer *n* on one machine to layer *n* on another machine. Instead, each layer passes data and control information to the layer immediately below it, until the lowest layer is reached. Below layer 1 is the **physical medium** through which actual communication occurs. In Fig. 1-9, virtual communication is shown by dotted lines and physical communication by solid lines.

Between each pair of adjacent layers there is an **interface**. The interface defines which primitive operations and services the lower layer offers to the upper one. When network designers decide how many layers to include in a network and what each one should do, one of the most important considerations is defining clean interfaces between the layers. Doing so, in turn, requires that each layer perform a specific collection of well-understood functions. In addition to minimizing the amount of information that must be passed between layers, clean-cut interfaces also make it simpler to replace the implementation of one layer with a completely different implementation (e.g., all the telephone lines are replaced by satellite channels), because all that is required of the new implementation is that it offers exactly the same set of services to its upstairs neighbor as the old implementation did.

A set of layers and protocols is called a **network architecture**. The specification of an architecture must contain enough information to allow an implementer to write the program or build the hardware for each layer so that it will correctly obey the appropriate protocol. Neither the details of the implementation nor the specification of the interfaces are part of the architecture because these are hidden away inside the machines and not visible from the outside. It is not even necessary that the interfaces on all machines in a network be the same, provided that each machine can correctly use all the protocols. A list of protocols used by a certain system, one protocol per layer, is called a **protocol stack**. The subjects of network architectures, protocol stacks, and the protocols themselves are the principal topics of this book.

An analogy may help explain the idea of multilayer communication. Imagine two philosophers (peer processes in layer 3), one of whom speaks Urdu and English and one of whom speaks Chinese and French. Since they have no common language, they each engage a translator (peer processes at layer 2), each of whom in turn contacts a secretary (peer processes in layer 1). Philosopher 1 wishes to convey his affection for *oryctolagus cuniculus* to his peer. To do so, he passes a message (in English) across the 2/3 interface, to his translator, saying "I like rabbits," as illustrated in Fig. 1-10. The translators have agreed on a neutral language, Dutch, so the message is converted to "Ik hou van konijnen." The choice of language is the layer 2 protocol and is up to the layer 2 peer processes.

The translator then gives the message to a secretary for transmission, by, for example, fax (the layer 1 protocol). When the message arrives, it is translated into French and passed across the 2/3 interface to philosopher 2. Note that each protocol is completely independent of the other ones as long as the interfaces are not changed. The translators can switch from Dutch to say, Finnish, at will, provided that they both agree, and neither changes his interface with either layer 1 or

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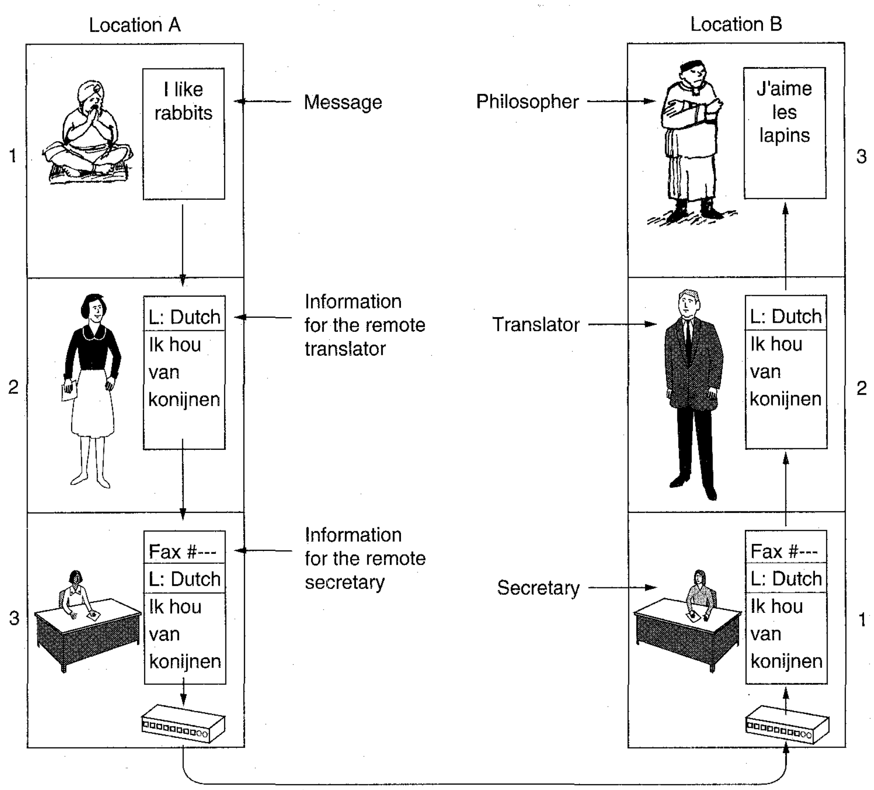

Fig. 1-10. The philosopher-translator-secretary architecture.

layer 3. Similarly the secretaries can switch from fax to email, or telephone without disturbing (or even informing) the other layers. Each process may add some information intended only for its peer. This information is not passed upward to the layer above.

Now consider a more technical example: how to provide communication to the top layer of the five-layer network in Fig. 1-11. A message, *M*, is produced by an application process running in layer 5 and given to layer 4 for transmission. Layer 4 puts a **header** in front of the message to identify the message and passes the result to layer 3. The header includes control information, such as sequence numbers, to allow layer 4 on the destination machine to deliver messages in the right order if the lower layers do not maintain sequence. In some layers, headers also contain sizes, times, and other control fields.

In many networks, there is no limit to the size of messages transmitted in the layer 4 protocol, but there is nearly always a limit imposed by the layer 3 protocol. Consequently, layer 3 must break up the incoming messages into smaller

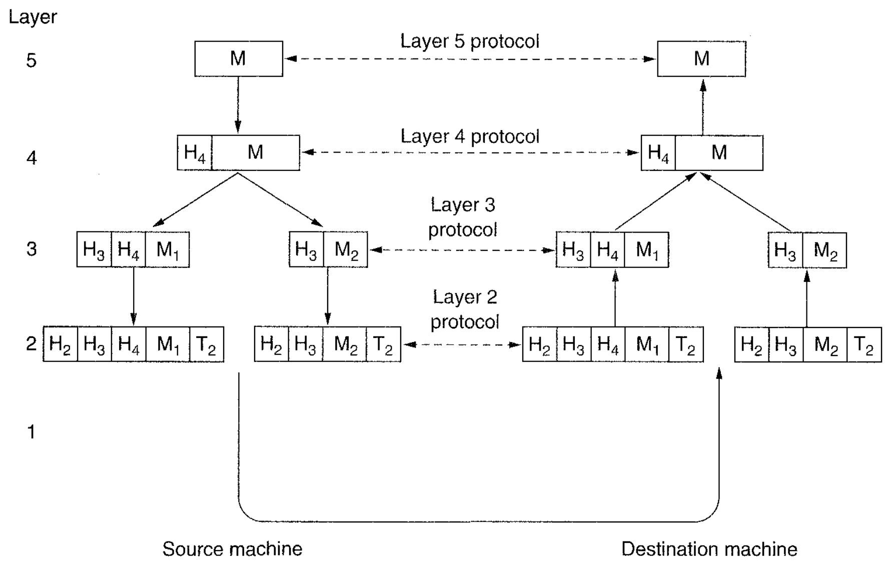

Fig. 1-11. Example information flow supporting virtual communication in layer 5.

units, packets, prepending a layer 3 header to each packet. In this example, M is split into two parts, M_1 and M_2.

Layer 3 decides which of the outgoing lines to use and passes the packets to layer 2. Layer 2 adds not only a header to each piece, but also a trailer, and gives the resulting unit to layer 1 for physical transmission. At the receiving machine the message moves upward, from layer to layer, with headers being stripped off as it progresses. None of the headers for layers below n are passed up to layer n.

The important thing to understand about Fig. 1-11 is the relation between the virtual and actual communication and the difference between protocols and interfaces. The peer processes in layer 4, for example, conceptually think of their communication as being "horizontal," using the layer 4 protocol. Each one is likely to have a procedure called something like *SendToOtherSide* and *GetFromOtherSide*, even though these procedures actually communicate with lower layers across the 3/4 interface, not with the other side.

The peer process abstraction is crucial to all network design. Using it, the unmanageable task of designing the complete network can be broken into several smaller, manageable, design problems, namely the design of the individual layers.

Although Section 1-3 is called "Network Software," it is worth pointing out that the lower layers of a protocol hierarchy are frequently implemented in hardware or firmware. Nevertheless, complex protocol algorithms are involved, even if they are embedded (in whole or in part) in hardware.

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1.3.2. Design Issues for the Layers

Some of the key design issues that occur in computer networking are present in several layers. Below, we will briefly mention some of the more important ones.

Every layer needs a mechanism for identifying senders and receivers. Since a network normally has many computers, some of which have multiple processes, a means is needed for a process on one machine to specify with whom it wants to talk. As a consequence of having multiple destinations, some form of addressing is needed in order to specify a specific destination.

Another set of design decisions concerns the rules for data transfer. In some systems, data only travel in one direction (**simplex communication**). In others they can travel in either direction, but not simultaneously (**half-duplex communication**). In still others they travel in both directions at once (**full-duplex communication**). The protocol must also determine how many logical channels the connection corresponds to, and what their priorities are. Many networks provide at least two logical channels per connection, one for normal data and one for urgent data.

Error control is an important issue because physical communication circuits are not perfect. Many error-detecting and error-correcting codes are known, but both ends of the connection must agree on which one is being used. In addition, the receiver must have some way of telling the sender which messages have been correctly received and which have not.

Not all communication channels preserve the order of messages sent on them. To deal with a possible loss of sequencing, the protocol must make explicit provision for the receiver to allow the pieces to be put back together properly. An obvious solution is to number the pieces, but this solution still leaves open the question of what should be done with pieces that arrive out of order.

An issue that occurs at every level is how to keep a fast sender from swamping a slow receiver with data. Various solutions have been proposed and will be discussed later. Some of them involve some kind of feedback from the receiver to the sender, either directly or indirectly, about the receiver's current situation. Others limit the sender to an agreed upon transmission rate.

Another problem that must be solved at several levels is the inability of all processes to accept arbitrarily long messages. This property leads to mechanisms for disassembling, transmitting, and then reassembling messages. A related issue is what to do when processes insist upon transmitting data in units that are so small that sending each one separately is inefficient. Here the solution is to gather together several small messages heading toward a common destination into a single large message and dismember the large message at the other side.

When it is inconvenient or expensive to set up a separate connection for each pair of communicating processes, the underlying layer may decide to use the same connection for multiple, unrelated conversations. As long as this multiplexing and demultiplexing is done transparently, it can be used by any layer. Multiplexing is needed in the physical layer, for example, where all the traffic for all connections has to be sent over at most a few physical circuits.

When there are multiple paths between source and destination, a route must be chosen. Sometimes this decision must be split over two or more layers. For example, to send data from London to Rome, a high-level decision might have to be made to go via France or Germany based on their respective privacy laws, and a low-level decision might have to be made to choose one of the many available circuits based on the current traffic load.

1.3.3. Interfaces and Services

The function of each layer is to provide services to the layer above it. In this section we will look at precisely what a service is in more detail, but first we will give some terminology.

The active elements in each layer are often called **entities**. An entity can be a software entity (such as a process), or a hardware entity (such as an intelligent I/O chip). Entities in the same layer on different machines are called **peer entities**. The entities in layer n implement a service used by layer $n + 1$. In this case layer n is called the **service provider** and layer $n + 1$ is called the **service user**. Layer n may use the services of layer $n - 1$ in order to provide its service. It may offer several classes of service, for example, fast, expensive communication and slow, cheap communication.

Services are available at **SAP**s (**Service Access Points**), The layer n SAPs are the places where layer $n + 1$ can access the services offered. Each SAP has an address that uniquely identifies it. To make this point clearer, the SAPs in the telephone system are the sockets into which modular telephones can be plugged, and the SAP addresses are the telephone numbers of these sockets. To call someone, you must know the callee's SAP address. Similarly, in the postal system, the SAP addresses are street addresses and post office box numbers. To send a letter, you must know the addressee's SAP address.

In order for two layers to exchange information, there has to be an agreed upon set of rules about the interface. At a typical interface, the layer $n + 1$ entity passes an **IDU** (**Interface Data Unit**) to the layer n entity through the SAP as shown in Fig. 1-12. The IDU consists of an **SDU** (**Service Data Unit**) and some control information. The SDU is the information passed across the network to the peer entity and then up to layer $n + 1$. The control information is needed to help the lower layer do its job (e.g., the number of bytes in the SDU) but is not part of the data itself.

In order to transfer the SDU, the layer n entity may have to fragment it into several pieces, each of which is given a header and sent as a separate **PDU** (**Protocol Data Unit**) such as a packet. The PDU headers are used by the peer entities

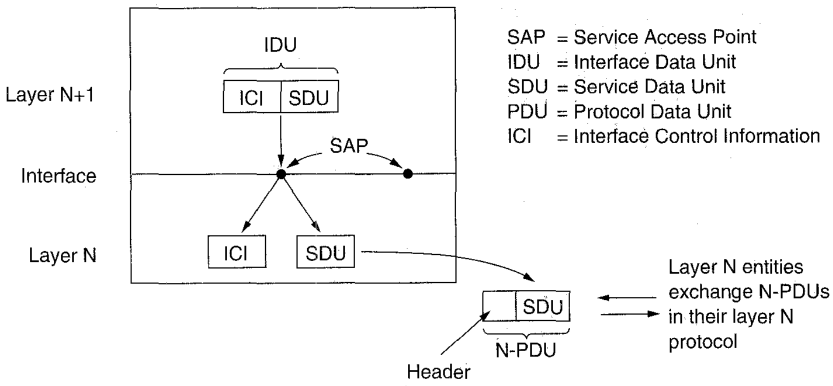

Fig. 1-12. Relation between layers at an interface.

to carry out their peer protocol. They identify which PDUs contain data and which contain control information, provide sequence numbers and counts, and so on.

1.3.4. Connection-Oriented and Connectionless Services

Layers can offer two different types of service to the layers above them: connection-oriented and connectionless. In this section we will look at these two types and examine the differences between them.

Connection-oriented service is modeled after the telephone system. To talk to someone, you pick up the phone, dial the number, talk, and then hang up. Similarly, to use a connection-oriented network service, the service user first establishes a connection, uses the connection, and then releases the connection. The essential aspect of a connection is that it acts like a tube: the sender pushes objects (bits) in at one end, and the receiver takes them out in the same order at the other end.

In contrast, **connectionless service** is modeled after the postal system. Each message (letter) carries the full destination address, and each one is routed through the system independent of all the others. Normally, when two messages are sent to the same destination, the first one sent will be the first one to arrive. However, it is possible that the first one sent can be delayed so that the second one arrives first. With a connection-oriented service this is impossible.

Each service can be characterized by a **quality of service**. Some services are reliable in the sense that they never lose data. Usually, a reliable service is implemented by having the receiver acknowledge the receipt of each message, so the sender is sure that it arrived. The acknowledgement process introduces overhead and delays, which are often worth it but are sometimes undesirable.

A typical situation in which a reliable connection-oriented service is appropriate is file transfer. The owner of the file wants to be sure that all the bits arrive correctly and in the same order they were sent. Very few file transfer customers would prefer a service that occasionally scrambles or loses a few bits, even if it is much faster.

Reliable connection-oriented service has two minor variations: message sequences and byte streams. In the former, the message boundaries are preserved. When two 1-KB messages are sent, they arrive as two distinct 1-KB messages, never as one 2-KB message. (Note: KB means kilobytes; kb means kilobits.) In the latter, the connection is simply a stream of bytes, with no message boundaries. When 2K bytes arrive at the receiver, there is no way to tell if they were sent as one 2-KB message, two 1-KB messages, or 2048 1-byte messages. If the pages of a book are sent over a network to a phototypesetter as separate messages, it might be important to preserve the message boundaries. On the other hand, with a terminal logging into a remote timesharing system, a byte stream from the terminal to the computer is all that is needed.

As mentioned above, for some applications, the delays introduced by acknowledgements are unacceptable. One such application is digitized voice traffic. It is preferable for telephone users to hear a bit of noise on the line or a garbled word from time to time than to introduce a delay to wait for acknowledgements. Similarly, when transmitting a video film, having a few pixels wrong is no problem, but having the film jerk along as the flow stops to correct errors is very irritating.

Not all applications require connections. For example, as electronic mail becomes more common, can electronic junk mail be far behind? The electronic junk mail sender probably does not want to go to the trouble of setting up and later tearing down a connection just to send one item. Nor is 100 percent reliable delivery essential, especially if it costs more. All that is needed is a way to send a single message that has a high probability of arrival, but no guarantee. Unreliable (meaning not acknowledged) connectionless service is often called **datagram service**, in analogy with telegram service, which also does not provide an acknowledgement back to the sender.

In other situations, the convenience of not having to establish a connection to send one short message is desired, but reliability is essential. The **acknowledged datagram service** can be provided for these applications. It is like sending a registered letter and requesting a return receipt. When the receipt comes back, the sender is absolutely sure that the letter was delivered to the intended party and not lost along the way.

Still another service is the **request-reply service**. In this service the sender transmits a single datagram containing a request; the reply contains the answer. For example, a query to the local library asking where Uighur is spoken falls into this category. Request-reply is commonly used to implement communication in the client-server model: the client issues a request and the server responds to it. Figure 1-13 summarizes the types of services discussed above.

	Service	Example
Connection-oriented	Reliable message stream	Sequence of pages
	Reliable byte stream	Remote login
	Unreliable connection	Digitized voice
Connection-less	Unreliable datagram	Electronic junk mail
	Acknowledged datagram	Registered mail
	Request-reply	Database query

Fig. 1-13. Six different types of service.

1.3.5. Service Primitives

A service is formally specified by a set of **primitives** (operations) available to a user or other entity to access the service. These primitives tell the service to perform some action or report on an action taken by a peer entity. One way to classify the service primitives is to divide them into four classes as shown in Fig. 1-14.

Primitive	Meaning
Request	An entity wants the service to do some work
Indication	An entity is to be informed about an event
Response	An entity wants to respond to an event
Confirm	The response to an earlier request has come back

Fig. 1-14. Four classes of service primitives.

To illustrate the uses of the primitives, consider how a connection is established and released. The initiating entity does a CONNECT.request which results in a packet being sent. The receiver then gets a CONNECT.indication announcing that an entity somewhere wants to set up a connection to it. The entity getting the CONNECT.indication then uses the CONNECT.response primitive to tell whether it wants to accept or reject the proposed connection. Either way, the entity issuing the initial CONNECT.request finds out what happened via a CONNECT.confirm primitive.

Primitives can have parameters, and most of them do. The parameters to a CONNECT.request might specify the machine to connect to, the type of service desired, and the maximum message size to be used on the connection. The parameters to a CONNECT.indication might contain the caller's identity, the type of

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service desired, and the proposed maximum message size. If the called entity did not agree to the proposed maximum message size, it could make a counterproposal in its *response* primitive, which would be made available to the original caller in the *confirm*. The details of this **negotiation** are part of the protocol. For example, in the case of two conflicting proposals about maximum message size, the protocol might specify that the smaller value is always chosen.

As an aside on terminology, we will carefully avoid the terms "open a connection" and "close a connection" because to electrical engineers, an "open circuit" is one with a gap or break in it. Electricity can only flow over "closed circuits." Computer scientists would never agree to having information flow over a closed circuit. To keep both camps pacified, we will use the terms "establish a connection" and "release a connection."

Services can be either **confirmed** or **unconfirmed**. In a confirmed service, there is a *request*, an *indication*, a *response*, and a *confirm*. In an unconfirmed service, there is just a *request* and an *indication*. CONNECT is always a confirmed service because the remote peer must agree to establish a connection. Data transfer, on the other hand, can be either confirmed or unconfirmed, depending on whether or not the sender needs an acknowledgement. Both kinds of services are used in networks.

To make the concept of a service more concrete, let us consider as an example a simple connection-oriented service with eight service primitives as follows:

1. CONNECT.request – Request a connection to be established.
2. CONNECT.indication – Signal the called party.
3. CONNECT.response – Used by the callee to accept/reject calls.
4. CONNECT.confirm – Tell the caller whether the call was accepted.
5. DATA.request – Request that data be sent.
6. DATA.indication – Signal the arrival of data.
7. DISCONNECT.request – Request that a connection be released.
8. DISCONNECT.indication – Signal the peer about the request.

In this example, CONNECT is a confirmed service (an explicit response is required), whereas DISCONNECT is unconfirmed (no response).

It may be helpful to make an analogy with the telephone system to see how these primitives are used. For this analogy, consider the steps required to call Aunt Millie on the telephone and invite her to your house for tea.

1. CONNECT.request – Dial Aunt Millie's phone number.
2. CONNECT.indication – Her phone rings.
3. CONNECT.response – She picks up the phone.

4. CONNECT.confirm – You hear the ringing stop.
5. DATA.request – You invite her to tea.
6. DATA.indication – She hears your invitation.
7. DATA.request – She says she would be delighted to come.
8. DATA.indication – You hear her acceptance.
9. DISCONNECT.request – You hang up the phone.
10. DISCONNECT.indication – She hears it and hangs up too.

Figure 1-15 shows this same sequence of steps as a series of service primitives, including the final confirmation of disconnection. Each step involves an interaction between two layers on one of the computers. Each *request* or *response* causes an *indication* or *confirm* at the other side a little later. In this example, the service users (you and Aunt Millie) are in layer $N + 1$ and the service provider (the telephone system) is in layer N.

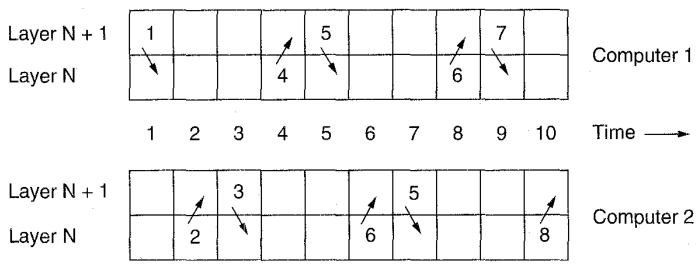

Fig. 1-15. How a computer would invite its Aunt Millie to tea. The numbers near the tail end of each arrow refer to the eight service primitives discussed in this section.

1.3.6. The Relationship of Services to Protocols

Services and protocols are distinct concepts, although they are frequently confused. This distinction is so important, however, that we emphasize it again here. A *service* is a set of primitives (operations) that a layer provides to the layer above it. The service defines what operations the layer is prepared to perform on behalf of its users, but it says nothing at all about how these operations are implemented. A service relates to an interface between two layers, with the lower layer being the service provider and the upper layer being the service user.

A *protocol*, in contrast, is a set of rules governing the format and meaning of the frames, packets, or messages that are exchanged by the peer entities within a layer. Entities use protocols in order to implement their service definitions. They

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are free to change their protocols at will, provided they do not change the service visible to their users. In this way, the service and the protocol are completely decoupled.

An analogy with programming languages is worth making. A service is like an abstract data type or an object in an object-oriented language. It defines operations that can be performed on an object but does not specify how these operations are implemented. A protocol relates to the *implementation* of the service and as such is not visible to the user of the service.

Many older protocols did not distinguish the service from the protocol. In effect, a typical layer might have had a service primitive SEND PACKET with the user providing a pointer to a fully assembled packet. This arrangement meant that all changes to the protocol were immediately visible to the users. Most network designers now regard such a design as a serious blunder.

1.4. REFERENCE MODELS

Now that we have discussed layered networks in the abstract, it is time to look at some examples. In the next two sections we will discuss two important network architectures, the OSI reference model and the TCP/IP reference model.

1.4.1. The OSI Reference Model

The OSI model is shown in Fig. 1-16 (minus the physical medium). This model is based on a proposal developed by the International Standards Organization (ISO) as a first step toward international standardization of the protocols used in the various layers (Day and Zimmermann, 1983). The model is called the **ISO OSI (Open Systems Interconnection) Reference Model** because it deals with connecting open systems—that is, systems that are open for communication with other systems. We will usually just call it the OSI model for short.

The OSI model has seven layers. The principles that were applied to arrive at the seven layers are as follows:

1. A layer should be created where a different level of abstraction is needed.
2. Each layer should perform a well defined function.
3. The function of each layer should be chosen with an eye toward defining internationally standardized protocols.
4. The layer boundaries should be chosen to minimize the information flow across the interfaces.
5. The number of layers should be large enough that distinct functions need not be thrown together in the same layer out of necessity, and small enough that the architecture does not become unwieldy.

Below we will discuss each layer of the model in turn, starting at the bottom layer. Note that the OSI model itself is not a network architecture because it does not specify the exact services and protocols to be used in each layer. It just tells what each layer should do. However, ISO has also produced standards for all the layers, although these are not part of the reference model itself. Each one has been published as a separate international standard.

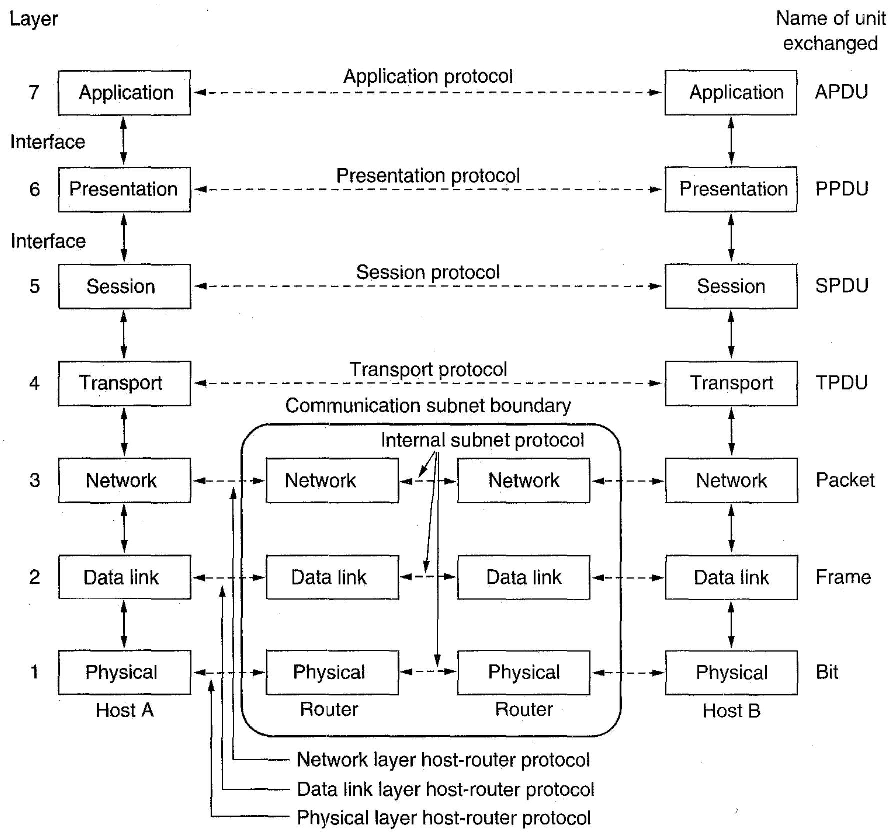

Fig. 1-16. The OSI reference model.

The Physical Layer

The **physical layer** is concerned with transmitting raw bits over a communication channel. The design issues have to do with making sure that when one side sends a 1 bit, it is received by the other side as a 1 bit, not as a 0 bit. Typical questions here are how many volts should be used to represent a 1 and how many for a 0, how many microseconds a bit lasts, whether transmission may proceed simultaneously in both directions, how the initial connection is established and how it is torn down when both sides are finished, and how many pins the network connector has and what each pin is used for. The design issues here largely deal with mechanical, electrical, and procedural interfaces, and the physical transmission medium, which lies below the physical layer.

The Data Link Layer

The main task of the **data link layer** is to take a raw transmission facility and transform it into a line that appears free of undetected transmission errors to the network layer. It accomplishes this task by having the sender break the input data up into **data frames** (typically a few hundred or a few thousand bytes), transmit the frames sequentially, and process the **acknowledgement frames** sent back by the receiver. Since the physical layer merely accepts and transmits a stream of bits without any regard to meaning or structure, it is up to the data link layer to create and recognize frame boundaries. This can be accomplished by attaching special bit patterns to the beginning and end of the frame. If these bit patterns can accidentally occur in the data, special care must be taken to make sure these patterns are not incorrectly interpreted as frame delimiters.

A noise burst on the line can destroy a frame completely. In this case, the data link layer software on the source machine can retransmit the frame. However, multiple transmissions of the same frame introduce the possibility of duplicate frames. A duplicate frame could be sent if the acknowledgement frame from the receiver back to the sender were lost. It is up to this layer to solve the problems caused by damaged, lost, and duplicate frames. The data link layer may offer several different service classes to the network layer, each of a different quality and with a different price.

Another issue that arises in the data link layer (and most of the higher layers as well) is how to keep a fast transmitter from drowning a slow receiver in data. Some traffic regulation mechanism must be employed to let the transmitter know how much buffer space the receiver has at the moment. Frequently, this flow regulation and the error handling are integrated.

If the line can be used to transmit data in both directions, this introduces a new complication that the data link layer software must deal with. The problem is that the acknowledgement frames for A to B traffic compete for the use of the line with data frames for the B to A traffic. A clever solution (piggybacking) has been devised; we will discuss it in detail later.

Broadcast networks have an additional issue in the data link layer: how to control access to the shared channel. A special sublayer of the data link layer, the medium access sublayer, deals with this problem.

The Network Layer

The **network layer** is concerned with controlling the operation of the subnet. A key design issue is determining how packets are routed from source to destination. Routes can be based on static tables that are "wired into" the network and rarely changed. They can also be determined at the start of each conversation, for example a terminal session. Finally, they can be highly dynamic, being determined anew for each packet, to reflect the current network load.

If too many packets are present in the subnet at the same time, they will get in each other's way, forming bottlenecks. The control of such congestion also belongs to the network layer.

Since the operators of the subnet may well expect remuneration for their efforts, there is often some accounting function built into the network layer. At the very least, the software must count how many packets or characters or bits are sent by each customer, to produce billing information. When a packet crosses a national border, with different rates on each side, the accounting can become complicated.

When a packet has to travel from one network to another to get to its destination, many problems can arise. The addressing used by the second network may be different from the first one. The second one may not accept the packet at all because it is too large. The protocols may differ, and so on. It is up to the network layer to overcome all these problems to allow heterogeneous networks to be interconnected.

In broadcast networks, the routing problem is simple, so the network layer is often thin or even nonexistent.

The Transport Layer

The basic function of the **transport layer** is to accept data from the session layer, split it up into smaller units if need be, pass these to the network layer, and ensure that the pieces all arrive correctly at the other end. Furthermore, all this must be done efficiently, and in a way that isolates the upper layers from the inevitable changes in the hardware technology.

Under normal conditions, the transport layer creates a distinct network connection for each transport connection required by the session layer. If the transport connection requires a high throughput, however, the transport layer might create multiple network connections, dividing the data among the network connections to improve throughput. On the other hand, if creating or maintaining a network connection is expensive, the transport layer might multiplex several transport connections onto the same network connection to reduce the cost. In all cases, the transport layer is required to make the multiplexing transparent to the session layer.

The transport layer also determines what type of service to provide the session layer, and ultimately, the users of the network. The most popular type of transport connection is an error-free point-to-point channel that delivers messages or bytes in the order in which they were sent. However, other possible kinds of transport service are transport of isolated messages with no guarantee about the order of delivery, and broadcasting of messages to multiple destinations. The type of service is determined when the connection is established.

The transport layer is a true end-to-end layer, from source to destination. In other words, a program on the source machine carries on a conversation with a similar program on the destination machine, using the message headers and control messages. In the lower layers, the protocols are between each machine and its immediate neighbors, and not by the ultimate source and destination machines, which may be separated by many routers. The difference between layers 1 through 3, which are chained, and layers 4 through 7, which are end-to-end, is illustrated in Fig. 1-16.

Many hosts are multiprogrammed, which implies that multiple connections will be entering and leaving each host. There needs to be some way to tell which message belongs to which connection. The transport header (H_4 in Fig. 1-11) is one place this information can be put.

In addition to multiplexing several message streams onto one channel, the transport layer must take care of establishing and deleting connections across the network. This requires some kind of naming mechanism, so that a process on one machine has a way of describing with whom it wishes to converse. There must also be a mechanism to regulate the flow of information, so that a fast host cannot overrun a slow one. Such a mechanism is called **flow control** and plays a key role in the transport layer (also in other layers). Flow control between hosts is distinct from flow control between routers, although we will later see that similar principles apply to both.

The Session Layer

The session layer allows users on different machines to establish **sessions** between them. A session allows ordinary data transport, as does the transport layer, but it also provides enhanced services useful in some applications. A session might be used to allow a user to log into a remote timesharing system or to transfer a file between two machines.

One of the services of the session layer is to manage dialogue control. Sessions can allow traffic to go in both directions at the same time, or in only one direction at a time. If traffic can only go one way at a time (analogous to a single railroad track), the session layer can help keep track of whose turn it is.

A related session service is **token management**. For some protocols, it is essential that both sides do not attempt the same operation at the same time. To manage these activities, the session layer provides tokens that can be exchanged. Only the side holding the token may perform the critical operation.

Another session service is **synchronization**. Consider the problems that might occur when trying to do a 2-hour file transfer between two machines with a 1-hour mean time between crashes. After each transfer was aborted, the whole transfer would have to start over again and would probably fail again the next time as well. To eliminate this problem, the session layer provides a way to insert checkpoints into the data stream, so that after a crash, only the data transferred after the last checkpoint have to be repeated.

The Presentation Layer

The **presentation layer** performs certain functions that are requested sufficiently often to warrant finding a general solution for them, rather than letting each user solve the problems. In particular, unlike all the lower layers, which are just interested in moving bits reliably from here to there, the presentation layer is concerned with the syntax and semantics of the information transmitted.

A typical example of a presentation service is encoding data in a standard agreed upon way. Most user programs do not exchange random binary bit strings. They exchange things such as people's names, dates, amounts of money, and invoices. These items are represented as character strings, integers, floating-point numbers, and data structures composed of several simpler items. Different computers have different codes for representing character strings (e.g., ASCII and Unicode), integers (e.g., one's complement and two's complement), and so on. In order to make it possible for computers with different representations to communicate, the data structures to be exchanged can be defined in an abstract way, along with a standard encoding to be used "on the wire." The presentation layer manages these abstract data structures and converts from the representation used inside the computer to the network standard representation and back.

The Application Layer

The **application layer** contains a variety of protocols that are commonly needed. For example, there are hundreds of incompatible terminal types in the world. Consider the plight of a full screen editor that is supposed to work over a network with many different terminal types, each with different screen layouts, escape sequences for inserting and deleting text, moving the cursor, etc.

One way to solve this problem is to define an abstract **network virtual terminal** that editors and other programs can be written to deal with. To handle each terminal type, a piece of software must be written to map the functions of the network virtual terminal onto the real terminal. For example, when the editor moves the virtual terminal's cursor to the upper left-hand corner of the screen, this software must issue the proper command sequence to the real terminal to get its cursor there too. All the virtual terminal software is in the application layer.

Another application layer function is file transfer. Different file systems have different file naming conventions, different ways of representing text lines, and so on. Transferring a file between two different systems requires handling these and other incompatibilities. This work, too, belongs to the application layer, as do electronic mail, remote job entry, directory lookup, and various other general-purpose and special-purpose facilities.

Data Transmission in the OSI Model

Figure 1-17 shows an example of how data can be transmitted using the OSI model. The sending process has some data it wants to send to the receiving process. It gives the data to the application layer, which then attaches the application header, *AH* (which may be null), to the front of it and gives the resulting item to the presentation layer.

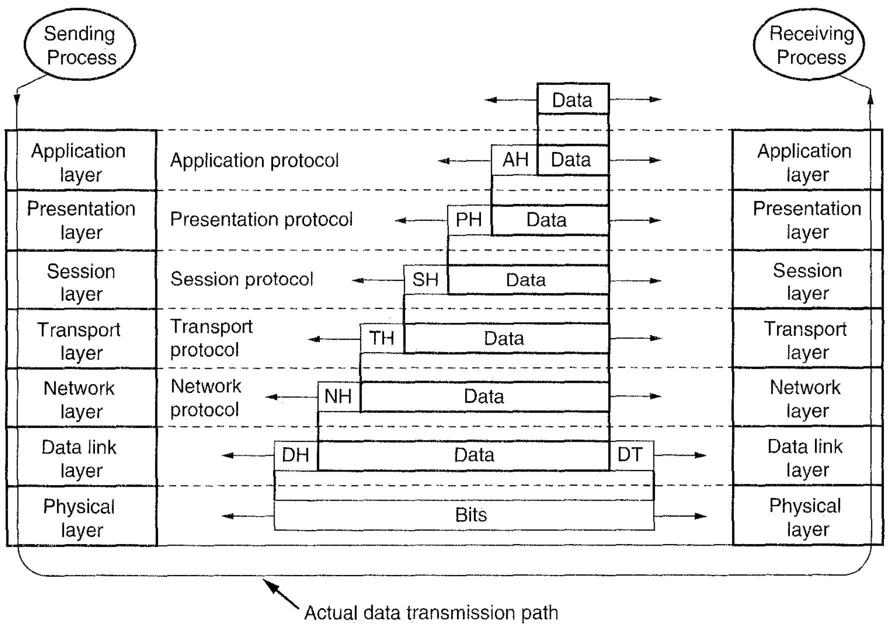

Fig. 1-17. An example of how the OSI model is used. Some of the headers may be null. (Source: H.C. Folts. Used with permission.)

The presentation layer may transform this item in various ways and possibly add a header to the front, giving the result to the session layer. It is important to realize that the presentation layer is not aware of which portion of the data given to it by the application layer is *AH*, if any, and which is true user data.

This process is repeated until the data reach the physical layer, where they are actually transmitted to the receiving machine. On that machine the various headers are stripped off one by one as the message propagates up the layers until it finally arrives at the receiving process.

The key idea throughout is that although actual data transmission is vertical in Fig. 1-17, each layer is programmed as though it were horizontal. When the sending transport layer, for example, gets a message from the session layer, it attaches a transport header and sends it to the receiving transport layer. From its point of view, the fact that it must actually hand the message to the network layer on its own machine is an unimportant technicality. As an analogy, when a Tagalog-speaking diplomat is addressing the United Nations, he thinks of himself as addressing the other assembled diplomats. That, in fact, he is really only speaking to his translator is seen as a technical detail.

1.4.2. The TCP/IP Reference Model

Let us now turn from the OSI reference model to the reference model used in the grandparent of all computer networks, the ARPANET, and its successor, the worldwide Internet. Although we will give a brief history of the ARPANET later, it is useful to mention a few key aspects of it now. The ARPANET was a research network sponsored by the DoD (U.S. Department of Defense). It eventually connected hundreds of universities and government installations using leased telephone lines. When satellite and radio networks were added later, the existing protocols had trouble interworking with them, so a new reference architecture was needed. Thus the ability to connect multiple networks together in a seamless way was one of the major design goals from the very beginning. This architecture later became known as the **TCP/IP Reference Model**, after its two primary protocols. It was first defined in (Cerf and Kahn, 1974). A later perspective is given in (Leiner et al., 1985). The design philosophy behind the model is discussed in (Clark, 1988).

Given the DoD's worry that some of its precious hosts, routers, and internetwork gateways might get blown to pieces at a moment's notice, another major goal was that the network be able to survive loss of subnet hardware, with existing conversations not being broken off. In other words, DoD wanted connections to remain intact as long as the source and destination machines were functioning, even if some of the machines or transmission lines in between were suddenly put out of operation. Furthermore, a flexible architecture was needed, since applications with divergent requirements were envisioned, ranging from transferring files to real-time speech transmission.

The Internet Layer

All these requirements led to the choice of a packet-switching network based on a connectionless internetwork layer. This layer, called the **internet layer**, is the linchpin that holds the whole architecture together. Its job is to permit hosts to inject packets into any network and have them travel independently to the destination (potentially on a different network). They may even arrive in a different order than they were sent, in which case it is the job of higher layers to rearrange them, if in-order delivery is desired. Note that "internet" is used here in a generic sense, even though this layer is present in the Internet.

The analogy here is with the (snail) mail system. A person can drop a sequence of international letters into a mail box in one country, and with a little luck, most of them will be delivered to the correct address in the destination country. Probably the letters will travel through one or more international mail gateways along the way, but this is transparent to the users. Furthermore, that each country (i.e., each network) has its own stamps, preferred envelope sizes, and delivery rules is hidden from the users.

The internet layer defines an official packet format and protocol called **IP** (**Internet Protocol**). The job of the internet layer is to deliver IP packets where they are supposed to go. Packet routing is clearly the major issue here, as is avoiding congestion. For these reasons, it is reasonable to say that the TCP/IP internet layer is very similar in functionality to the OSI network layer. Figure 1-18 shows this correspondence.

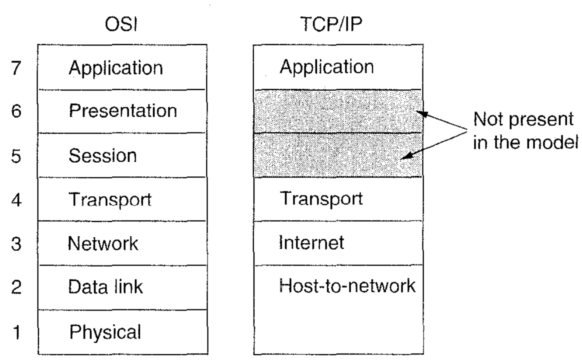

Fig. 1-18. The TCP/IP reference model.

The Transport Layer

The layer above the internet layer in the TCP/IP model is now usually called the **transport layer**. It is designed to allow peer entities on the source and destination hosts to carry on a conversation, the same as in the OSI transport layer. Two end-to-end protocols have been defined here. The first one, **TCP** (**Transmission Control Protocol**) is a reliable connection-oriented protocol that allows a byte stream originating on one machine to be delivered without error on any other machine in the internet. It fragments the incoming byte stream into discrete messages and passes each one onto the internet layer. At the destination, the receiving TCP process reassembles the received messages into the output stream. TCP also handles flow control to make sure a fast sender cannot swamp a slow receiver with more messages than it can handle.

The second protocol in this layer, **UDP** (**User Datagram Protocol**), is an unreliable, connectionless protocol for applications that do not want TCP's sequencing or flow control and wish to provide their own. It is also widely used for one-shot, client-server type request-reply queries and applications in which prompt delivery is more important than accurate delivery, such as transmitting speech or video. The relation of IP, TCP, and UDP is shown in Fig. 1-19. Since the model was developed, IP has been implemented on many other networks.

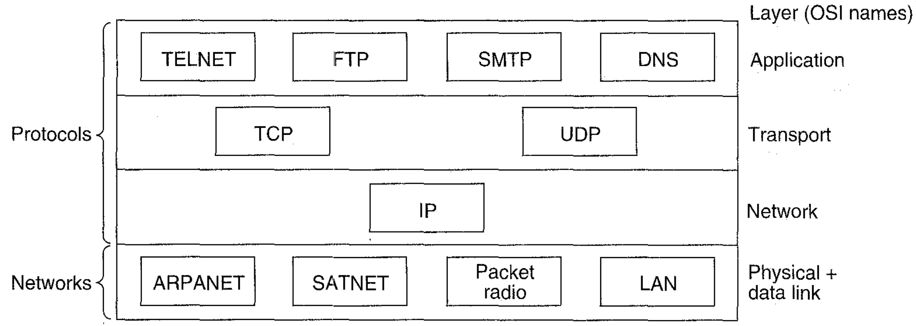

Fig. 1-19. Protocols and networks in the TCP/IP model initially.

The Application Layer

The TCP/IP model does not have session or presentation layers. No need for them was perceived, so they were not included. Experience with the OSI model has proven this view correct: they are of little use to most applications.

On top of the transport layer is the **application layer**. It contains all the higher-level protocols. The early ones included virtual terminal (TELNET), file transfer (FTP), and electronic mail (SMTP), as shown in Fig. 1-19. The virtual terminal protocol allows a user on one machine to log into a distant machine and work there. The file transfer protocol provides a way to move data efficiently from one machine to another. Electronic mail was originally just a kind of file transfer, but later a specialized protocol was developed for it. Many other protocols have been added to these over the years, such as the Domain Name Service (DNS) for mapping host names onto their network addresses, NNTP, the protocol used for moving news articles around, and HTTP, the protocol used for fetching pages on the World Wide Wide, and many others.

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The Host-to-Network Layer

Below the internet layer is a great void. The TCP/IP reference model does not really say much about what happens here, except to point out that the host has to connect to the network using some protocol so it can send IP packets over it. This protocol is not defined and varies from host to host and network to network. Books and papers about the TCP/IP model rarely discuss it.

1.4.3. A Comparison of the OSI and TCP Reference Models

The OSI and TCP/IP reference models have much in common. Both are based on the concept of a stack of independent protocols. Also, the functionality of the layers is roughly similar. For example, in both models the layers up through and including the transport layer are there to provide an end-to-end network-independent transport service to processes wishing to communicate. These layers form the transport provider. Again in both models, the layers above transport are application-oriented users of the transport service.

Despite these fundamental similarities, the two models also have many differences. In this section we will focus on the key differences between the two reference models. It is important to note that we are comparing the *reference models* here, not the corresponding *protocol stacks*. The protocols themselves will be discussed later. For an entire book comparing and contrasting TCP/IP and OSI, see (Piscitello and Chapin, 1993).

Three concepts are central to the OSI model:

1. Services
2. Interfaces
3. Protocols

Probably the biggest contribution of the OSI model is to make the distinction between these three concepts explicit. Each layer performs some services for the layer above it. The *service* definition tells what the layer does, not how entities above it access it or how the layer works.

A layer's *interface* tells the processes above it how to access it. It specifies what the parameters are and what results to expect. It, too, says nothing about how the layer works inside.

Finally, the peer *protocols* used in a layer are the layer's own business. It can use any protocols it wants to, as long as it gets the job done (i.e., provides the offered services). It can also change them at will without affecting software in higher layers.

These ideas fit very nicely with modern ideas about object-oriented programming. An object, like a layer, has a set of methods (operations) that processes outside the object can invoke. The semantics of these methods define the set of services that the object offers. The methods' parameters and results form the object's interface. The code internal to the object is its protocol and is not visible or of any concern outside the object.

The TCP/IP model did not originally clearly distinguish between service, interface, and protocol, although people have tried to retrofit it after the fact to make it more OSI-like. For example, the only real services offered by the internet layer are SEND IP PACKET and RECEIVE IP PACKET.

As a consequence, the protocols in the OSI model are better hidden than in the TCP/IP model and can be replaced relatively easily as the technology changes. Being able to make such changes is one of the main purposes of having layered protocols in the first place.

The OSI reference model was devised *before* the protocols were invented. This ordering means that the model was not biased toward one particular set of protocols, which made it quite general. The down side of this ordering is that the designers did not have much experience with the subject and did not have a good idea of which functionality to put in which layer.

For example, the data link layer originally dealt only with point-to-point networks. When broadcast networks came around, a new sublayer had to be hacked into the model. When people started to build real networks using the OSI model and existing protocols, it was discovered that they did not match the required service specifications (wonder of wonders), so convergence sublayers had to be grafted onto the model to provide a place for papering over the differences. Finally, the committee originally expected that each country would have one network, run by the government and using the OSI protocols, so no thought was given to internetworking. To make a long story short, things did not turn out that way.

With the TCP/IP the reverse was true: the protocols came first, and the model was really just a description of the existing protocols. There was no problem with the protocols fitting the model. They fit perfectly. The only trouble was that the *model* did not fit any other protocol stacks. Consequently, it was not especially useful for describing other non-TCP/IP networks.

Turning from philosophical matters to more specific ones, an obvious difference between the two models is the number of layers: the OSI model has seven layers and the TCP/IP has four layers. Both have (inter)network, transport, and application layers, but the other layers are different.

Another difference is in the area of connectionless versus connection-oriented communication. The OSI model supports both connectionless and connection-oriented communication in the network layer, but only connection-oriented communication in the transport layer, where it counts (because the transport service is visible to the users). The TCP/IP model has only one mode in the network layer (connectionless) but supports both modes in the transport layer, giving the users a choice. This choice is especially important for simple request-response protocols.

1.4.4. A Critique of the OSI Model and Protocols

Neither the OSI model and its protocols nor the TCP/IP model and its protocols are perfect. Quite a bit of criticism can be, and has been, directed at both of them. In this section and the next one, we will look at some of these criticisms. We will begin with OSI and examine TCP/IP afterward.

At the time the second edition of this book was published (1989), it appeared to most experts in the field that the OSI model and its protocols were going to take over the world and push everything else out of their way. This did not happen. Why? A look back at some of the lessons may be useful. These lessons can be summarized as:

1. Bad timing.
2. Bad technology.
3. Bad implementations.
4. Bad politics.

Bad Timing

First let us look at reason one: bad timing. The time at which a standard is established is absolutely critical to its success. David Clark of M.I.T. has a theory of standards that he calls the *apocalypse of the two elephants*, and which is illustrated in Fig. 1-20.

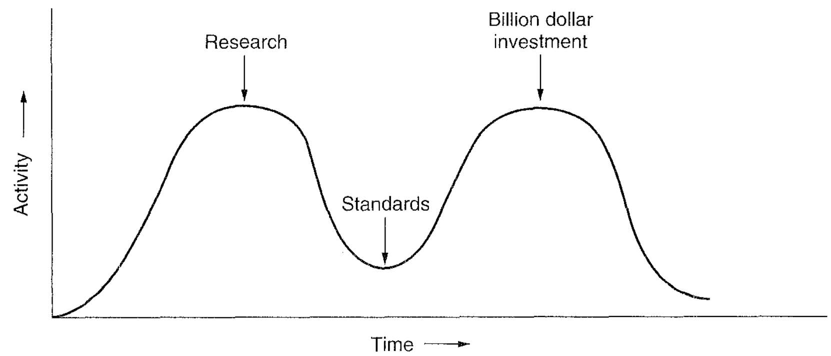

Fig. 1-20. The apocalypse of the two elephants.

This figure shows the amount of activity surrounding a new subject. When the subject is first discovered, there is a burst of research activity in the form of discussions, papers, and meetings. After a while this subsides, corporations discover the subject, and the billion-dollar wave of investment hits.

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It is essential that the standards be written in the trough between the two "elephants." If they are written too early, before the research is finished, the subject may still be poorly understood, which leads to bad standards. If they are written too late, so many companies may have already made major investments in different ways of doing things that the standards are effectively ignored. If the interval between the two elephants is very short (because everyone is in a hurry to get started), the people developing the standards may get crushed.

It now appears that the standard OSI protocols got crushed. The competing TCP/IP protocols were already in widespread use by research universities by the time the OSI protocols appeared. While the billion-dollar wave of investment had not yet hit, the academic market was large enough that many vendors had begun cautiously offering TCP/IP products. When OSI came around, they did not want to support a second protocol stack until they were forced to, so there were no initial offerings. With every company waiting for every other company to go first, no company went first and OSI never happened.

Bad Technology

The second reason that OSI never caught on is that both the model and the protocols are flawed. Most discussions of the seven-layer model give the impression that the number and contents of the layers eventually chosen were the only way, or at least the obvious way. This is far from true. The session layer has little use in most applications, and the presentation layer is nearly empty. In fact, the British proposal to ISO only had five layers, not seven. In contrast to the session and presentation layers, the data link and network layers are so full that subsequent work has split them into multiple sublayers, each with different functions.

Although hardly anyone ever admits it in public, the real reason that the OSI model has seven layers is that at the time it was designed, IBM had a proprietary seven-layer protocol called **SNA**™ (**Systems Network Architecture**). At that time, IBM so dominated the computer industry that everyone else, including telephone companies, competing computer companies, and even major governments, were scared to death that IBM would use its market clout to effectively force everybody to use SNA, which it could change whenever it wished. The idea behind OSI was to produce an IBM-like reference model and protocol stack that would become the world standard, and controlled not by one company, but by a neutral organization, ISO.

The OSI model, along with the associated service definitions and protocols, is extraordinarily complex. When piled up, the printed standards occupy a significant fraction of a meter of paper. They are also difficult to implement and inefficient in operation. In this context, a riddle posed by Paul Mockapetris and cited in (Rose, 1993) comes to mind:

Q: What do you get when you cross a mobster with an international standard?
A: Someone who makes you an offer you can't understand.

In addition to being incomprehensible, another problem with OSI is that some functions, such as addressing, flow control, and error control reappear again and again in each layer. Saltzer et al. (1984), for example, have pointed out that to be effective, error control must be done in the highest layer, so that repeating it over and over in each of the lower layers is often unnecessary and inefficient.

Another issue is that the decision to place certain features in particular layers is not always obvious. The virtual terminal handling (now in the application layer) was in the presentation layer during much of the development of the standard. It was moved to the application layer because the committee had trouble deciding what the presentation layer was good for. Data security and encryption were so controversial that no one could agree which layer to put them in, so they were left out altogether. Network management was also omitted from the model for similar reasons.

Another criticism of the original standard is that it completely ignored connectionless services and connectionless protocols, even though most local area networks work that way. Subsequent addenda (known in the software world as bug fixes) corrected this problem.

Perhaps the most serious criticism is that the model is dominated by a communications mentality. The relationship of computing to communications is barely mentioned anywhere, and some of the choices made are wholly inappropriate to the way computers and software work. As an example, consider the OSI primitives, listed in Fig. 1-14. In particular, think carefully about the primitives and how one might use them in a programming language.

The CONNECT.request primitive is simple. One can imagine a library procedure, *connect*, that programs can call to establish a connection. Now think about CONNECT.indication. When a message arrives, the destination process has to be signaled. In effect, it has to get an interrupt—hardly an appropriate concept for programs written in any modern high-level language. Of course, in the lowest layer, an indication (interrupt) does occur.

If the program were expecting an incoming call, it could call a library procedure *receive* to block itself. But if this were the case, why was *receive* not the primitive instead of *indication*? *Receive* is clearly oriented toward the way computers work, whereas *indication* is equally clearly oriented toward the way telephones work. Computers are different from telephones. Telephones ring. Computers do not ring. In short, the semantic model of an interrupt-driven system is conceptually a poor idea and totally at odds with all modern ideas of structured programming. This and similar problems are discussed by Langsford (1984).

Bad Implementations

Given the enormous complexity of the model and the protocols, it will come as no surprise that the initial implementations were huge, unwieldy, and slow. Everyone who tried them got burned. It did not take long for people to associate

"OSI" with "poor quality." While the products got better in the course of time, the image stuck.

In contrast, one of the first implementations of TCP/IP was part of Berkeley UNIX® and was quite good (not to mention, free). People began using it quickly, which led to a large user community, which led to improvements, which led to an even larger community. Here the spiral was upward instead of downward.

Bad Politics

On account of the initial implementation, many people, especially in academia, thought of TCP/IP as part of UNIX, and UNIX in the 1980s in academia was not unlike parenthood (then incorrectly called motherhood) and apple pie.

OSI, on the other hand, was thought to be the creature of the European telecommunication ministries, the European Community, and later the U.S. Government. This belief was only partly true, but the very idea of a bunch of government bureaucrats trying to shove a technically inferior standard down the throats of the poor researchers and programmers down in the trenches actually developing computer networks did not help much. Some people viewed this development in the same light as IBM announcing in the 1960s that PL/I was the language of the future, or DoD correcting this later by announcing that it was actually Ada®.

Despite the fact that the OSI model and protocols have been less than a resounding success, there are still a few organizations interested in it, mostly European PTTs that still have a monopoly on telecommunication. Consequently a feeble effort has been made to update OSI, resulting in a revised model published in 1994. For what was changed (little) and what should have been changed (a lot), see (Day, 1995).

1.4.5. A Critique of the TCP/IP Reference Model

The TCP/IP model and protocols have their problems too. First, the model does not clearly distinguish the concepts of service, interface, and protocol. Good software engineering practice requires differentiating between the specification and the implementation, something that OSI does very carefully, and TCP/IP does not. Consequently, the TCP/IP model is not much of a guide for designing new networks using new technologies.

Second, the TCP/IP model is not at all general and is poorly suited to describing any protocol stack other than TCP/IP. Trying to describe SNA using the TCP/IP model would be nearly impossible, for example.

Third, the host-to-network layer is not really a layer at all in the normal sense that the term is used in the context of layered protocols. It is an interface (between the network and data link layers). The distinction between an interface and a layer is a crucial one and one should not be sloppy about it.

Fourth, the TCP/IP model does not distinguish (or even mention) the physical and data link layers. These are completely different. The physical layer has to do with the transmission characteristics of copper wire, fiber optics, and wireless communication. The data link layer's job is to delimit the start and end of frames and get them from one side to the other with the desired degree of reliability. A proper model should include both as separate layers. The TCP/IP model does not do this.

Finally, although the IP and TCP protocols were carefully thought out, and well implemented, many of the other protocols were ad hoc, generally produced by a couple of graduate students hacking away until they got tired. The protocol implementations were then distributed free, which resulted in their becoming widely used, deeply entrenched, and thus hard to replace. Some of them are a bit of an embarrassment now. The virtual terminal protocol, TELNET, for example, was designed for a ten-character per second mechanical Teletype terminal. It knows nothing of graphical user interfaces and mice. Nevertheless, 25 years later, it is still in widespread use.

In summary, despite its problems, the OSI *model* (minus the session and presentation layers) has proven to be exceptionally useful for discussing computer networks. In contrast, the OSI *protocols* have not become popular. The reverse is true of TCP/IP: the *model* is practically nonexistent, but the *protocols* are widely used. Since computer scientists like to have their cake and eat it, too, in this book we will use a modified OSI model but concentrate primarily on the TCP/IP and related protocols, as well as newer ones such as SMDS, frame relay, SONET, and ATM. In effect, we will use the hybrid model of Fig. 1-21 as the framework for this book.

5	Application layer
4	Transport layer
3	Network layer
2	Data Link layer
1	Physical layer

Fig. 1-21. The hybrid reference model to be used in this book.

1.5. EXAMPLE NETWORKS

Numerous networks are currently operating around the world. Some of these are public networks run by common carriers or PTTs, others are research networks, yet others are cooperative networks run by their users, and still others are commercial or corporate networks. In the following sections we will take a look

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at a few current and historical networks to get an idea of what they are (or were) like and how they differ from one another.

Networks differ in their history, administration, facilities offered, technical design, and user communities. The history and administration can vary from a network carefully planned by a single organization with a well-defined goal, to an ad hoc collection of machines that have been connected to one another over the years without any master plan or central administration at all. The facilities available range from arbitrary process-to-process communication to electronic mail, file transfer, remote login, and remote execution. The technical designs can differ in the transmission media used, the naming and routing algorithms employed, the number and contents of the layers present, and the protocols used. Finally, the user community can vary from a single corporation to all the academic computer scientists in the industrialized world.

In the following sections we will look at a few examples. These are the popular commercial LAN networking package, Novell NetWare®, the worldwide Internet (including its predecessors, the ARPANET and NSFNET), and the first gigabit networks.

1.5.1. Novell NetWare

The most popular network system in the PC world is **Novell NetWare**. It was designed to be used by companies downsizing from a mainframe to a network of PCs. In such systems, each user has a desktop PC functioning as a client. In addition, some number of powerful PCs operate as servers, providing file services, database services, and other services to a collection of clients. In other words, Novell NetWare is based on the client-server model.

NetWare uses a proprietary protocol stack illustrated in Fig. 1-22. It is based on the old Xerox Network System, XNS™ but with various modifications. Novell NetWare predates OSI and is not based on it. If anything, it looks more like TCP/IP than like OSI.

Layer			
Application	SAP	File server	. . .
Transport	NCP		SPX
Network	IPX		
Data link	Ethernet	Token ring	ARCnet
Physical	Ethernet	Token ring	ARCnet

Fig. 1-22. The Novell NetWare reference model.

The physical and data link layers can be chosen from among various industry standards, including Ethernet, IBM token ring, and ARCnet. The network layer

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runs an unreliable connectionless internetwork protocol called **IPX** (**Internet Packet eXchange**). It passes packets transparently from source to destination, even if the source and destination are on different networks. IPX is functionally similar to IP, except that it uses 12-byte addresses instead of 4-byte addresses. The wisdom of this choice will become apparent in Chap. 5.

Above IPX comes a connection-oriented transport protocol called **NCP** (**Network Core Protocol**). NCP also provides various other services besides user data transport and is really the heart of NetWare. A second protocol, **SPX** (**Sequenced Packet eXchange**), is also available, but provides only transport. TCP is another option. Applications can choose any of them. The file system uses NCP and Lotus Notes® uses SPX, for example. The session and presentation layers do not exist. Various application protocols are present in the application layer.

As in TCP/IP, the key to the entire architecture is the internet datagram packet on top of which everything else is built. The format of an IPX packet is shown in Fig. 1-23. The *Checksum* field is rarely used, since the underlying data link layer also provides a checksum. The *Packet length* field tells how long the entire packet is, header plus data. The *Transport control* field counts how many networks the packet has traversed. When this exceeds a maximum, the packet is discarded. The *Packet type* field is used to mark various control packets. The two addresses each contain a 32-bit network number, a 48-bit machine number (the 802 LAN address), and 16-bit local address (socket) on that machine. Finally, we have the data, which occupy the rest of the packet, with the maximum size being determined by the underlying network.

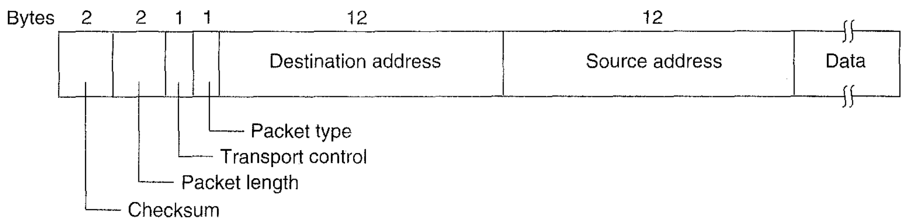

Fig. 1-23. A Novell NetWare IPX packet.

About once a minute, each server broadcasts a packet giving its address and telling what services it offers. These broadcasts use the **SAP** (**Service Advertising Protocol**) protocol. The packets are seen and collected by special agent processes running on the router machines. The agents use the information contained in them to construct databases of which servers are running where.

When a client machine is booted, it broadcasts a request asking where the nearest server is. The agent on the local router machine sees this request, looks in its database of servers, and matches up the request with the best server. The choice of server to use is then sent back to the client. The client can now establish an NCP connection with the server. Using this connection, the client and server negotiate the maximum packet size. From this point on, the client can access the file system and other services using this connection. It can also query the server's database to look for other (more distant) servers.

1.5.2. The ARPANET

Let us now switch gears from LANs to WANs. In the mid-1960s, at the height of the Cold War, the DoD wanted a command and control network that could survive a nuclear war. Traditional circuit-switched telephone networks were considered too vulnerable, since the loss of one line or switch would certainly terminate all conversations using them and might even partition the network. To solve this problem, DoD turned to its research arm, ARPA (later DARPA, now ARPA again), the (periodically Defense) Advanced Research Projects Agency.

ARPA was created in response to the Soviet Union's launching Sputnik in 1957 and had the mission of advancing technology that might be useful to the military. ARPA had no scientists or laboratories, in fact, it had nothing more than an office and a small (by Pentagon standards) budget. It did its work by issuing grants and contracts to universities and companies whose ideas looked promising to it.

Several early grants went to universities for investigating the then-radical idea of packet switching, something that had been suggested by Paul Baran in a series of RAND Corporation reports published in the early 1960s. After some discussions with various experts, ARPA decided that the network the DoD needed should be a packet-switched network, consisting of a subnet and host computers.

The subnet would consist of minicomputers called **IMP**s (**Interface Message Processors**) connected by transmission lines. For high reliability, each IMP would be connected to at least two other IMPs. The subnet was to be a datagram subnet, so if some lines and IMPs were destroyed, messages could be automatically rerouted along alternative paths.

Each node of the network was to consist of an IMP and a host, in the same room, connected by a short wire. A host could send messages of up to 8063 bits to its IMP, which would then break these up into packets of at most 1008 bits and forward them independently toward the destination. Each packet was received in its entirety before being forwarded, so the subnet was the first electronic store-and-forward packet-switching network.

ARPA then put out a tender for building the subnet. Twelve companies bid for it. After evaluating all the proposals, ARPA selected BBN, a consulting firm in Cambridge, Massachusetts, and in December 1968, awarded it a contract to build the subnet and write the subnet software. BBN chose to use specially modified Honeywell DDP-316 minicomputers with 12K 16-bit words of core memory as the IMPs. The IMPs did not have disks, since moving parts were considered unreliable. The IMPs were interconnected by 56-kbps lines leased from telephone companies.

The software was split into two parts: subnet and host. The subnet software consisted of the IMP end of the host-IMP connection, the IMP-IMP protocol, and a source IMP to destination IMP protocol designed to improve reliability. The original ARPANET design is shown in Fig. 1-24.

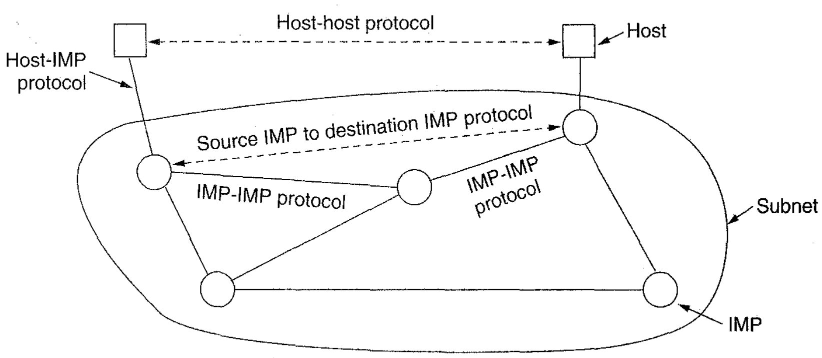

Fig. 1-24. The original ARPANET design.

Outside the subnet, software was also needed, namely, the host end of the host-IMP connection, the host-host protocol, and the application software. It soon became clear that BBN felt that when it had accepted a message on a host-IMP wire and placed it on the host-IMP wire at the destination, its job was done.

To deal with problem of host software, Larry Roberts of ARPA convened a meeting of network researchers, mostly graduate students, at Snowbird, Utah, in the summer of 1969. The graduate students expected some network expert to explain the design of the network and its software to them and then to assign each of them the job of writing part of it. They were astounded when there was no network expert and no grand design. They had to figure out what to do on their own.

Nevertheless, somehow an experimental network went on the air in December 1969 with four nodes, at UCLA, UCSB, SRI, and the University of Utah. These four were chosen because all had a large number of ARPA contracts, and all had different and completely incompatible host computers (just to make it more fun). The network grew quickly as more IMPs were delivered and installed; it soon spanned the United States. Figure 1-25 shows how rapidly the ARPANET grew in the first 3 years.

Later the IMP software was changed to allow terminals to connect directly to a special IMP, called a **TIP** (**Terminal Interface Processor**), without having to go through a host. Subsequent changes included having multiple hosts per IMP (to save money), hosts talking to multiple IMPs (to protect against IMP failures),

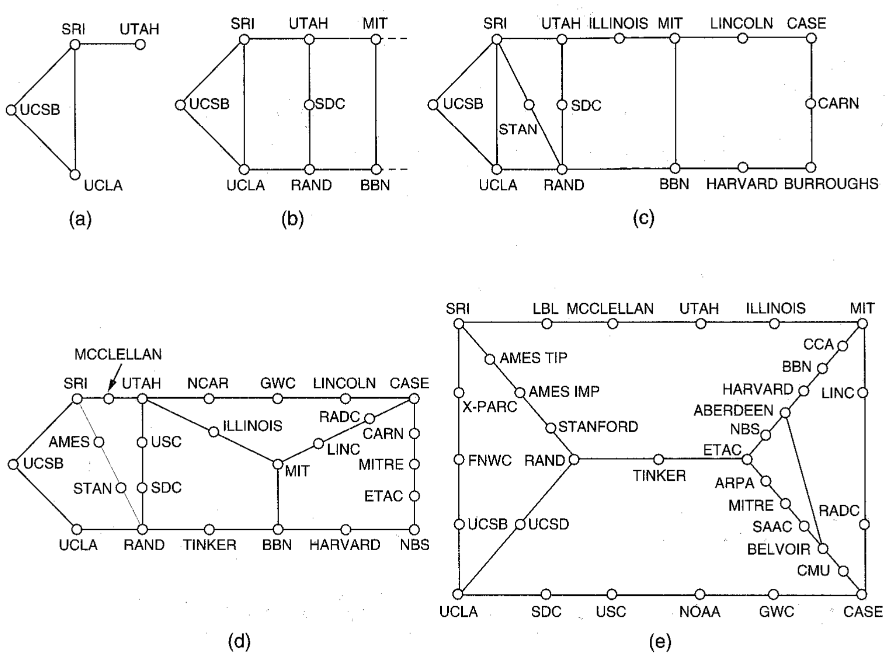

Fig. 1-25. Growth of the ARPANET. (a) Dec. 1969. (b) July 1970. (c) March 1971. (d) April 1972. (e) Sept. 1972.

and hosts and IMPs separated by a large distance (to accommodate hosts far from the subnet).

In addition to helping the fledgling ARPANET grow, ARPA also funded research on satellite networks and mobile packet radio networks. In one famous demonstration, a truck driving around in California used the packet radio network to send messages to SRI, which were then forwarded over the ARPANET to the East Coast, where they were shipped to University College in London over the satellite network. This allowed a researcher in the truck to use a computer in London while driving around in California.

This experiment also demonstrated that the existing ARPANET protocols were not suitable for running over multiple networks. This observation led to more research on protocols, culminating with the invention of the TCP/IP model and protocols (Cerf and Kahn, 1974). TCP/IP was specifically designed to handle communication over internetworks, something becoming increasingly important as more and more networks were being hooked up to the ARPANET.

To encourage adoption of these new protocols, ARPA awarded several contracts to BBN and the University of California at Berkeley to integrate them into Berkeley UNIX. Researchers at Berkeley developed a convenient program

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interface to the network (sockets) and wrote many application, utility, and management programs to make networking easier.

The timing was perfect. Many universities had just acquired a second or third VAX computer and a LAN to connect them, but they had no networking software. When 4.2BSD came along, with TCP/IP, sockets, and many network utilities, the complete package was adopted immediately. Furthermore, with TCP/IP, it was easy for the LANs to connect to the ARPANET, and many did.

By 1983, the ARPANET was stable and successful, with over 200 IMPs and hundreds of hosts. At this point, ARPA turned the management of the network over to the Defense Communications Agency (DCA), to run it as an operational network. The first thing DCA did was to separate the military portion (about 160 IMPs, of which 110 in the United States and 50 abroad) into a separate subnet, **MILNET**, with stringent gateways between MILNET and the remaining research subnet.

During the 1980s, additional networks, especially LANs, were connected to the ARPANET. As the scale increased, finding hosts became increasingly expensive, so **DNS** (**Domain Naming System**) was created to organize machines into domains and map host names onto IP addresses. Since then, DNS has become a generalized, distributed database system for storing a variety of information related to naming. We will study it in detail in Chap. 7.

By 1990, the ARPANET had been overtaken by newer networks that it itself had spawned, so it was shut down and dismantled, but it lives on in the hearts and minds of network researchers everywhere. MILNET continues to operate, however.

1.5.3. NSFNET

By the late 1970s, NSF (the U.S. National Science Foundation) saw the enormous impact the ARPANET was having on university research, allowing scientists across the country to share data and collaborate on research projects. However, to get on the ARPANET, a university had to have a research contract with the DoD, which many did not have. This lack of universal access prompted NSF to set up a virtual network, **CSNET**, centered around a single machine at BBN that supported dial-up lines and had connections to the ARPANET and other networks. Using CSNET, academic researchers could call up and leave email for other people to pick up later. It was simple, but it worked.

By 1984 NSF began designing a high-speed successor to the ARPANET that would be open to all university research groups. To have something concrete to start with, NSF decided to build a backbone network to connect its six supercomputer centers, in San Diego, Boulder, Champaign, Pittsburgh, Ithaca, and Princeton. Each supercomputer was given a little brother, consisting of an LSI-11 microcomputer called a **fuzzball**. The fuzzballs were connected with 56 kbps leased lines and formed the subnet, the same hardware technology as the

ARPANET used. The software technology was different however: the fuzzballs spoke TCP/IP right from the start, making it the first TCP/IP WAN.

NSF also funded some (eventually about 20) regional networks that connected to the backbone to allow users at thousands of universities, research labs, libraries, and museums to access any of the supercomputers and to communicate with one another. The complete network, including the backbone and the regional networks, was called **NSFNET**. It connected to the ARPANET through a link between an IMP and a fuzzball in the Carnegie-Mellon machine room. The first NSFNET backbone is illustrated in Fig. 1-26.

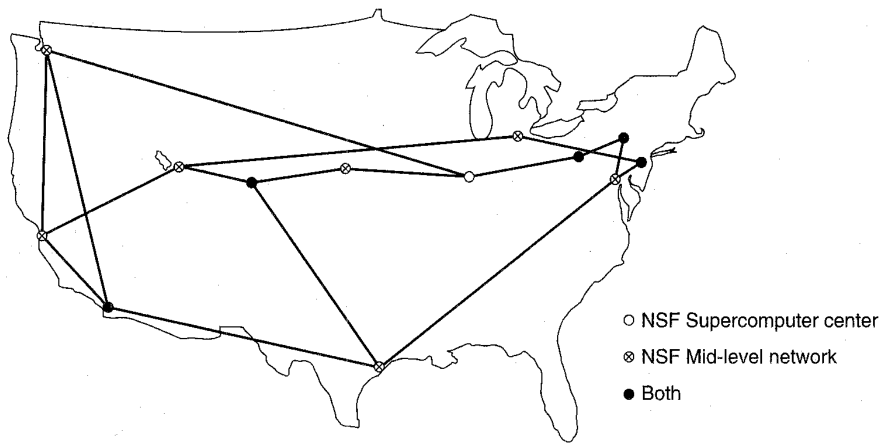

Fig. 1-26. The NSFNET backbone in 1988.

NSFNET was an instantaneous success and was overloaded from the word go. NSF immediately began planning its successor and awarded a contract to the Michigan-based MERIT consortium to run it. Fiber optic channels at 448 kbps were leased from MCI to provide the version 2 backbone. IBM RS6000s were used as routers. This, too, was soon overwhelmed, and by 1990, the second backbone was upgraded to 1.5 Mbps.

As growth continued, NSF realized that the government could not continue financing networking forever. Furthermore, commercial organizations wanted to join but were forbidden by NSF's charter from using networks NSF paid for. Consequently, NSF encouraged MERIT, MCI, and IBM to form a nonprofit corporation, **ANS** (**Advanced Networks and Services**) as a step along the road to commercialization. In 1990, ANS took over NSFNET and upgraded the 1.5-Mbps links to 45 Mbps to form **ANSNET**.

In December 1991, the U.S. Congress passed a bill authorizing **NREN**, the **National Research and Educational Network**, the research successor to NSFNET, only running at gigabits speeds. The goal was a national network running at 3 Gbps before the millenium. This network is to act as a prototype for the much-discussed information superhighway.

By 1995, the NSFNET backbone was no longer needed to interconnect the NSF regional networks because numerous companies were running commercial IP networks. When ANSNET was sold to America Online in 1995, the NSF regional networks had to go out and buy commercial IP service to interconnect.

To ease the transition and make sure every regional network could communicate with every other regional network, NSF awarded contracts to four different network operators to establish a **NAP** (**Network Access Point**). These operators were PacBell (San Francisco), Ameritech (Chicago), MFS (Washington, D.C.), and Sprint (New York City, where for NAP purposes, Pennsauken, N.J. counts as New York City). Every network operator that wanted to provide backbone service to the NSF regional networks had to connect to all the NAPs. This arrangement meant that a packet originating on any regional network had a choice of backbone carriers to get from its NAP to the destination's NAP. Consequently, the backbone carriers were forced to compete for the regional networks' business on the basis of service and price, which was the idea, of course. In addition to the NSF NAPs, various government NAPs (e.g., FIX-E, FIX-W, MAE-East and MAE-West) and commercial NAPs (e.g., CIX) have also been created, so the concept of a single default backbone was replaced by a commercially-driven competitive infrastructure.

Other countries and regions are also building networks comparable to NSFNET. In Europe, for example, EBONE is an IP backbone for research organizations and EuropaNET is a more commercially oriented network. Both connect numerous cities in Europe with 2-Mbps lines. Upgrades to 34 Mbps are in progress. Each country in Europe has one or more national networks, which are roughly comparable to the NSF regional networks.

1.5.4. The Internet

The number of networks, machines, and users connected to the ARPANET grew rapidly after TCP/IP became the only official protocol on Jan. 1, 1983. When NSFNET and the ARPANET were interconnected, the growth became exponential. Many regional networks joined up, and connections were made to networks in Canada, Europe, and the Pacific.

Sometime in the mid-1980s, people began viewing the collection of networks as an internet, and later as the Internet, although there was no official dedication with some politician breaking a bottle of champagne over a fuzzball.

Growth continued exponentially, and by 1990 the Internet had grown to 3000 networks and 200,000 computers. In 1992, the one millionth host was attached. By 1995, there were multiple backbones, hundreds of mid-level (i.e., regional) networks, tens of thousands of LANs, millions of hosts, and tens of millions of users. The size doubles approximately every year (Paxson, 1994).

Much of the growth comes from connecting existing networks to the Internet. In the past these have included SPAN, NASA's space physics network, HEPNET, a high energy physics network, BITNET, IBM's mainframe network, EARN, a European academic network now widely used in Eastern Europe, and many others. Numerous transatlantic links are in use, running from 64 kbps to 2 Mbps.

The glue that holds the Internet together is the TCP/IP reference model and TCP/IP protocol stack. TCP/IP makes universal service possible and can be compared to the telephone system or the adoption of standard gauge by the railroads in the 19th Century.

What does it actually mean to be on the Internet? Our definition is that a machine is on the Internet if it runs the TCP/IP protocol stack, has an IP address, and has the ability to send IP packets to all the other machines on the Internet. The mere ability to send and receive electronic mail is not enough, since email is gatewayed to many networks outside the Internet. However, the issue is clouded somewhat by the fact that many personal computers have the ability to call up an Internet service provider using a modem, be assigned a temporary IP address, and send IP packets to other Internet hosts. It make sense to regard such machines as being on the Internet for as long as they are connected to the service provider's router.

With exponential growth, the old informal way of running the Internet no longer works. In January 1992, the **Internet Society** was set up, to promote the use of the Internet and perhaps eventually take over managing it.

Traditionally, the Internet had four main applications, as follows:

1. **Email**. The ability to compose, send, and receive electronic mail has been around since the early days of the ARPANET and is enormously popular. Many people get dozens of messages a day and consider it their primary way of interacting with the outside world, far outdistancing the telephone and snail mail. Email programs are available on virtually every kind of computer these days.

2. **News**. Newsgroups are specialized forums in which users with a common interest can exchange messages. Thousands of newsgroups exist, on technical and nontechnical topics, including computers, science, recreation, and politics. Each newsgroup has its own etiquette, style, and customs, and woe be to anyone violating them.

3. **Remote login**. Using the Telnet, Rlogin, or other programs, users anywhere on the Internet can log into any other machine on which they have an account.

4. **File transfer**. Using the FTP program, it is possible to copy files from one machine on the Internet to another. Vast numbers of articles, databases, and other information are available this way.

Up until the early 1990s, the Internet was largely populated by academic, government, and industrial researchers. One new application, the **WWW** (**World Wide Web**) changed all that and brought millions of new, nonacademic users to the net. This application, invented by CERN physicist Tim Berners-Lee, did not change any of the underlying facilities but made them easier to use. Together with the Mosaic viewer, written at the National Center for Supercomputer Applications, the WWW made it possible for a site to set up a number of pages of information containing text, pictures, sound, and even video, with embedded links to other pages. By clicking on a link, the user is suddenly transported to the page pointed to by that link. For example, many companies have a home page with entries pointing to other pages for product information, price lists, sales, technical support, communication with employees, stockholder information, and much more.

Numerous other kinds of pages have come into existence in a very short time, including maps, stock market tables, library card catalogs, recorded radio programs, and even a page pointing to the complete text of many books whose copyrights have expired (Mark Twain, Charles Dickens, etc.). Many people also have personal pages (home pages).

In the first year after Mosaic was released, the number of WWW servers grew from 100 to 7000. Enormous growth will undoubtedly continue for years to come, and will probably be the force driving the technology and use of the Internet into the next millenium.

Many books have been written about the Internet and its protocols. For more information, see (Black, 1995; Carl-Mitchell and Quarterman, 1993; Comer, 1995; and Santifaller, 1994).

1.5.5. Gigabit Testbeds

The Internet backbones operate at megabit speeds, so for people who want to push the technological envelope, the next step is gigabit networking. With each increase in network bandwidth, new applications become possible, and gigabit networks are no exception. In this section we will first say a few words about gigabit applications, mention two of them, and then list some example gigabit testbeds that have been built.

Gigabit networks provide better bandwidth than megabit networks, but not always much better delay. For example, sending a 1-kbit packet from New York to San Francisco at 1 Mbps takes 1 msec to pump the bits out and 20 msec for the transcontinental delay, for a total of 21 msec. A 1-Gbps network can reduce this to 20.001 msec. While the bits go out faster, the transcontinental delay remains the same, since the speed of light in optical fiber (or copper wire) is about 200,000 km/sec, independent of the data rate. Thus for wide area applications in which low delay is critical, going to higher speeds may not help much. Fortunately, for some applications, bandwidth is what counts, and these are the applications for which gigabit networks will make a big difference.

One application is telemedicine. Many people think that a way to reduce medical costs is to reintroduce family doctors and family clinics on a large scale, so everyone has convenient access to first line medical care. When a serious medical problem occurs, the family doctor can order lab tests and medical imaging, such as X-rays, CAT scans, and MRI scans. The test results and images can then be sent electronically to a specialist who then makes the diagnosis.

Doctors are generally unwilling to make diagnoses from computer images unless the quality of the transmitted image is as good as the original image. This requirement means images will probably need 4K × 4K pixels, with 8 bits per pixel (black and white images) or 24 bits per pixel (color images). Since many tests require up to 100 images (e.g., different cross sections of the organ in question), a single series for one patient can generate 40 gigabits. Moving images (e.g., a beating heart) generate even more data. Compression can help some but doctors are leary of it because the most efficient algorithms reduce image quality. Furthermore, all the images must be stored for years but may need to be retrieved at a moment's notice in the event of a medical emergency. Hospitals do not want to become computer centers, so off-site storage combined with high-bandwidth electronic retrieval is essential.

Another gigabit application is the virtual meeting. Each meeting room contains a spherical camera and one or more people. The bit streams from each of the cameras are combined electronically to give the illusion that everyone is in the same room. Each person sees this image using virtual reality goggles. In this way meetings can happen without travel, but again, the data rates required are stupendous.

Starting in 1989, ARPA and NSF jointly agreed to finance a number of university-industry gigabit testbeds, later as part of the NREN project. In some of these, the data rate in each direction was 622 Mbps, so only by counting the data going in both directions do you get a gigabit. This kind of gigabit is sometimes called a "government gigabit." (Some cynics call it a gigabit after taxes.) Below we will briefly mention the first five projects. They have done their job and been shut down, but deserve some credit as pioneers, in the same way the ARPANET does.

1. **Aurora** was a testbed linking four sites in the Northeast: M.I.T., the University of Pennsylvania, IBM's T.J. Watson Lab, and Bellcore (Morristown, N.J.) at 622 Mbps using fiber optics provided by MCI, Bell Atlantic, and NYNEX. Aurora was largely designed to help debug Bellcore's Sunshine switch and IBM's (proprietary) plaNET switch using parallel networks. Research issues included switching technology, gigabit protocols, routing, network control, distributed virtual memory, and collaboration using videoconferencing. For more information, see (Clark et al., 1993).

2. **Blanca** was originally a research project called XUNET involving AT&T Bell Labs, Berkeley, and the University of Wisconsin. In 1990 it added some new sites (LBL, Cray Research, and the University of Illinois) and acquired NSF/ARPA funding. Some of it ran at 622 Mbps, but other parts ran at lower speeds. Blanca was the only nationwide testbed; the rest were regional. Consequently, much of the research was concerned with the effects of speed-of-light delay. The interest here was in protocols, especially network control protocols, host interfaces, and gigabit applications such as medical imaging, meteorological modeling, and radio astronomy. For more information, see (Catlett, 1992; and Fraser, 1993).

3. **CASA** was aimed at doing research on supercomputer applications, especially those in which part of the problem ran best on one kind of supercomputer (e.g., a Cray vector supercomputer) and part ran best on a different kind of supercomputer (e.g., a parallel supercomputer). The applications investigated included geology (analyzing Landsat images), climate modeling, and understanding chemical reactions. It operated in California and New Mexico and connected Los Alamos, Cal Tech, JPL, and the San Diego Supercomputer Center.

4. **Nectar** differed from the three testbeds given above in that it was an experimental gigabit MAN running from CMU to the Pittsburgh Supercomputer Center. The designers were interested in applications involving chemical process flowsheeting and operations research, as well as the tools for debugging them.

5. **VISTAnet** was a small gigabit testbed operated in Research Triangle Park, North Carolina, and connecting the University of North Carolina, North Carolina State University, and MCNC. The interest here was in a prototype for a public switched gigabit network with switches having hundreds of gigabit lines, meaning that the switches had to be capable of processing terabits/sec. The scientific research focused on using 3D images to plan radiation therapy for cancer patients, with the oncologist being able to vary the beam parameters and instantaneously see the radiation dosages being delivered to the tumor and surrounding tissue (Ransom, 1992).

1.6. EXAMPLE DATA COMMUNICATION SERVICES

Telephone companies and others have begun to offer networking services to any organization that wishes to subscribe. The subnet is owned by the network operator, providing communication service for the customers' hosts and terminals.

Such a system is called a **public network**. It is analogous to, and often a part of, the public telephone system. We already briefly looked at one new service, DQDB, in Fig. 1-4. In the following sections we will study four other example services, SMDS, X.25, frame relay, and broadband ISDN.

1.6.1. SMDS—Switched Multimegabit Data Service

The first service we will look at, **SMDS (Switched Multimegabit Data Service)**, was designed to connect together multiple LANs, typically at the branch offices and factories of a single company. It was designed by Bellcore in the 1980s and deployed in the early 1990s by regional and a few long distance carriers. The goal was to produce a high-speed data service and get it out into the world with a minimum of fuss. SMDS is the first broadband (i.e., high-speed) switched service offered to the public.

To see a situation in which SMDS would be useful, consider a company with four offices in four different cities, each with its own LAN. The company would like to connect all the LANs, so that packets can go from one LAN to another. One solution would be to lease six high-speed lines and fully connect the LANs as shown in Fig. 1-27(a). Such a solution is certainly possible, but expensive.

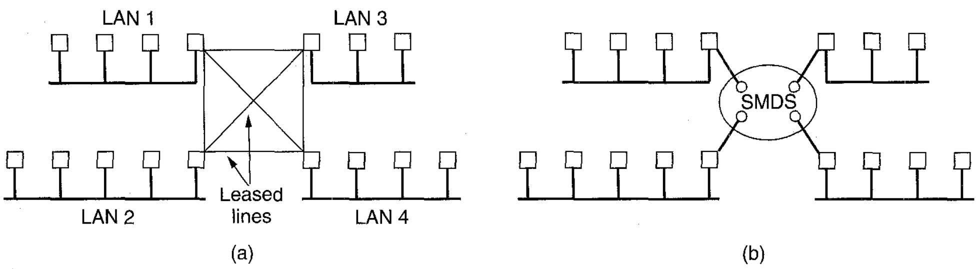

Fig. 1-27. (a) Four LANs interconnected with leased lines. (b) Interconnection using SMDS.

An alternative solution is to use SMDS, as shown in Fig. 1-27(b). The SMDS network acts like a high-speed LAN backbone, allowing packets from any LAN to flow to any other LAN. Between the LANs, in the customer's offices, and the SMDS network, in the telephone company's offices, is a (short) access line leased from the telephone company. Usually, this line is a MAN and uses DQDB, but other options may also be available.

Whereas most telephone company services are designed for continuous traffic, SMDS is designed to handle bursty traffic. In other words, once in a while a packet has to be carried from one LAN to another quickly, but much of the time there is no LAN to LAN traffic. The leased line solution of Fig. 1-27(a) has the problem of high monthly bills; once installed, the customer has to pay for the lines whether or not they are used continuously. For intermittent traffic, leased lines are an expensive solution, and SMDS is priced to compete with them. With n LANs, a fully connected leased line network requires leasing $n(n-1)/2$ possibly long (i.e., expensive) lines, whereas SMDS only requires leasing n short access lines to the nearest SMDS router.

Since the goal of SMDS is to carry LAN to LAN traffic, it must be fast enough to do the job. The standard speed is 45 Mbps, although sometimes lower speed options are available. MANs can also operate at 45 Mbps, but they are not switched, that is, to connect four LANs using a MAN, the telephone company would have to run a single wire from LAN 1 to LAN 2 to LAN 3 to LAN 4, which is only possible if they are in the same city. With SMDS, each LAN connects to a telephone company switch which routes packets through the SMDS network as needed to reach the destination, possibly traversing multiple switches in the process.

The basic SMDS service is a simple connectionless packet delivery service. The packet format is shown in Fig. 1-28. It has three fields: the destination (where the packet is to go to), the source (who sent it), and a variable length payload field for up to 9188 bytes of user data. The machine on the sending LAN that is connected to the access line puts the packet on the access line, and SMDS makes a best effort attempt to deliver it to the correct destination. No guarantee is given.

Bytes	8	8	≤ 9188
	Destination address	Source address	User data

Fig. 1-28. The SMDS packet format.

The source and destination addresses consist of a 4-bit code followed by a telephone number of up to 15 decimal digits. Each digit is coded in a separate 4-bit field. The telephone numbers contain country code, area code, and subscriber number, so the service could eventually be offered internationally. It was thought that having decimal telephone numbers as network addresses would make the new offering seem familiar to nervous users.

When a packet arrives at the SMDS network, the first router checks to make sure that the source address corresponds to the incoming line, to prevent billing fraud. If the address is incorrect, the packet is simply discarded. If it is correct, the packet is sent along toward its destination.

A useful SMDS feature is broadcasting. The customer can specify a list of SMDS telephone numbers, and be assigned a special number for the whole list. Any packet sent to that number is delivered to all members on that list. The National Association of Securities Dealers uses this feature of MCI's SMDS service to broadcast new stock prices to all of its 5000 members.

An additional user feature is address screening, on both outgoing and incoming packets. With outgoing screening, the customer can give a list of telephone numbers and specify that no packets may be sent to any other addresses. With incoming screening, only packets from certain pre-arranged telephone numbers will be accepted. When both features are enabled, the user can effectively build a private network with no SMDS connections to the outside world. For companies with confidential data, this feature is highly valuable.

The payload can contain any byte sequence the user wishes, up to 9188 bytes. SMDS does not look at it. It can contain an Ethernet packet, an IBM token ring packet, an IP packet, or anything else. Whatever is present in the payload field is moved without modification from the source LAN to the destination LAN.

SMDS handles bursty traffic as follows. The router connected to each access line contains a counter that is incremented at a constant rate, say once every 10 μsec. When a packet arrives at the router, a check is made to see if the counter is greater than the packet length, in bytes. If it is, the packet is sent without delay and the counter is decremented by the packet length. If the packet length is greater than the counter, the packet is discarded.

In effect, with a tick every 10 μsec the user may send at an *average* rate of 100,000 bytes/sec, but the burst rate may be much higher. If, for example, the line has been idle for 10 msec, the counter will be 1000, and the user will be allowed to send a 1-kilobyte burst at the full 45 Mbps, so it will be transmitted in about 180 μsec. With a 100,000 byte/sec leased line, the same kilobyte would take 10 msec. Thus SMDS offers short delays for widely spaced independent data bursts, as long as the average rate remains below the agreed upon value. This mechanism provides fast response when needed but prevents users from using up more bandwidth than they have agreed to pay for.

1.6.2. X.25 Networks

Many older public networks, especially outside the United States, follow a standard called **X.25**. It was developed during the 1970s by CCITT to provide an interface between public packet-switched networks and their customers.

The physical layer protocol, called **X.21**, specifies the physical, electrical, and procedural interface between the host and the network. Very few public networks actually support this standard, because it requires digital, rather than analog signaling on the telephone lines. As an interim measure, an analog interface similar to the familiar RS-232 standard was defined.

The data link layer standard has a number of (slightly incompatible) variations. They all are designed to deal with transmission errors on the telephone line between the user's equipment (host or terminal) and the public network (router).

The network layer protocol deals with addressing, flow control, delivery confirmation, interrupts, and related issues. Basically, it allows the user to establish virtual circuits and then send packets of up to 128 bytes on them. These packets are delivered reliably and in order. Most X.25 networks work at speeds up to 64 kbps, which makes them obsolete for many purposes. Nevertheless, they are still widespread, so readers should be aware of their existence.

X.25 is connection-oriented and supports both switched virtual circuits and permanent ones. A **switched virtual circuit** is created when one computer sends a packet to the network asking to make a call to a remote computer. Once established, packets can be sent over the connection, always arriving in order. X.25 provides flow control, to make sure a fast sender cannot swamp a slow or busy receiver.

A **permanent virtual circuit** is used the same way as a switched one, but it is set up in advance by agreement between the customer and the carrier. It is always present, and no call setup is required to use it. It is analogous to a leased line.

Because the world is still full of terminals that do not speak X.25, another set of standards was defined that describes how an ordinary (nonintelligent) terminal communicates with an X.25 public network. In effect, the user or network operator installs a "black box" to which these terminals can connect. The black box is called a **PAD** (**Packet Assembler Disassembler**), and its function is described in a document known as **X.3**. A standard protocol has been defined between the terminal and the PAD, called **X.28**; another standard protocol exists between the PAD and the network, called **X.29**. Together, these three recommendations are often called **triple X**.

1.6.3. Frame Relay

Frame relay is a service for people who want an absolute bare-bones connection-oriented way to move bits from *A* to *B* at reasonable speed and low cost (Smith, 1993). Its existence is due to changes in technology over the past two decades. Twenty years ago, communication using telephone lines was slow, analog, and unreliable, and computers were slow and expensive. As a result, complex protocols were required to mask errors, and the users' computers were too expensive to have them do this work.

The situation has changed radically. Leased telephone lines are now fast, digital, and reliable, and computers are fast and inexpensive. This suggests the use of simple protocols, with most of the work being done by the users' computers, rather than by the network. It is this environment that frame relay addresses.

Frame relay can best be thought of as a virtual leased line. The customer leases a permanent virtual circuit between two points and can then send frames (i.e., packets) of up to 1600 bytes between them. It is also possible to lease permanent virtual circuits between a given site and multiple other sites, so each frame carries a 10-bit number telling which virtual circuit to use.

The difference between an actual leased line and a virtual leased line is that with an actual one, the user can send traffic all day long at the maximum speed. With a virtual one, data bursts may be sent at full speed, but the long-term average usage must be below a predetermined level. In return, the carrier charges much less for a virtual line than a physical one.

In addition to competing with leased lines, frame relay also competes with X.25 permanent virtual circuits, except that it operates at higher speeds, usually 1.5 Mbps, and provides fewer features.

Frame relay provides a minimal service, primarily a way to determine the start and end of each frame, and detection of transmission errors. If a bad frame is received, the frame relay service simply discards it. It is up to the user to discover that a frame is missing and take the necessary action to recover. Unlike X.25, frame relay does not provide acknowledgements or normal flow control. It does have a bit in the header, however, which one end of a connection can set to indicate to the other end that problems exist. The use of this bit is up to the users.

1.6.4. Broadband ISDN and ATM

Even if the above services become popular, the telephone companies are still faced with a far more fundamental problem: multiple networks. POTS (Plain Old Telephone Service) and Telex use the old circuit-switched network. Each of the new data services such as SMDS and frame relay uses its own packet-switching network. DQDB is different from these, and the internal telephone company call management network (SSN 7) is yet another network. Maintaining all these separate networks is a major headache, and there is another network, cable television, that the telephone companies do not control and would like to.

The perceived solution is to invent a single new network for the future that will replace the entire telephone system and all the specialized networks with a single integrated network for all kinds of information transfer. This new network will have a huge data rate compared to all existing networks and services and will make it possible to offer a large variety of new services. This is not a small project, and it is certainly not going to happen overnight, but it is now under way.

The new wide area service is called **B-ISDN** (**Broadband Integrated Services Digital Network**). It will offer video on demand, live television from many sources, full motion multimedia electronic mail, CD-quality music, LAN interconnection, high-speed data transport for science and industry and many other services that have not yet even been thought of, all over the telephone line.

The underlying technology that makes B-ISDN possible is called **ATM** (**Asynchronous Transfer Mode**) because it is not synchronous (tied to a master clock), as most long distance telephone lines are. Note that the acronym ATM here has nothing to do with the Automated Teller Machines many banks provide (although an ATM machine can use an ATM network to talk to its bank).

A great deal of work has already been done on ATM and on the B-ISDN system that uses it, although there is more ahead. For more information on this subject, see (Fischer et al., 1994; Gasman, 1994; Goralski, 1995; Kim et al., 1994; Kyas, 1995; McDysan and Spohn, 1995; and Stallings, 1995a).

The basic idea behind ATM is to transmit all information in small, fixed-size packets called **cells**. The cells are 53 bytes long, of which 5 bytes are header and 48 bytes are payload, as shown in Fig. 1-29. ATM is both a technology (hidden from the users) and potentially a service (visible to the users). Sometimes the service is called **cell relay**, as an analogy to frame relay.

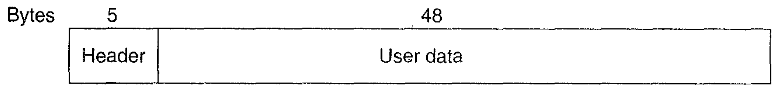

Fig. 1-29. An ATM cell.

The use of a cell-switching technology is a gigantic break with the 100-year old tradition of circuit switching (establishing a copper path) within the telephone system. There are a variety of reasons why cell switching was chosen, among them are the following. First, cell switching is highly flexible and can handle both constant rate traffic (audio, video) and variable rate traffic (data) easily. Second, at the very high speeds envisioned (gigabits per second are within reach), digital switching of cells is easier than using traditional multiplexing techniques, especially using fiber optics. Third, for television distribution, broadcasting is essential; cell switching can provide this and circuit switching cannot.

ATM networks are connection-oriented. Making a call requires first sending a message to set up the connection. After that, subsequent cells all follow the same path to the destination. Cell delivery is not guaranteed, but their order is. If cells 1 and 2 are sent in that order, then if both arrive, they will arrive in that order, never first 2 then 1.

ATM networks are organized like traditional WANs, with lines and switches (routers). The intended speeds for ATM networks are 155 Mbps and 622 Mbps, with the possibility of gigabit speeds later. The 155-Mbps speed was chosen because this is about what is needed to transmit high definition television. The exact choice of 155.52 Mbps was made for compatibility with AT&T's SONET transmission system. The 622 Mbps speed was chosen so four 155-Mbps channels could be sent over it. By now it should be clear why some of the gigabit testbeds operated at 622 Mbps: they used ATM.

When ATM was proposed, virtually all the discussion (i.e., the hype) was about video on demand to every home and replacing the telephone system, as described above. Since then, other developments have become important. Many organizations have run out of bandwidth on their campus or building-wide LANs and are being forced to go to some kind of switched system that has more bandwidth than does a single LAN. Also, in client-server computing, some applications need the ability to talk to certain servers at high speed. ATM is certainly a major candidate for both of these applications. Nevertheless, it is a bit of a letdown to go from a goal of trying to replace the entire low-speed analog telephone system with a high-speed digital one to a goal of trying connect all the Ethernets on campus. LAN interconnection using ATM is discussed in (Kavak, 1995; Newman, 1994; and Truong et al., 1995).

It is also worth pointing out that different organizations involved in ATM have different (financial) interests. The long-distance telephone carriers and PTTs are mostly interested in using ATM to upgrade the telephone system and compete with the cable TV companies in electronic video distribution. The computer vendors see campus ATM LANs as the big moneymaker (for them). All these competing interests do not make the ongoing standardization process any easier, faster, or more coherent. Also, politics and power within the organization standardizing ATM (The ATM Forum) have considerable influence on where ATM is going.

The B-ISDN ATM Reference Model

Let us now turn back to the technology of ATM, especially as used in the (future) telephone system. Broadband ISDN using ATM has its own reference model, different from the OSI model and also different from the TCP/IP model. This model is shown in Fig. 1-30. It consists of three layers, the physical, ATM, and ATM adaptation layers, plus whatever the users want to put on top of that.

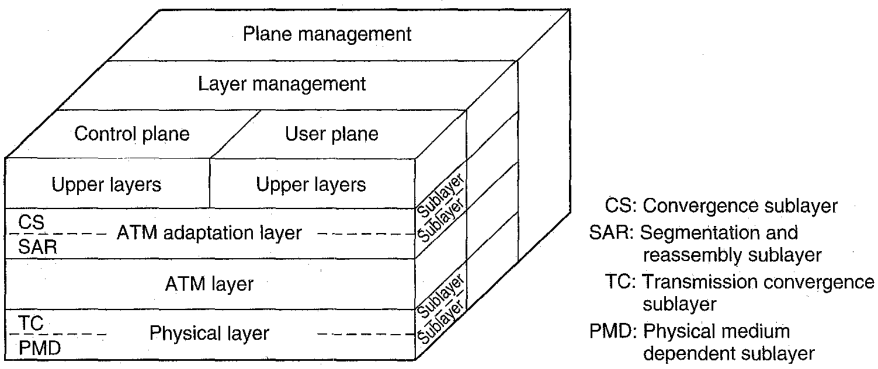

Fig. 1-30. The B-ISDN ATM reference model.

The physical layer deals with the physical medium: voltages, bit timing, and various other issues. ATM does not prescribe a particular set of rules, but instead says that ATM cells may be sent on a wire or fiber by themselves, but they may also be packaged inside the payload of other carrier systems. In other words, ATM has been designed to be independent of the transmission medium.

The **ATM layer** deals with cells and cell transport. It defines the layout of a cell and tells what the header fields mean. It also deals with establishment and release of virtual circuits. Congestion control is also located here.

Because most applications do not want to work directly with cells (although some may), a layer above the ATM layer has been defined that allows users to send packets larger than a cell. The ATM interface segments these packets, transmits the cells individually, and reassembles them at the other end. This layer is the **AAL** (**ATM Adaptation Layer**).

Unlike the earlier two-dimensional reference models, the ATM model is defined as being three-dimensional, as shown in Fig. 1-30. The **user plane** deals with data transport, flow control, error correction, and other user functions. In contrast, the **control plane** is concerned with connection management. The layer and plane management functions relate to resource management and interlayer coordination.

The physical and AAL layers are each divided into two sublayers, one at the bottom that does the work and a convergence sublayer on top that provides the proper interface to the layer above it. The functions of the layers and sublayers are given in Fig. 1-31.

OSI layer	ATM layer	ATM sublayer	Functionality
3/4	AAL	CS	Providing the standard interface (convergence)
		SAR	Segmentation and reassembly
2/3	ATM		Flow control Cell header generation/extraction Virtual circuit/path management Cell multiplexing/demultiplexing
2	Physical	TC	Cell rate decoupling Header checksum generation and verification Cell generation Packing/unpacking cells from the enclosing envelope Frame generation
1		PMD	Bit timing Physical network access

Fig. 1-31. The ATM layers and sublayers, and their functions.

The **PMD** (**Physical Medium Dependent**) sublayer interfaces to the actual cable. It moves the bits on and off and handles the bit timing. For different carriers and cables, this layer will be different.

The other sublayer of the physical layer is the **TC** (**Transmission Convergence**) sublayer. When cells are transmitted, the TC layer sends them as a string of bits to the PMD layer. Doing this is easy. At the other end, the TC sublayer gets a pure incoming bit stream from the PMD sublayer. Its job is to convert this bit stream into a cell stream for the ATM layer. It handles all the issues related to telling where cells begin and end in the bit stream. In the ATM model, this functionality is in the physical layer. In the OSI model and in pretty much all other networks, the job of framing, that is, turning a raw bit stream into a sequence of frames or cells, is the data link layer's task. For that reason we will discuss it in this book along with the data link layer, not with the physical layer.

As we mentioned earlier, the ATM layer manages cells, including their generation and transport. Most of the interesting aspects of ATM are located here. It is a mixture of the OSI data link and network layers, but it is not split into sublayers.

The AAL layer is split into a **SAR** (**Segmentation And Reassembly**) sublayer and a **CS** (**Convergence Sublayer**). The lower sublayer breaks packets up into cells on the transmission side and puts them back together again at the destination. The upper sublayer makes it possible to have ATM systems offer different kinds of services to different applications (e.g., file transfer and video on demand have different requirements concerning error handling, timing, etc.).

Perspective on ATM

To a considerable extent, ATM is a project invented by the telephone industry because after Ethernet was widely installed, the computer industry never rallied around any higher-speed network technology to make it standard. The telephone companies filled this vacuum with ATM, although in October 1991, many computer vendors joined with the telephone companies to set up the **ATM Forum**, an industry group that will guide the future of ATM.

Although ATM promises the ability to deliver information anywhere at speeds soon to exceed 1 Gbps, delivering on this promise will not be easy. ATM is basically high-speed packet-switching, a technology the telephone companies have little experience with. What they do have, is a massive investment in a different technology (circuit switching) that is in concept unchanged since the days of Alexander Graham Bell. Needless to say, this transition will not happen quickly, all the more so because it is a revolutionary change rather than an evolutionary one, and revolutions never go smoothly.

The economics of installing ATM worldwide also have to be considered. A substantial fraction of the existing telephone system will have to be replaced. Who will pay for this? How much will consumers be willing to pay to get a movie on demand electronically, when they can get one at the local video store for a couple of dollars? Finally, the question of where many of the advanced services are provided is crucial. If they are provided by the network, the telephone companies will profit from them. If they are provided by computers attached to the network, the manufacturers and operators of these devices make the profits. The users may not care, but the telephone companies and computer vendors certainly do, and this will surely affect their interest in making ATM happen.

1.6.5. Comparison of Services

The reader may be wondering why so many incompatible and overlapping services exist, including DQDB, SMDS, X.25, frame relay, ATM, and more. The underlying reason is the 1984 decision to break up AT&T and foster competition in the telecommunications industry. Different companies with different interests and technologies are now free to offer whatever services they think there is a demand for, and many of them are doing this with a vengeance.

To recap some of the services we have touched on in this chapter, DQDB is an unswitched MAN technology that allows 53-byte cells (of which 44 are payload) to be sent down long wires within a city. SMDS is a switched datagram technology for sending datagrams anywhere in a network at 45 Mbps. X.25 is an older connection-oriented networking technology for transmitting small variable-sized packets at 64 kbps. Frame relay is a service that provides virtual leased lines at speeds around 1.5 Mbps. Finally, ATM is designed to replace the entire circuit-switched telephone system with cell switching and be able to handle data and television as well. Some differences between these competitors are summarized in Fig. 1-32.

Issue	DQDB	SMDS	X.25	Frame Relay	ATM AAL
Connection oriented	Yes	No	Yes	Yes	Yes
Normal speed (Mbps)	45	45	.064	1.5	155
Switched	No	Yes	Yes	No	Yes
Fixed-size payload	Yes	No	No	No	No
Max payload	44	9188	128	1600	Variable
Permanent VCs	No	No	Yes	Yes	Yes
Multicasting	No	Yes	No	No	Yes

Fig. 1-32. Different networking services.

1.7. NETWORK STANDARDIZATION

Many network vendors and suppliers exist, each with their own ideas of how things should be done. Without coordination, there would be complete chaos, and users would be able to get nothing done. The only way out is to agree upon some network standards.

Not only do standards allow different computers to communicate, but they also increase the market for products adhering to the standard, which leads to mass production, economies of scale in manufacturing, VLSI implementations, and other benefits that decrease price and further increase acceptance. In the following sections we will take a quick look at the important, but little-known, world of international standardization.

Standards fall into two categories: de facto and de jure. **De facto** (Latin for "from the fact") standards are those that have just happened, without any formal plan. The IBM PC and its successors are de facto standards for small office computers because dozens of manufacturers have chosen to copy IBM's machines very closely. UNIX is the de facto standard for operating systems in university computer science departments.

De jure (Latin for "by law") standards, in contrast, are formal, legal standards adopted by some authorized standardization body. International standardization authorities are generally divided into two classes: those established by treaty among national governments, and voluntary, nontreaty organizations. In the area of computer network standards, there are several organizations of each type, which are discussed below.

1.7.1. Who's Who in the Telecommunications World

The legal status of the world's telephone companies varies considerably from country to country. At one extreme is the United States, which has 1500 separate, privately owned telephone companies. Before it was broken up in 1984, AT&T, at that time the world's largest corporation, completely dominated the scene. It provided telephone service to about 80 percent of America's telephones, spread throughout half of its geographical area, with all the other companies combined servicing the remaining (mostly rural) customers. Since the breakup, AT&T continues to provide long-distance service, although now in competition with other companies. The seven Regional Bell Operating Companies that were split off from AT&T and 1500 independents provide local and cellular telephone service. Some of these independents, such as GTE, are very large companies.

Companies in the United States that provide communication services to the public are called **common carriers**. Their offerings and prices are described by a document called a **tariff**, which must be approved by the Federal Communications Commission for the interstate and international traffic, and by the state public utilities commissions for intrastate traffic.

At the other extreme are countries in which the national government has a complete monopoly on all communication, including the mail, telegraph, telephone, and often radio and television as well. Most of the world falls in this category. In some cases the telecommunication authority is a nationalized company, and in others it is simply a branch of the government, usually known as the **PTT** (**Post, Telegraph & Telephone** administration). Worldwide, the trend is toward liberalization and competition and away from government monopoly.

With all these different suppliers of services, there is clearly a need to provide compatibility on a worldwide scale to ensure that people (and computers) in one country can call their counterparts in another one. Actually, this need has existed for a long time. In 1865, representatives from many European governments met to form the predecessor to today's **ITU** (**International Telecommunication Union**). ITU's job was standardizing international telecommunications, which in those days meant telegraphy. Even then it was clear that if half the countries used Morse code and the other half used some other code, there was going to be a problem. When the telephone was put into international service, ITU took over the job of standardizing telephony as well. In 1947, ITU became an agency of the United Nations.

ITU has three main sectors:

1. Radiocommunications Sector (ITU-R).
2. Telecommunications Standardization Sector (ITU-T).
3. Development Sector (ITU-D).

ITU-R is concerned with allocating radio frequencies worldwide to the competing interest groups. We will be primarily concerned with ITU-T, which is concerned with telephone and data communication systems. From 1956 to 1993, ITU-T was known as **CCITT**, an acronym for its French name: Comité Consultatif International Télégraphique et Téléphonique. On March 1, 1993, CCITT was reorganized to make it less bureaucratic and renamed to reflect its new role. Both ITU-T and CCITT issued recommendations in the area of telephone and data communications. One still frequently runs into CCITT recommendations, such as CCITT X.25, although since 1993 recommendations bear the ITU-T label.

ITU-T has five classes of members:

1. Administrations (national PTTs).
2. Recognized private operators (e.g., AT&T, MCI, British Telecom).
3. Regional telecommunications organizations (e.g., the European ETSI).
4. Telecommunications vendors and scientific organizations.
5. Other interested organizations (e.g., banking and airline networks).

ITU-T has about 200 administrations, 100 private operators, and several hundred other members. Only administrations may vote, but all members may participate in ITU-T's work. Since the United States does not have a PTT, somebody else had to represent it in ITU-T. This task fell to the State Department, probably on the grounds that ITU-T had to do with foreign countries, the State Department's specialty.

ITU-T's task is to make technical recommendations about telephone, telegraph, and data communication interfaces. These often become internationally

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recognized standards, for example, V.24 (also known as EIA RS-232 in the United States), which specifies the placement and meaning of the various pins on the connector used by most asynchronous terminals.

It should be noted that ITU-T recommendations are technically only suggestions that governments can adopt or ignore, as they wish. In practice, a country that wishes to adopt a different telephone standard than the rest of the world is free to do so, but at the price of cutting itself off from everyone else. This might work for Albania, but elsewhere it would be a real problem. The fiction of calling ITU-T standards "recommendations" was and is necessary to keep nationalist forces in many countries placated.

The real work of ITU-T is done in Study Groups, often as large as 400 people. To make it possible to get anything at all done, the Study Groups are divided into Working Parties, which are in turn divided into Expert Teams, which are in turn divided into ad hoc groups. Once a bureaucracy, always a bureaucracy.

Despite all this, ITU-T actually gets things done. Its current output runs to about 5000 pages of recommendations a year. The members chip in to cover ITU's costs. Big, rich countries are supposed to pay up to 30 contributory units a year; small, poor ones can get away with 1/16 of a contributory unit (a contributory unit is about 250,000 dollars). It is a testimony to ITU-T's value that pretty much everyone pays their fair share, even though contributions are completely voluntary.

As telecommunications completes the transition started in the 1980s from being entirely national to being entirely global, standards will become increasingly important, and more and more organizations will want to become involved in setting them. For more information about ITU, see (Irmer, 1994).

1.7.2. Who's Who in the International Standards World

International standards are produced by **ISO** (**International Standards Organization**†), a voluntary, nontreaty organization founded in 1946. Its members are the national standards organizations of the 89 member countries. These members include ANSI (U.S.), BSI (Great Britain), AFNOR (France), DIN (Germany), and 85 others.

ISO issues standards on a vast number of subjects, ranging from nuts and bolts (literally) to telephone pole coatings. Over 5000 standards have been issued, including the OSI standards. ISO has almost 200 Technical Committees, numbered in the order of their creation, each dealing with a specific subject. TC1 deals with the nuts and bolts (standardizing screw thread pitches). TC97 deals with computers and information processing. Each TC has subcommittees (SCs) divided into working groups (WGs).

The real work is done largely in the WGs by over 100,000 volunteers

† For the purist, ISO's true name is the International Organization for Standardization.

worldwide. Many of these "volunteers" are assigned to work on ISO matters by their employers, whose products are being standardized. Others are government officials keen on having their country's way of doing things become the international standard. Academic experts also are active in many of the WGs.

On issues of telecommunication standards, ISO and ITU-T often cooperate (ISO is a member of ITU-T) to avoid the irony of two official and mutually incompatible international standards.

The U.S. representative in ISO is **ANSI** (**American National Standards Institute**), which despite its name, is a private, nongovernmental, nonprofit organization. Its members are manufacturers, common carriers, and other interested parties. ANSI standards are frequently adopted by ISO as international standards.

The procedure used by ISO for adopting standards is designed to achieve as broad a consensus as possible. The process begins when one of the national standards organizations feels the need for an international standard in some area. A working group is then formed to come up with a **CD** (**Committee Draft**). The CD is then circulated to all the member bodies, which get 6 months to criticize it. If a substantial majority approves, a revised document, called a **DIS** (**Draft International Standard**) is produced and circulated for comments and voting. Based on the results of this round, the final text of the **IS** (**International Standard**) is prepared, approved, and published. In areas of great controversy, a CD or DIS may have to go through several versions before acquiring enough votes, and the whole process can take years.

NIST (**National Institute of Standards and Technology**) is an agency of the U.S. Dept. of Commerce. It was formerly known as the National Bureau of Standards. It issues standards that are mandatory for purchases made by the U.S. Government, except for those of the Department of Defense, which has its own standards.

Another major player in the standards world is **IEEE** (**Institute of Electrical and Electronics Engineers**), the largest professional organization in the world. In addition to publishing scores of journals and running numerous conferences each year, IEEE has a standardization group that develops standards in the area of electrical engineering and computing. IEEE's 802 standard for local area networks is the key standard for LANs. It has subsequently been taken over by ISO as the basis for ISO 8802.

1.7.3. Who's Who in the Internet Standards World

The worldwide Internet has its own standardization mechanisms, very different from those of ITU-T and ISO. The difference can be crudely summed up by saying that the people who come to ITU or ISO standardization meetings wear suits. The people who come to Internet standardization meetings wear either jeans or military uniforms.

ITU-T and ISO meetings are populated by corporate officials and government civil servants for whom standardization is their job. They regard standardization as a good thing and devote their lives to it. Internet people, on the other hand, definitely prefer anarchy as a matter of principle, but sometimes agreement is needed to make things work. Thus standards, however regrettable, are occasionally needed.

When the ARPANET was set up, DoD created an informal committee to oversee it. In 1983, the committee was renamed the **IAB** (**Internet Activities Board**) and given a slighter broader mission, namely, to keep the researchers involved with the ARPANET and Internet pointed more-or-less in the same direction, an activity not unlike herding cats. The meaning of the acronym "IAB" was later changed to **Internet Architecture Board.**

Each of the approximately ten members of the IAB headed a task force on some issue of importance. The IAB met several times a year to discuss results and give feedback to the DoD and NSF, which were providing most of the funding at this time. When a standard was needed (e.g., a new routing algorithm), the IAB members would thrash it out and then announce the change so the graduate students who were the heart of the software effort could implement it. Communication was done by a series of technical reports called **RFCs** (**Request For Comments**). RFCs are stored on-line and can be fetched by anyone interested in them. They are numbered in chronological order of creation. Close to 2000 now exist.

By 1989, the Internet had grown so large that this highly informal style no longer worked. Many vendors by then offered TCP/IP products and did not want to change them just because ten researchers had thought of a better idea. In the summer of 1989, the IAB was reorganized again. The researchers were moved to the **IRTF** (**Internet Research Task Force**), which was made subsidiary to IAB, along with the **IETF** (**Internet Engineering Task Force**). The IAB was repopulated with people representing a broader range of organizations than just the research community. It was initially a self-perpetuating group, with members serving for a 2-year term, and new members being appointed by the old ones. Later, the **Internet Society** was created, populated by people interested in the Internet. The Internet Society is thus in a sense comparable to ACM or IEEE. It is governed by elected trustees who appoint the IAB members.

The idea of this split was to have the IRTF concentrate on long-term research, while the IETF dealt with short-term engineering issues. The IETF was divided up into working groups, each with a specific problem to solve. The chairmen of these working groups initially met together as a steering committee to direct the engineering effort. The working group topics include new applications, user information, OSI integration, routing and addressing, security, network management, and standards. Eventually, so many working groups were formed (more than 70) that they were grouped into areas, and the area chairmen met as the steering committee.

In addition, a more formal standardization process was adopted, patterned after ISOs. To become a **Proposed Standard**, the basic idea must be completely explained in an RFC and have sufficient interest in the community to warrant consideration. To advance to the **Draft Standard** stage, there must be a working implementation that has been thoroughly tested by at least two independent sites for 4 months. If the IAB is convinced that the idea is sound and the software works, it can declare the RFC to be an Internet Standard. Some Internet Standards have become DoD standards (MIL-STD), making them mandatory for DoD suppliers. David Clark once made a now-famous remark about Internet standardization consisting of "rough consensus and running code."

1.8. OUTLINE OF THE REST OF THE BOOK

This book discusses both the principles and practice of computer networking. Most chapters start with a discussion of the relevant principles, followed by a number of examples that illustrate these principles. Two networks are used as running examples throughout the text: the Internet and ATM networks. In a way, the two are complementary: ATM is mostly concerned with the lower layers, and the Internet is mostly concerned with upper layers. In the future, the Internet may run largely on an ATM backbone, so both of them may coexist. Other examples will be given where relevant.

The book is structured according to the hybrid model of Fig. 1-21. Starting with Chap. 2, we begin working our way up the protocol hierarchy beginning at the bottom. The second chapter provides some background in the field of data communication. It covers analog and digital transmission, multiplexing, switching, and the telephone system, past current, and future. This material is concerned with the physical layer, although we cover only the architectural rather than the hardware aspects. Several examples of the physical layer are also discussed, such as SONET and cellular radio.

Chap. 3 discusses the data link layer and its protocols by means of a number of increasingly complex examples. The analysis of these protocols is also covered. After that, some important real-world protocols are discussed, including HDLC (used in low- and medium-speed networks), SLIP, and PPP (used in the Internet), and ATM (used in B-ISDN).

Chap. 4 concerns the medium access sublayer, which is part of the data link layer. The basic question it deals with is how to determine who may use the network next when the network consists of a single shared channel, as in most LANs and some satellite networks. Many examples are given from the areas of LANs, fiber optic networks, and satellite networks. Bridges, which are used to connect LANs together, are also discussed here.

Chap. 5 deals with the network layer, especially routing, congestion control, and internetworking. It discusses both static and dynamic routing algorithms. Broadcast routing is also covered. The effect of poor routing, congestion, is

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discussed in some detail. Connecting heterogeneous networks together to form internetworks leads to numerous problems that are discussed here. The network layers in the Internet and ATM networks are given extensive coverage.

Chap. 6 deals with the transport layer. Much of the emphasis is on connection-oriented protocols, since many applications need these. An example transport service and its implementation are discussed in detail. Both the Internet transport protocols (TCP and UDP) and the ATM transport protocols (AAL 1-5) are covered in detail.

The OSI session and presentation layers are not discussed in this book as they are not widely used for anything.

Chapter 7 deals with the application layer, its protocols and applications. Among the applications covered are security, naming, electronic mail, net news, network management, the World Wide Web, and multimedia.

Chap. 8 contains an annotated list of suggested readings arranged by chapter. It is intended to help those readers who would like to pursue their study of networking further. The chapter also has an alphabetical bibliography of all references cited in this book.

1.9. SUMMARY

Computer networks can be used for numerous services, both for companies and for individuals. For companies, networks of personal computers using shared servers often provide flexibility and a good price/performance ratio. For individuals, networks offer access to a variety of information and entertainment resources.

Roughly speaking, networks can be divided up into LANs, MANs, WANs, and internetworks, each with their own characteristics, technologies, speeds, and niches. LANs cover a building, MANs cover a city, and WANs cover a country or continent. LANs and MANs are unswitched (i.e., do not have routers); WANs are switched.

Network software consists of protocols, or rules by which processes can communicate. Protocols can be either connectionless or connection-oriented. Most networks support protocol hierarchies, with each layer providing services to the layers above it and insulating them from the details of the protocols used in the lower layers. Protocol stacks are typically based either on the OSI model or the TCP/IP model. Both of these have network, transport, and application layers, but they differ on the other layers.

Well-known networks have included Novell's NetWare, the ARPANET (now defunct), NSFNET, the Internet, and various gigabit testbeds. Network services have included DQDB, SMDS, X.25, frame relay, and broadband ISDN. All of these are available commercially, from a variety of suppliers. The marketplace will determine which ones will survive and which ones will not.

PROBLEMS

1. In the future, when everyone has a home terminal connected to a computer network, instant public referendums on important pending legislation will become possible. Ultimately, existing legislatures could be eliminated, to let the will of the people be expressed directly. The positive aspects of such a direct democracy are fairly obvious; discuss some of the negative aspects.

2. An alternative to a LAN is simply a big timesharing system with terminals for all users. Give two advantages of a client-server system using a LAN.

3. A collection of five routers is to be connected in a point-to-point subnet. Between each pair of routers, the designers may put a high-speed line, a medium-speed line, a low-speed line, or no line. If it takes 100 ms of computer time to generate and inspect each topology, how long will it take to inspect all of them to find the one that best matches the expected load?

4. A group of $2^n - 1$ routers are interconnected in a centralized binary tree, with a router at each tree node. Router i communicates with router j by sending a message to the root of the tree. The root then sends the message back down to j. Derive an approximate expression for the mean number of hops per message for large n, assuming that all router pairs are equally likely.

5. A disadvantage of a broadcast subnet is the capacity wasted due to multiple hosts attempting to access the channel at the same time. As a simplistic example, suppose that time is divided into discrete slots, with each of the n hosts attempting to use the channel with probability p during each slot. What fraction of the slots are wasted due to collisions?

6. What are the SAP addresses in FM radio broadcasting?

7. What is the principal difference between connectionless communication and connection-oriented communication?

8. Two networks each provide reliable connection-oriented service. One of them offers a reliable byte stream and the other offers a reliable message stream. Are these identical? If so, why is the distinction made? If not, give an example of how they differ.

9. What is the difference between a confirmed service and an unconfirmed service? For each of the following, tell whether it might be a confirmed service, an unconfirmed service, both, or neither.
   (a) Connection establishment.
   (b) Data transmission.
   (c) Connection release.

10. What does "negotiation" mean when discussing network protocols? Give an example of it.

11. What are two reasons for using layered protocols?

12. List two ways in which the OSI reference model and the TCP/IP reference model are the same. Now list two ways in which they differ.

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13. The president of the Specialty Paint Corp. gets the idea to work together with a local beer brewer for the purpose of producing an invisible beer can (as an anti-litter measure). The president tells her legal department to look into it, and they in turn ask engineering for help. As a result, the chief engineer calls his counterpart at the other company to discuss the technical aspects of the project. The engineers then report back to their respective legal departments, which then confer by telephone to arrange the legal aspects. Finally, the two corporate presidents discuss the financial side of the deal. Is this an example of a multilayer protocol in the sense of the OSI model?

14. In most networks, the data link layer handles transmission errors by requesting damaged frames to be retransmitted. If the probability of a frame's being damaged is p, what is the mean number of transmissions required to send a frame if acknowledgements are never lost?

15. Which of the OSI layers handles each of the following:
(a) Breaking the transmitted bit stream into frames.
(b) Determining which route through the subnet to use.

16. Do TPDUs encapsulate packets or the other way around? Discuss.

17. A system has an n-layer protocol hierarchy. Applications generate messages of length M bytes. At each of the layers, an h-byte header is added. What fraction of the network bandwidth is filled with headers?

18. What is the main difference between TCP and UDP?

19. Does the Novell NetWare architecture look more like X.25 or like the Internet? Explain your answer.

20. The Internet is roughly doubling in size every 18 months. Although no one really knows for sure, one estimate put the number of hosts on it at 7 million in January 1996. Use these data to compute the expected number of Internet hosts in the year 2008.

21. Why was SMDS designed as a connectionless network and frame relay as a connection-oriented one?

22. Imagine that you have trained your St. Bernard, Bernie, to carry a box of three 8mm Exabyte tapes instead of a flask of brandy. (When your disk fills up, you consider that an emergency.) These tapes each contain 7 gigabytes. The dog can travel to your side, wherever you may be, at 18 km/hour. For what range of distances does Bernie have a higher data rate than a 155-Mbps ATM line?

23. When transferring a file between two computers, (at least) two acknowledgement strategies are possible. In the first one, the file is chopped up into packets, which are individually acknowledged by the receiver, but the file transfer as a whole is not acknowledged. In the second one, the packets are not acknowledged individually, but the entire file is acknowledged when it arrives. Discuss these two approaches.

24. Imagine that the SMDS packet of Fig. 1-28 were to be incorporated in OSI protocol hierarchy. In which layer would it appear?

25. Give an advantage and a disadvantage of frame relay over a leased telephone line.

26. Why does ATM use small, fixed-length cells?

27. List two advantages and two disadvantages of having international standards for network protocols.

28. When a system has a permanent part and a removable part, such as a diskette drive and the diskette, it is important that the system be standardized, so that different companies can make both the permanent and removable parts and have everything work together. Give three examples outside the computer industry where such international standards exist. Now give three areas outside the computer industry where they do not exist.

2

THE PHYSICAL LAYER

In this chapter we will look at the lowest layer depicted in the hierarchy of Fig. 1-21. We will begin with a theoretical analysis of data transmission, only to discover that Mother (Parent?) Nature puts some limits on what can be sent over a channel.

Then we will cover transmission media, both guided (copper wire and fiber optics) and unguided (wireless). This material will provide background information on the key transmission technologies used in modern networks.

The remainder of the chapter is devoted to examples of communication systems that use these underlying transmission media. We will start with the telephone system, looking at three different versions: the current (partly) analog system, a potential digital system for the near future (N-ISDN), and a likely digital system for the distant future (ATM). Then we will look at two wireless systems, cellular radio and communication satellites.

2.1. THE THEORETICAL BASIS FOR DATA COMMUNICATION

Information can be transmitted on wires by varying some physical property such as voltage or current. By representing the value of this voltage or current as a single-valued function of time, $f(t)$, we can model the behavior of the signal and analyze it mathematically. This analysis is the subject of the following sections.

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2.1.1. Fourier Analysis

In the early 19th Century, the French mathematician Jean-Baptiste Fourier proved that any reasonably behaved periodic function, $g(t)$, with period T can be constructed by summing a (possibly infinite) number of sines and cosines:

$$g(t) = \frac{1}{2}c + \sum_{n=1}^{\infty} a_n \sin(2\pi nft) + \sum_{n=1}^{\infty} b_n \cos(2\pi nft) \qquad (2\text{-}1)$$

where $f = 1/T$ is the fundamental frequency and a_n and b_n are the sine and cosine amplitudes of the nth **harmonics** (terms). Such a decomposition is called a **Fourier series**. From the Fourier series, the function can be reconstructed; that is, if the period, T, is known and the amplitudes are given, the original function of time can be found by performing the sums of Eq. (2-1).

A data signal that has a finite duration (which all of them do) can be handled by just imagining that it repeats the entire pattern over and over forever (i.e., the interval from T to $2T$ is the same as from 0 to T, etc.).

The a_n amplitudes can be computed for any given $g(t)$ by multiplying both sides of Eq. (2-1) by $\sin(2\pi kft)$ and then integrating from 0 to T. Since

$$\int_0^T \sin(2\pi kft)\sin(2\pi nft)\,dt = \begin{cases} 0 \text{ for } k \neq n \\ T/2 \text{ for } k = n \end{cases}$$

only one term of the summation survives: a_n. The b_n summation vanishes completely. Similarly, by multiplying Eq. (2-1) by $\cos(2\pi kft)$ and integrating between 0 and T, we can derive b_n. By just integrating both sides of the equation as it stands, c can be found. The results of performing these operations are as follows:

$$a_n = \frac{2}{T}\int_0^T g(t)\sin(2\pi nft)\,dt \qquad b_n = \frac{2}{T}\int_0^T g(t)\cos(2\pi nft)\,dt \qquad c = \frac{2}{T}\int_0^T g(t)\,dt$$

2.1.2. Bandwidth-Limited Signals

To see what all this has to do with data communication, let us consider a specific example: the transmission of the ASCII character "b" encoded in an 8-bit byte. The bit pattern that is to be transmitted is 01100010. The left-hand part of Fig. 2-1(a) shows the voltage output by the transmitting computer. The Fourier analysis of this signal yields the coefficients:

$$a_n = \frac{1}{\pi n}[\cos(\pi n/4) - \cos(3\pi n/4) + \cos(6\pi n/4) - \cos(7\pi n/4)]$$

$$b_n = \frac{1}{\pi n}[\sin(3\pi n/4) - \sin(\pi n/4) + \sin(7\pi n/4) - \sin(6\pi n/4)]$$

$$c = 3/8$$

The root-mean-square amplitudes, $\sqrt{a_n^2 + b_n^2}$, for the first few terms are shown on the right-hand side of Fig. 2-1(a). These values are of interest because their squares are proportional to the energy transmitted at the corresponding frequency.

No transmission facility can transmit signals without losing some power in the process. If all the Fourier components were equally diminished, the resulting signal would be reduced in amplitude but not distorted [i.e., it would have the same nice squared-off shape as Fig. 2-1(a)]. Unfortunately, all transmission facilities diminish different Fourier components by different amounts, thus introducing distortion. Usually, the amplitudes are transmitted undiminished from 0 up to some frequency f_c [measured in cycles/sec or Hertz (Hz)] with all frequencies above this cutoff frequency strongly attenuated. In some cases this is a physical property of the transmission medium, and in other cases a filter is intentionally introduced into the circuit to limit the amount of (scarce) bandwidth available to each customer.

Now let us consider how the signal of Fig. 2-1(a) would look if the bandwidth were so low that only the lowest frequencies were transmitted [i.e., the function were being approximated by the first few terms of Eq. (2-1)]. Figure 2-1(b) shows the signal that results from a channel that allows only the first harmonic (the fundamental, *f*) to pass through. Similarly, Fig. 2-1(c)-(e) show the spectra and reconstructed functions for higher bandwidth channels.

The time *T* required to transmit the character depends on both the encoding method and the signaling speed [the number of times per second that the signal changes its value (e.g., its voltage)]. The number of changes per second is measured in **baud**. A *b* baud line does not necessarily transmit *b* bits/sec, since each signal might convey several bits. If the voltages 0, 1, 2, 3, 4, 5, 6, and 7 were used, each signal value could be used to convey 3 bits, so the bit rate would be three times the baud rate. In our example, only 0s and 1s are being used as signal levels, so the bit rate is equal to the baud rate.

Given a bit rate of *b* bits/sec, the time required to send 8 bits (for example) is $8/b$ sec, so the frequency of the first harmonic is $b/8$ Hz. An ordinary telephone line, often called a **voice-grade line**, has an artificially introduced cutoff frequency near 3000 Hz. This restriction means that the number of the highest harmonic passed through is $3000/(b/8)$ or $24,000/b$, roughly (the cutoff is not sharp).

For some data rates, the numbers work out as shown in Fig. 2-2. From these numbers, it is clear that trying to send at 9600 bps over a voice-grade telephone line will transform Fig. 2-1(a) into something looking like Fig. 2-1(c), making accurate reception of the original binary bit stream tricky. It should be obvious that at data rates much higher than 38.4 kbps there is no hope at all for *binary* signals, even if the transmission facility is completely noiseless. In other words, limiting the bandwidth limits the data rate, even for perfect channels. However, sophisticated coding schemes that use several voltage levels do exist and can achieve higher data rates. We will discuss these later in this chapter.

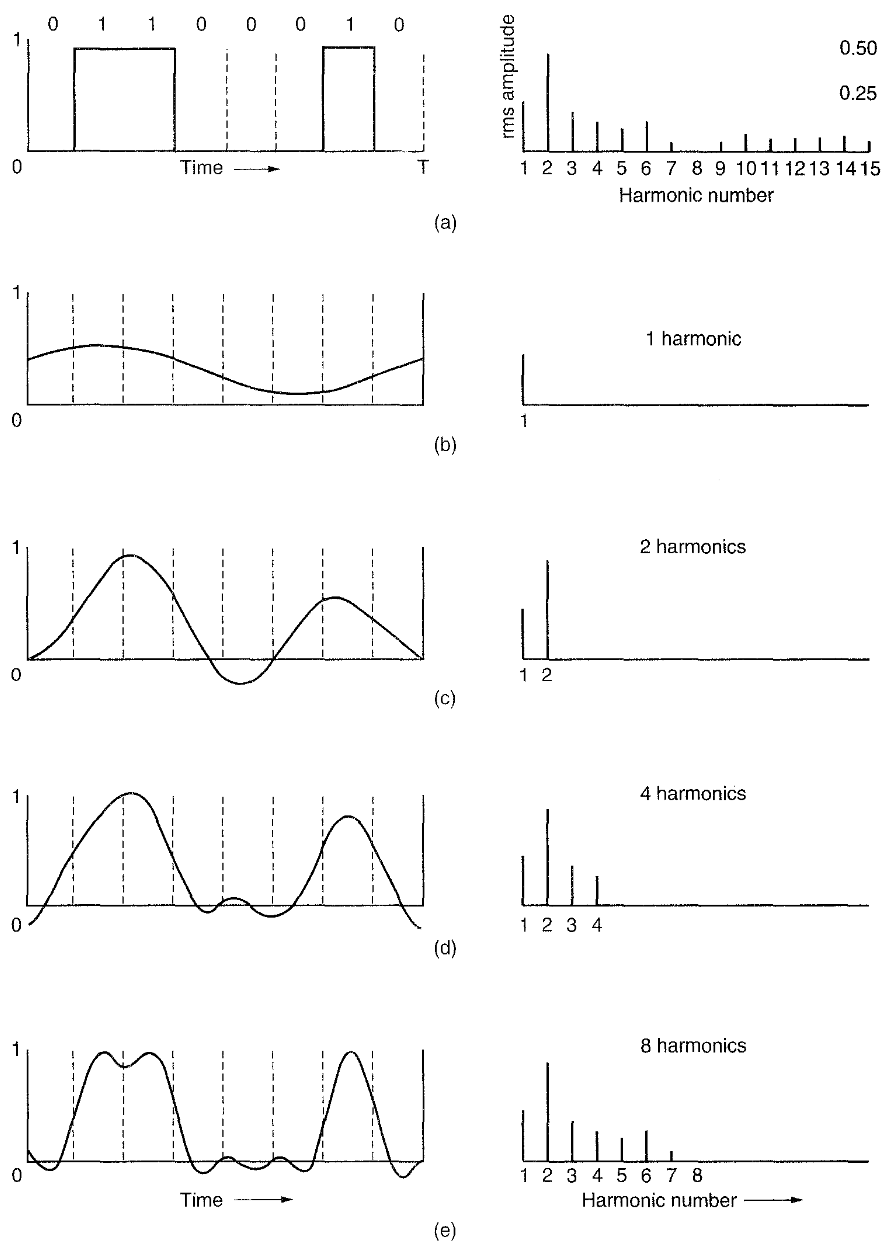

Fig. 2-1. (a) A binary signal and its root-mean-square Fourier amplitudes. (b)-(e) Successive approximations to the original signal.

Bps	T (msec)	First harmonic (Hz)	# Harmonics sent
300	26.67	37.5	80
600	13.33	75	40
1200	6.67	150	20
2400	3.33	300	10
4800	1.67	600	5
9600	0.83	1200	2
19200	0.42	2400	1
38400	0.21	4800	0

Fig. 2-2. Relation between data rate and harmonics.

2.1.3. The Maximum Data Rate of a Channel

As early as 1924, H. Nyquist realized the existence of this fundamental limit and derived an equation expressing the maximum data rate for a finite bandwidth noiseless channel. In 1948, Claude Shannon carried Nyquist's work further and extended it to the case of a channel subject to random (that is, thermodynamic) noise (Shannon, 1948). We will just briefly summarize their now classical results here.

Nyquist proved that if an arbitrary signal has been run through a low-pass filter of bandwidth H, the filtered signal can be completely reconstructed by making only $2H$ (exact) samples per second. Sampling the line faster than $2H$ times per second is pointless because the higher frequency components that such sampling could recover have already been filtered out. If the signal consists of V discrete levels, Nyquist's theorem states:

$$\text{maximum data rate} = 2H \log_2 V \text{ bits/sec}$$

For example, a noiseless 3-kHz channel cannot transmit binary (i.e., two-level) signals at a rate exceeding 6000 bps.

So far we have considered only noiseless channels. If random noise is present, the situation deteriorates rapidly. The amount of thermal noise present is measured by the ratio of the signal power to the noise power, called the **signal-to-noise ratio**. If we denote the signal power by S and the noise power by N, the signal-to-noise ratio is S/N. Usually, the ratio itself is not quoted; instead, the quantity $10 \log_{10} S/N$ is given. These units are called **decibels** (dB). An S/N ratio of 10 is 10 dB, a ratio of 100 is 20 dB, a ratio of 1000 is 30 dB and so on. The manufacturers of stereo amplifiers often characterize the bandwidth (frequency range) over which their product is linear by giving the 3-dB frequency on each end. These are the points at which the amplification factor has been approximately halved.

Shannon's major result is that the maximum data rate of a noisy channel whose bandwidth is H Hz, and whose signal-to-noise ratio is S/N, is given by

$$\text{maximum number of bits/sec} = H \log_2 (1 + S/N)$$

For example, a channel of 3000-Hz bandwidth, and a signal to thermal noise ratio of 30 dB (typical parameters of the analog part of the telephone system) can never transmit much more than 30,000 bps, no matter how many or few signal levels are used and no matter how often or how infrequent samples are taken. Shannon's result was derived using information-theory arguments and applies to any channel subject to Gaussian (thermal) noise. Counterexamples should be treated in the same category as perpetual motion machines. It should be noted, however, that this is only an upper bound and real systems rarely achieve it.

2.2. TRANSMISSION MEDIA

The purpose of the physical layer is to transport a raw bit stream from one machine to another. Various physical media can be used for the actual transmission. Each one has its own niche in terms of bandwidth, delay, cost, and ease of installation and maintenance. Media are roughly grouped into guided media, such as copper wire and fiber optics, and unguided media, such as radio and lasers through the air. We will look at these in this section and the next one.

2.2.1. Magnetic Media

One of the most common ways to transport data from one computer to another is to write them onto magnetic tape or floppy disks, physically transport the tape or disks to the destination machine, and read them back in again. While this method is not as sophisticated as using a geosynchronous communication satellite, it is often much more cost effective, especially for applications in which high bandwidth or cost per bit transported is the key factor.

A simple calculation will make this point clear. An industry standard 8-mm video tape (e.g., Exabyte) can hold 7 gigabytes. A box $50 \times 50 \times 50$ cm can hold about 1000 of these tapes, for a total capacity of 7000 gigabytes. A box of tapes can be delivered anywhere in the United States in 24 hours by Federal Express and other companies. The effective bandwidth of this transmission is 56 gigabits/86400 sec or 648 Mbps, which is slightly better than the high-speed version of ATM (622 Mbps). If the destination is only an hour away by road, the bandwidth is increased to over 15 Gbps.

For a bank with gigabytes of data to be backed up daily on a second machine (so the bank can continue to function even in the face of a major flood or earthquake) it is likely that no other transmission technology can even begin to approach magnetic tape for performance.

If we now look at cost, we get a similar picture. The cost of 1000 video tapes is perhaps 5000 dollars when bought in bulk. A video tape can be reused at least ten times, so the tape cost is maybe 500 dollars. Add to this another 200 dollars for shipping, and we have a cost of roughly 700 dollars to ship 7000 gigabytes. This amounts to 10 cents per gigabyte. No network carrier on earth can compete with that. The moral of the story is:

> *Never underestimate the bandwidth of a station wagon full of tapes hurtling down the highway.*

2.2.2. Twisted Pair

Although the bandwidth characteristics of magnetic tape are excellent, the delay characteristics are poor. Transmission time is measured in minutes or hours, not milliseconds. For many applications an on-line connection is needed. The oldest and still most common transmission medium is **twisted pair**. A twisted pair consists of two insulated copper wires, typically about 1 mm thick. The wires are twisted together in a helical form, just like a DNA molecule. The purpose of twisting the wires is to reduce electrical interference from similar pairs close by. (Two parallel wires constitute a simple antenna; a twisted pair does not.)

The most common application of the twisted pair is the telephone system. Nearly all telephones are connected to the telephone company office by a twisted pair. Twisted pairs can run several kilometers without amplification, but for longer distances, repeaters are needed. When many twisted pairs run in parallel for a substantial distance, such as all the wires coming from an apartment building to the telephone company office, they are bundled together and encased in a protective sheath. The pairs in these bundles would interfere with one another if it were not for the twisting. In parts of the world where telephone lines run on poles above ground, it is common to see bundles several centimeters in diameter.

Twisted pairs can be used for either analog or digital transmission. The bandwidth depends on the thickness of the wire and the distance traveled, but several megabits/sec can be achieved for a few kilometers in many cases. Due to their adequate performance and low cost, twisted pairs are widely used and are likely to remain so for years to come.

Twisted pair cabling comes in several varieties, two of which are important for computer networks. **Category 3** twisted pairs consist of two insulated wires gently twisted together. Four such pairs are typically grouped together in a plastic sheath for protection and to keep the eight wires together. Prior to about 1988, most office buildings had one category 3 cable running from a central **wiring closet** on each floor into each office. This scheme allowed up to four regular telephones or two multiline telephones in each office to connect to the telephone company equipment in the wiring closet.

Starting around 1988, the more advanced **category 5** twisted pairs were introduced. They are similar to category 3 pairs, but with more twists per centimeter and Teflon insulation, which results in less crosstalk and a better quality signal over longer distances, making them more suitable for high-speed computer communication. Both of these wiring types are often referred to as **UTP** (**Unshielded Twisted Pair**), to contrast them with the bulky, expensive, shielded twisted pair cables IBM introduced in the early 1980s, but which have not proven popular outside of IBM installations.

2.2.3. Baseband Coaxial Cable

Another common transmission medium is the **coaxial cable** (known to its many friends as just "coax"). It has better shielding than twisted pairs, so it can span longer distances at higher speeds. Two kinds of coaxial cable are widely used. One kind, 50-ohm cable, is commonly used for digital transmission and is the subject of this section. The other kind, 75-ohm cable, is commonly used for analog transmission and will be described in the next section. This distinction is based on historical, rather than technical, factors (e.g., early dipole antennas had an impedance of 300 ohms, and it was easy to build 4:1 impedance matching transformers).

A coaxial cable consists of a stiff copper wire as the core, surrounded by an insulating material. The insulator is encased by a cylindrical conductor, often as a closely woven braided mesh. The outer conductor is covered in a protective plastic sheath. A cutaway view of a coaxial cable is shown in Fig. 2-3.

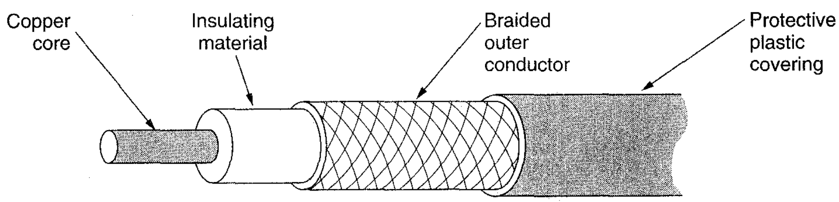

Fig. 2-3. A coaxial cable.

The construction and shielding of the coaxial cable give it a good combination of high bandwidth and excellent noise immunity. The bandwidth possible depends on the cable length. For 1-km cables, a data rate of 1 to 2 Gbps is feasible. Longer cables can also be used, but only at lower data rates or with periodic amplifiers. Coaxial cables used to be widely used within the telephone system but have now largely been replaced by fiber optics on long-haul routes. In the United States alone, 1000 km of fiber is installed every day (counting a 100-km bundle

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with 10 strands of fiber as 1000 km). Sprint is already 100 percent fiber, and the other major carriers are rapidly approaching that. Coax is still widely used for cable television and some local area networks, however.

2.2.4. Broadband Coaxial Cable

The other kind of coaxial cable system uses analog transmission on standard cable television cabling. It is called **broadband**. Although the term "broadband" comes from the telephone world, where it refers to anything wider than 4 kHz, in the computer networking world "broadband cable" means any cable network using analog transmission (see Cooper, 1986).

Since broadband networks use standard cable television technology, the cables can be used up to 300 MHz (and often up to 450 MHz) and can run for nearly 100 km due to the analog signaling, which is much less critical than digital signaling. To transmit digital signals on an analog network, each interface must contain electronics to convert the outgoing bit stream to an analog signal, and the incoming analog signal to a bit stream. Depending on the type of these electronics, 1 bps may occupy roughly 1 Hz of bandwidth. At higher frequencies, many bits per Hz are possible using advanced modulation techniques.

Broadband systems are divided up into multiple channels, frequently the 6-Mhz channels used for television broadcasting. Each channel can be used for analog television, CD-quality audio (1.4 Mbps), or a digital bit stream at, say, 3 Mbps, independent of the others. Television and data can be mixed on one cable.

One key difference between baseband and broadband is that broadband systems typically cover a large area and therefore need analog amplifiers to strengthen the signal periodically. These amplifiers can only transmit signals in one direction, so a computer outputting a packet will not be able to reach computers "upstream" from it if an amplifier lies between them. To get around this problem, two types of broadband systems have been developed: dual cable and single cable systems.

Dual cable systems have two identical cables running in parallel, next to each other. To transmit data, a computer outputs the data onto cable 1, which runs to a device called the **head-end** at the root of the cable tree. The head-end then transfers the signal to cable 2 for transmission back down the tree. All computers transmit on cable 1 and receive on cable 2. A dual cable system is shown in Fig. 2-4(a).

The other scheme allocates different frequency bands for inbound and outbound communication on a single cable [see Fig. 2-4(b)]. The low-frequency band is used for communication from the computers to the head-end, which then shifts the signal to the high-frequency band and rebroadcasts it. In the **subsplit** system, frequencies from 5 to 30 MHz are used for inbound traffic, and frequencies from 40 to 300 MHz are used for outbound traffic.

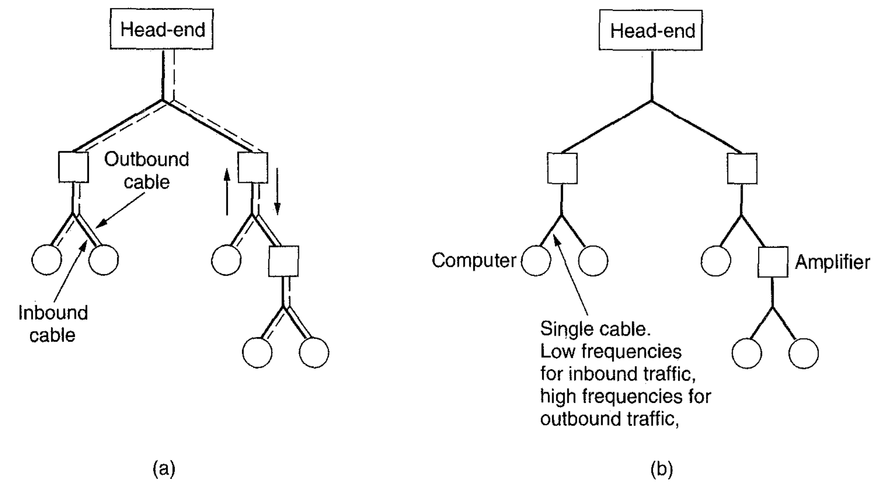

Fig. 2-4. Broadband networks. (a) Dual cable. (b) Single cable.

In the **midsplit** system, the inbound band is 5 to 116 MHz and the outbound band is 168 MHz to 300 MHz. The choice of these frequency bands is historical, having to do with how the U.S. Federal Communications Commission has assigned frequencies for television broadcasting, for which broadband was designed. Both split systems require an active head-end that accepts inbound signals on one band and rebroadcasts them on another. These techniques and frequencies were developed for cable television and have been taken over for networking without modification due to the availability of reliable and relatively inexpensive hardware.

Broadband can be used in various ways. Some computer pairs may be given a permanent channel for their exclusive use. Other computers may be able to request a channel for a temporary connection on a control channel, and then switch their frequencies to that channel for the duration of the connection. Still another arrangement is to have all the computers compete for access to a single channel or a group of channels, using techniques to be covered in Chap. 4.

Technically, broadband cable is inferior to baseband (i.e., single channel) cable for sending digital data but has the advantage that a huge amount of it is already in place. In the Netherlands, for example, 90 percent of all homes have a cable TV connection. In the United States, a TV cable runs past 80 percent of all homes. About 60 percent of these actually have a cable connection. With the competition between telephone companies and cable TV companies already in full swing, we can expect cable TV systems to begin operating as MANs and offering telephone and other services more and more often. For more information about using cable TV as a computer network, see (Karshmer and Thomas, 1992).

2.2.5. Fiber Optics

Many people in the computer industry take enormous pride in how fast computer technology is improving. In the 1970s, a fast computer (e.g., CDC 6600) could execute an instruction in 100 nsec. Twenty years later, a fast Cray computer could execute an instruction in 1 nsec, a factor of 10 improvement per decade. Not too bad.

In the same period, data communication went from 56 kbps (the ARPANET) to 1 Gbps (modern optical communication), a gain of more than a factor of 100 per decade, while at the same time the error rate went from 10^{-5} per bit to almost zero.

Furthermore, single CPUs are beginning to approach physical limits, such as speed of light and heat dissipation problems. In contrast, with *current* fiber technology, the achievable bandwidth is certainly in excess of 50,000 Gbps (50 Tbps) and many people are looking very hard for better materials. The current practical signaling limit of about 1 Gbps is due to our inability to convert between electrical and optical signals any faster. In the laboratory, 100 Gbps is feasible on short runs. A speed of 1 terabit/sec is only a few years down the road. Fully optical systems, including getting into and out of the computer, are within reach (Miki, 1994a).

In the race between computing and communication, communication won. The full implications of essentially infinite bandwidth (although not at zero cost) have not yet sunk in to a generation of computer scientists and engineers taught to think in terms of the low Nyquist and Shannon limits imposed by copper wire. The new conventional wisdom should be that all computers are hopelessly slow, and networks should try to avoid computation at all costs, no matter how much bandwidth that wastes. In this section we will study fiber optics to see how that transmission technology works.

An optical transmission system has three components: the light source, the transmission medium, and the detector. Conventionally, a pulse of light indicates a 1 bit and the absence of light indicates a zero bit. The transmission medium is an ultra-thin fiber of glass. The detector generates an electrical pulse when light falls on it. By attaching a light source to one end of an optical fiber and a detector to the other, we have a unidirectional data transmission system that accepts an electrical signal, converts and transmits it by light pulses, and then reconverts the output to an electrical signal at the receiving end.

This transmission system would leak light and be useless in practice except for an interesting principle of physics. When a light ray passes from one medium to another, for example, from fused silica to air, the ray is refracted (bent) at the silica/air boundary as shown in Fig. 2-5. Here we see a light ray incident on the boundary at an angle α_1 emerging at an angle β_1. The amount of refraction depends on the properties of the two media (in particular, their indices of refraction). For angles of incidence above a certain critical value, the light is refracted back into the silica; none of it escapes into the air. Thus a light ray incident at or above the critical angle is trapped inside the fiber, as shown in Fig. 2-5(b), and can propagate for many kilometers with virtually no loss.

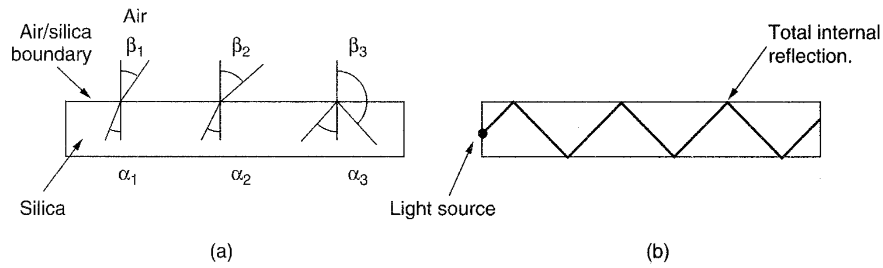

Fig. 2-5. (a) Three examples of a light ray from inside a silica fiber impinging on the air/silica boundary at different angles. (b) Light trapped by total internal reflection.

The sketch of Fig. 2-5(b) shows only one trapped ray, but since any light ray incident on the boundary above the critical angle will be reflected internally, many different rays will be bouncing around at different angles. Each ray is said to have a different **mode** so a fiber having this property is called a **multimode fiber**.

However, if the fiber's diameter is reduced to a few wavelengths of light, the fiber acts like a wave guide, and the light can only propagate in a straight line, without bouncing, yielding a **single-mode fiber**. Single mode fibers are more expensive but can be used for longer distances. Currently available single-mode fibers can transmit data at several Gbps for 30 km. Even higher data rates have been achieved in the laboratory for shorter distances. Experiments have shown that powerful lasers can drive a fiber 100 km long without repeaters, although at lower speeds. Research on erbium-doped fibers promises even longer runs without repeaters.

Transmission of Light through Fiber

Optical fibers are made of glass, which, in turn, is made from sand, an inexpensive raw material available in unlimited amounts. Glass making was known to the ancient Egyptians, but their glass had to be no more than 1 mm thick or the light could not shine through. Glass transparent enough to be useful for windows was developed during the Renaissance. The glass used for modern optical fibers is so transparent that if the oceans were full of it instead of water, the seabed would as visible from the surface as the ground is from an airplane on a clear day.

The attenuation of light through glass depends on the wavelength of the light.

For the kind of glass used in fibers, the attenuation is shown in Fig. 2-6 in decibels per linear kilometer of fiber. The attenuation in decibels is given by the formula

$$\text{Attenuation in decibels} = 10 \log_{10} \frac{\text{transmitted power}}{\text{received power}}$$

For example, a factor of two loss gives an attenuation of $10 \log_{10} 2 = 3$ dB. The figure shows the near infrared part of the spectrum, which is what is used in practice. Visible light has slightly shorter wavelengths, from 0.4 to 0.7 microns (1 micron is 10^{-6} meters).

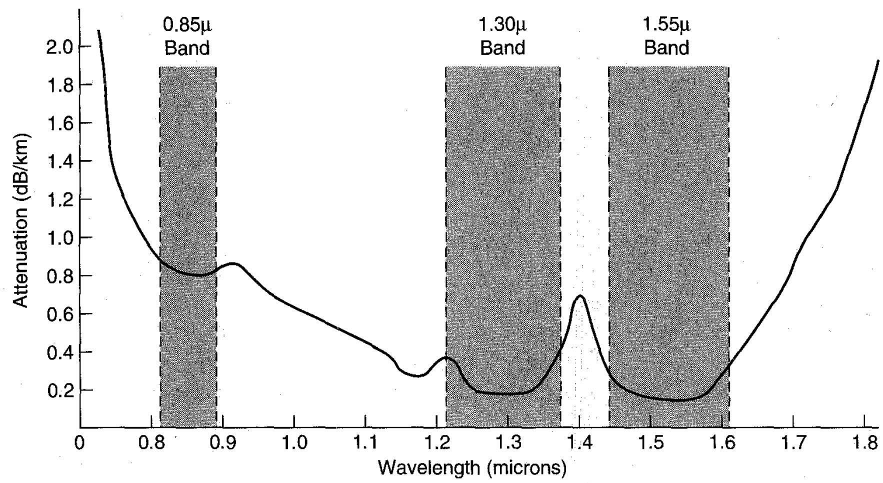

Fig. 2-6. Attenuation of light through fiber in the infrared region.

Three wavelength bands are used for communication. They are centered at 0.85, 1.30, and 1.55 microns, respectively. The latter two have good attenuation properties (less than 5 percent loss per kilometer). The 0.85 micron band has higher attenuation, but the nice property that at that wavelength, the lasers and electronics can be made from the same material (gallium arsenide). All three bands are 25,000 to 30,000 GHz wide.

Light pulses sent down a fiber spread out in length as they propagate. This spreading is called **dispersion**. The amount of it is wavelength dependent. One way to keep these spread-out pulses from overlapping is to increase the distance between them, but this can only be done by reducing the signaling rate. Fortunately, it has been discovered that by making the pulses in a special shape related to the reciprocal of the hyperbolic cosine, all the dispersion effects cancel out, and it may be possible to send pulses for thousands of kilometers without appreciable shape distortion. These pulses are called **solitons**. A considerable amount of research is going on to take solitons out of the lab and into the field.

Fiber Cables

Fiber optic cables are similar to coax, except without the braid. Figure 2-7(a) shows a single fiber viewed from the side. At the center is the glass core through which the light propagates. In multimode fibers, the core is 50 microns in diameter, about the thickness of a human hair. In single-mode fibers the core is 8 to 10 microns.

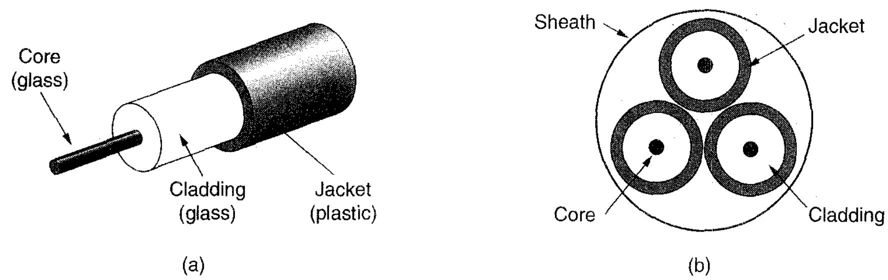

Fig. 2-7. (a) Side view of a single fiber. (b) End view of a sheath with three fibers.

The core is surrounded by a glass cladding with a lower index of refraction than the core, to keep all the light in the core. Next comes a thin plastic jacket to protect the cladding. Fibers are typically grouped together in bundles, protected by an outer sheath. Figure 2-7(b) shows a sheath with three fibers.

Terrestrial fiber sheaths are normally laid in the ground within a meter of the surface, where they are occasionally subject to attacks by backhoes or gophers. Near the shore, transoceanic fiber sheaths are buried in trenches by a kind of seaplow. In deep water, they just lie on the bottom, where they can be snagged by fishing trawlers or eaten by sharks.

Fibers can be connected in three different ways. First, they can terminate in connectors and be plugged into fiber sockets. Connectors lose about 10 to 20 percent of the light, but they make it easy to reconfigure systems.

Second, they can be spliced mechanically. Mechanical splices just lay the two carefully cut ends next to each other in a special sleeve and clamp them in place. Alignment can be improved by passing light through the junction and then making small adjustments to maximize the signal. Mechanical splices take trained personnel about 5 minutes, and result in a 10 percent light loss.

Third, two pieces of fiber can be fused (melted) to form a solid connection. A fusion splice is almost as good as a single drawn fiber, but even here, a small amount of attenuation occurs. For all three kinds of splices, reflections can occur at the point of the splice, and the reflected energy can interfere with the signal.

Two kinds of light sources can be used to do the signaling, LEDs (Light Emitting Diodes) and semiconductor lasers. They have different properties, as shown in Fig. 2-8. They can be tuned in wavelength by inserting Fabry-Perot or Mach-Zehnder interferometers between the source and the fiber. Fabry-Perot interferometers are simple resonant cavities consisting of two parallel mirrors. The light is incident perpendicularly to the mirrors. The length of the cavity selects out those wavelengths that fit inside an integral number of times. Mach-Zehnder interferometers separate the light into two beams. The two beams travel slightly different distances. They are recombined at the end and are in phase for only certain wavelengths.

Item	LED	Semiconductor laser
Data rate	Low	High
Mode	Multimode	Multimode or single mode
Distance	Short	Long
Lifetime	Long life	Short life
Temperature sensitivity	Minor	Substantial
Cost	Low cost	Expensive

Fig. 2-8. A comparison of semiconductor diodes and LEDs as light sources.

The receiving end of an optical fiber consists of a photodiode, which gives off an electrical pulse when struck by light. The typical response time of a photodiode is 1 nsec, which limits data rates to about 1 Gbps. Thermal noise is also an issue, so a pulse of light must carry enough energy to be detected. By making the pulses powerful enough, the error rate can be made arbitrarily small.

Fiber Optic Networks

Fiber optics can be used for LANs as well as for long-haul transmission, although tapping onto it is more complex than connecting to an Ethernet. One way around the problem is to realize that a ring network is really just a collection of point-to-point links, as shown in Fig. 2-9. The interface at each computer passes the light pulse stream through to the next link and also serves as a T junction to allow the computer to send and accept messages.

Two types of interfaces are used. A passive interface consists of two taps fused onto the main fiber. One tap has an LED or laser diode at the end of it (for transmitting), and the other has a photodiode (for receiving). The tap itself is completely passive and is thus extremely reliable because a broken LED or photodiode does not break the ring. It just takes one computer off-line.

The other interface type, shown in Fig. 2-9, is the **active repeater**. The incoming light is converted to an electrical signal, regenerated to full strength if it

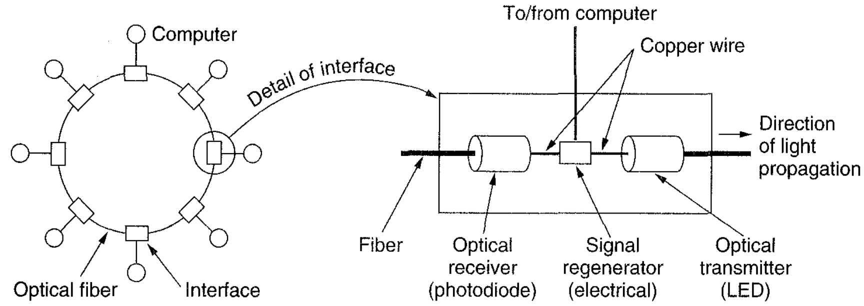

Fig. 2-9. A fiber optic ring with active repeaters.

has been weakened, and retransmitted as light. The interface with the computer is an ordinary copper wire that comes into the signal regenerator. Purely optical repeaters are now being used, too. These devices do not require the optical to electrical to optical conversions, which means they can operate at extremely high bandwidths.

If an active repeater fails, the ring is broken and the network goes down. On the other hand, since the signal is regenerated at each interface, the individual computer-to-computer links can be kilometers long, with virtually no limit on the total size of the ring. The passive interfaces lose light at each junction, so the number of computers and total ring length are greatly restricted.

A ring topology is not the only way to build a LAN using fiber optics. It is also possible to have hardware broadcasting using the **passive star** construction of Fig. 2-10. In this design, each interface has a fiber running from its transmitter to a silica cylinder, with the incoming fibers fused to one end of the cylinder. Similarly, fibers fused to the other end of the cylinder are run to each of the receivers. Whenever an interface emits a light pulse, it is diffused inside the passive star to illuminate all the receivers, thus achieving broadcast. In effect, the passive star combines all the incoming signals and transmits the merged result out on all lines. Since the incoming energy is divided among all the outgoing lines, the number of nodes in the network is limited by the sensitivity of the photodiodes.

Comparison of Fiber Optics and Copper Wire

It is instructive to compare fiber to copper. Fiber has many advantages. To start with, it can handle much higher bandwidths than copper. This alone would require its use in high-end networks. Due to the low attenuation, repeaters are needed only about every 30 km on long lines, versus about every 5 km for copper, a substantial cost saving. Fiber also has the advantage of not being affected by

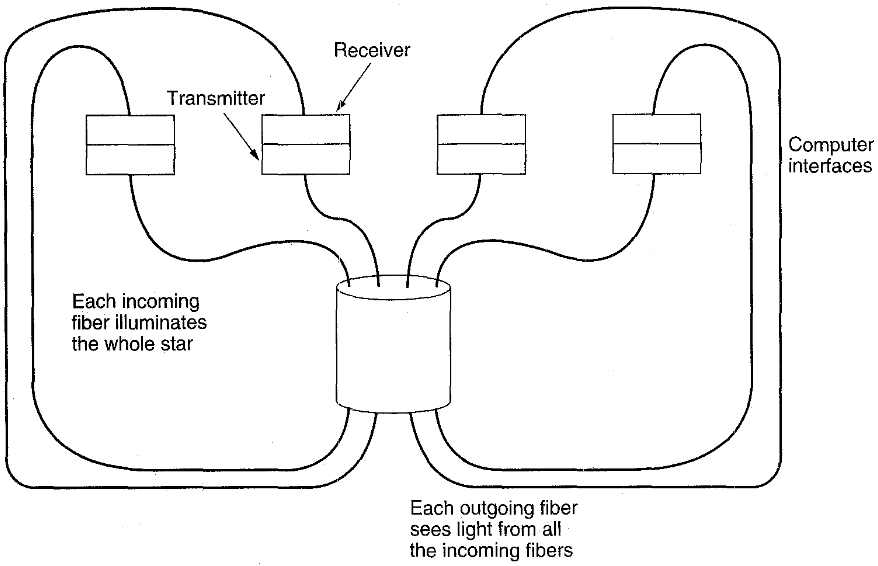

Fig. 2-10. A passive star connection in a fiber optics network.

power surges, electromagnetic interference, or power failures. Nor is it affected by corrosive chemicals in the air, making it ideal for harsh factory environments.

Oddly enough, telephone companies like fiber for a different reason: it is thin and lightweight. Many existing cable ducts are completely full, so there is no room to add new capacity. Removing all the copper and replacing it by fibers empties up the ducts, and the copper has excellent resale value to copper refiners who see it as very high grade ore. Also fiber is lighter than copper. One thousand twisted pairs 1 km long weigh 8000 kg. Two fibers have more capacity and weigh only 100 kg, which greatly reduces the need for expensive mechanical support systems that must be maintained. For new routes, fiber wins hands down due to its much lower installation cost.

Finally, fibers do not leak light and are quite difficult to tap. This gives them excellent security against potential wiretappers.

The reason that fiber is better than copper is inherent in the underlying physics. When electrons move in a wire, they affect one another and are themselves affected by electrons outside the wire. Photons in a fiber do not affect one another (they have no electric charge) and are not affected by stray photons outside the fiber.

On the downside, fiber is an unfamiliar technology requiring skills most engineers do not have. Since optical transmission is inherently unidirectional, two-way communication requires either two fibers or two frequency bands on one

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fiber. Finally, fiber interfaces cost more than electrical interfaces. Nevertheless, the future of all fixed data communication for distances of more than a few meters is clearly with fiber. For a detailed discussion of all aspects of fiber optic networks, see (Green, 1993).

2.3. WIRELESS TRANSMISSION

Our age has given rise to information junkies: people who need to be on-line all the time. For these mobile users, twisted pair, coax, and fiber optics are of no use. They need to get their hits of data for their laptop, notebook, shirt pocket, palmtop, or wristwatch computers without being tethered to the terrestrial communication infrastructure. For these users, wireless communication is the answer. In this section we will look at wireless communication in general, as it has many other important applications besides providing connectivity to users who want to read their email in airplanes.

Some people even believe that the future holds only two kinds of communication: fiber and wireless. All fixed (i.e., nonmobile) computers, telephones, faxes, and so on will be by fiber, and all mobile ones will use wireless.

However wireless also has advantages for even fixed devices in some circumstances. For example, if running a fiber to a building is difficult due to the terrain (mountains, jungles, swamps, etc.) wireless may be preferable. It is noteworthy that modern wireless digital communication began in the Hawaiian Islands, where large chunks of Pacific Ocean separated the users and the telephone system was inadequate.

2.3.1. The Electromagnetic Spectrum

When electrons move, they create electromagnetic waves that can propagate through free space (even in a vacuum). These waves were predicted by the British physicist James Clerk Maxwell in 1865 and first produced and observed by the German physicist Heinrich Hertz in 1887. The number of oscillations per second of an electromagnetic wave is called its **frequency**, f, and is measured in **Hz** (in honor of Heinrich Hertz). The distance between two consecutive maxima (or minima) is called the **wavelength**, which is universally designated by the Greek letter λ (lambda).

By attaching an antenna of the appropriate size to an electrical circuit, the electromagnetic waves can be broadcast efficiently and received by a receiver some distance away. All wireless communication is based on this principle.

In vacuum, all electromagnetic waves travel at the same speed, no matter what their frequency. This speed, usually called the **speed of light**, c, is approximately 3×10^8 m/sec, or about 1 foot (30 cm) per nanosecond. In copper or fiber the speed slows to about 2/3 of this value and becomes slightly frequency dependent. The speed of light is the ultimate speed limit. No object or signal can ever move faster than it.

The fundamental relation between f, λ, and c (in vacuum) is

$$\lambda f = c \tag{2-2}$$

Since c is a constant, if we know f we can find λ and vice versa. For example, 1-MHz waves are about 300 meters long and 1-cm waves have a frequency of 30 GHz.

The electromagnetic spectrum is shown in Fig. 2-11. The radio, microwave, infrared, and visible light portions of the spectrum can all be used for transmitting information by modulating the amplitude, frequency, or phase of the waves. Ultraviolet light, X-rays, and gamma rays would be even better, due to their higher frequencies, but they are hard to produce and modulate, do not propagate well through buildings, and are dangerous to living things. The bands listed at the bottom of Fig. 2-11 are the official ITU names and are based on the wavelengths, so the LF band goes from 1 km to 10 km (approximately 30 kHz to 300 kHz). The terms LF, MF, and HF refer to low, medium, and high frequency, respectively. Clearly, when the names were assigned, nobody expected to go above 10 MHz, so the higher bands were later named the Very, Ultra, Super, Extremely, and Tremendously High Frequency bands. Beyond that there are no names, but Incredibly, Astonishingly, and Prodigiously high frequency (IHF, AHF, and PHF) would sound nice.

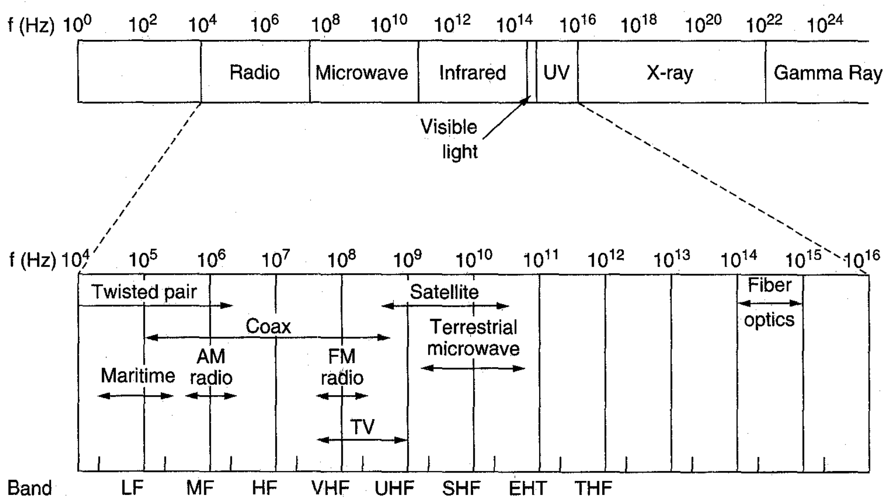

Fig. 2-11. The electromagnetic spectrum and its uses for communication.

The amount of information that an electromagnetic wave can carry is related to its bandwidth. With current technology, it is possible to encode a few bits per Hertz at low frequencies, but often as many as 40 under certain conditions at high frequencies, so a cable with a 500 MHz bandwidth can carry several gigabits/sec. From Fig. 2-11 it should now be obvious why networking people like fiber optics so much.

If we solve Eq. (2-2) for f and differentiate with respect to λ we get

$$\frac{df}{d\lambda} = -\frac{c}{\lambda^2}$$

If we now go to finite differences instead of differentials and only look at absolute values, we get

$$\Delta f = \frac{c\,\Delta\lambda}{\lambda^2} \tag{2-3}$$

Thus given the width of a wavelength band, $\Delta\lambda$, we can compute the corresponding frequency band, Δf, and from that the data rate the band can produce. The wider the band, the higher the data rate. As an example, consider the 1.30-micron band of Fig. 2-6. Here we have $\lambda = 1.3 \times 10^{-6}$ and $\Delta\lambda = 0.17 \times 10^{-6}$, so Δf is about 30 THz.

To prevent total chaos, there are national and international agreements about who gets to use which frequencies. Since everyone wants a higher data rate, everyone wants more spectrum. In the United States, the FCC allocates spectrum for AM and FM radio, television, and cellular phones, as well as for telephone companies, police, maritime, navigation, military, government, and many other competing users. Worldwide, an agency of ITU-R (WARC) does this work. In the meeting in Spain in 1991, for example, WARC allocated some spectrum to hand-held personal communicators. Unfortunately, the FCC, which is not bound by WARC's recommendations, chose a different piece (because the people in the United States who had the band WARC chose did not want to give it up and had enough political clout to prevent that). Consequently, personal communicators built for the U.S. market will not work in Europe or Asia, and vice versa.

Most transmissions use a narrow frequency band (i.e., $\Delta f / f \ll 1$) to get the best reception (many watts/Hz). However, in some cases, the transmitter hops from frequency to frequency in a regular pattern or the transmissions are intentionally spread out over a wide frequency band. This technique is called **spread spectrum** (Kohno et al., 1995). It is popular for military communication because it makes transmissions hard to detect and next to impossible to jam. Frequency hopping is not of much interest to us (other than to note that it was co-invented by the movie actress Hedy Lamarr). True spread spectrum, sometimes called **direct sequence spread spectrum**, is gaining popularity in the commercial world, and we will come back to it in Chap. 4. For a fascinating and detailed history of spread spectrum communication, see (Scholtz, 1982).

For the moment, we will assume that all transmissions use a narrow frequency band. We will now discuss how the various parts of the spectrum are used, starting with radio.

2.3.2. Radio Transmission

Radio waves are easy to generate, can travel long distances, and penetrate buildings easily, so they are widely used for communication, both indoors and outdoors. Radio waves also are omnidirectional, meaning that they travel in all directions from the source, so that the transmitter and receiver do not have to be carefully aligned physically.

Sometimes omnidirectional radio is good, but sometimes it is bad. In the 1970s, General Motors decided to equip its new Cadillacs with computer-controlled antilock brakes. When the driver stepped on the brake pedal, the computer pulsed the brakes on and off instead of locking them on hard. One fine day an Ohio Highway Patrolman began using his new mobile radio to call headquarters, and suddenly the Cadillac next to him began behaving like a bucking bronco. When the officer pulled the car over, the driver claimed that he had done nothing and that the car had gone crazy.

Eventually, a pattern began to emerge: Cadillacs would sometimes go berserk, but only on major highways in Ohio and then only when the Highway Patrol was watching. For a long, long time General Motors could not understand why Cadillacs worked fine in all the other states, and also on minor roads in Ohio. Only after a considerable amount of searching did they discover that the Cadillac's wiring made a fine antenna for the frequency the Ohio Highway Patrol's new radio system used.

The properties of radio waves are frequency dependent. At low frequencies, radio waves pass through obstacles well, but the power falls off sharply with distance from the source, roughly as $1/r^3$ in air. At high frequencies, radio waves tend to travel in straight lines and bounce off obstacles. They are also absorbed by rain. At all frequencies, radio waves are subject to interference from motors and other electrical equipment.

Due to radio's ability to travel long distances, interference between users is a problem. For this reason, all governments tightly license the user of radio transmitters, with one exception (discussed below).

In the VLF, LF, and MF bands, radio waves follow the ground, as illustrated in Fig. 2-12(a). These waves can be detected for perhaps 1000 km at the lower frequencies, less at the higher ones. AM radio broadcasting uses the MF band, which is why Boston AM radio stations cannot be heard easily in New York. Radio waves in these bands easily pass through buildings, which is why portable radios work indoors. The main problem with using these bands for data communication is the relative low bandwidth they offer [see Eq. (2-2)].

In the HF and VHF bands, the ground waves tend to be absorbed by the earth.

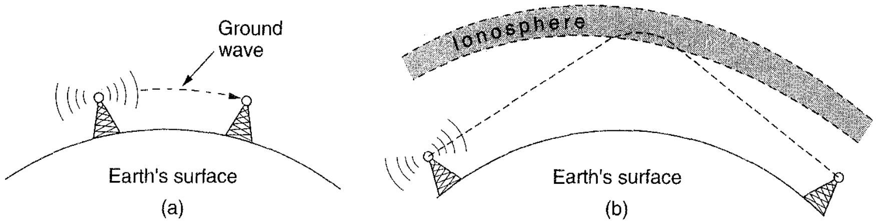

Fig. 2-12. (a) In the VLF, VF, and MF bands, radio waves follow the curvature of the earth. (b) In the HF they bounce off the ionosphere.

However, the waves that reach the ionosphere, a layer of charged particles circling the earth at a height of 100 to 500 km, are refracted by it and sent back to earth, as shown in Fig. 2-12(b). Under certain atmospheric conditions, the signals may bounce several times. Amateur radio operators (hams) use these bands to talk long distance. The military also communicates in the HF and VHF bands.

2.3.3. Microwave Transmission

Above 100 MHz, the waves travel in straight lines and can therefore be narrowly focused. Concentrating all the energy into a small beam using a parabolic antenna (like the familiar satellite TV dish) gives a much higher signal to noise ratio, but the transmitting and receiving antennas must be accurately aligned with each other. In addition, this directionality allows multiple transmitters lined up in a row to communicate with multiple receivers in a row without interference. Before fiber optics, for decades these microwaves formed the heart of the long-distance telephone transmission system. In fact, the long-distance carrier MCI's name first stood for Microwave Communications, Inc., because its entire system was originally built on microwave towers (it has since upgraded major portions of its network to fiber).

Since the microwaves travel in a straight line, if the towers are too far apart, the earth will get in the way (think about a San Francisco to Amsterdam link). Consequently, repeaters are needed periodically. The higher the towers are, the further apart they can be. The distance between repeaters goes up very roughly with the square root of the tower height. For 100-m high towers, repeaters can be spaced 80 km apart.

Unlike radio waves at lower frequencies, microwaves do not pass through buildings well. In addition, even though the beam may be well focused at the transmitter, there is still some divergence in space. Some waves may be refracted off low-lying atmospheric layers and may take slightly longer to arrive than direct waves. The delayed waves may arrive out of phase with the direct wave and thus cancel the signal. This effect is called **multipath fading** and is often a serious problem. It is weather and frequency dependent. Some operators keep 10 percent of their channels idle as spares to switch on when multipath fading wipes out some frequency band temporarily.

The demand for more and more spectrum works to keep improving the technology so transmissions can use still higher frequencies. Bands up to 10 GHz are now in routine use, but at about 8 GHz a new problem sets in: absorption by water. These waves are only a few centimeters long and are absorbed by rain. This effect would be fine if one were planning to build a huge outdoor microwave oven, but for communication, it is a severe problem. As with multipath fading, the only solution is to shut off links that are being rained on and route around them.

In summary, microwave communication is so widely used for long-distance telephone communication, cellular telephones, television distribution, and other uses, that a severe shortage of spectrum has developed. It has several significant advantages over fiber. The main one is that no right of way is needed, and by buying a small plot of ground every 50 km and putting a microwave tower on it, one can bypass the telephone system and communicate directly. This is how MCI managed to get started as a new long-distance telephone company so quickly. (Sprint went a different route: it was formed by the Southern Pacific Railroad, which already owned a large amount of right of way, and just buried fiber next to the tracks.)

Microwave is also relatively inexpensive. Putting up two simple towers (maybe just big poles with four guy wires) and putting antennas on each one may be cheaper than burying 50 km of fiber through a congested urban area or up over a mountain, and it may also be cheaper than leasing the telephone company's fiber, especially if the telephone company has not yet even fully paid for the copper it ripped out when it put in the fiber.

In addition to being used for long-distance transmission, microwaves have another important use, namely, the **Industrial/Scientific/Medical** bands. These bands form the one exception to the licensing rule: transmitters using these bands do not require government licensing. One band is allocated worldwide: 2.400–2.484 GHz. In addition, in the United States and Canada, bands also exist from 902–928 MHz and from 5.725–5.850 GHz. These bands are used for cordless telephones, garage door openers, wireless hi-fi speakers, security gates, etc. The 900-MHz band works best but is crowded and equipment using it may only be operated in North America. The higher bands require more expensive electronics and are subject to interference from microwave ovens and radar installations. Nevertheless, these bands are popular for various forms of short-range wireless networking because they avoid the problems associated with licensing.

2.3.4. Infrared and Millimeter Waves

Unguided infrared and millimeter waves are widely used for short-range communication. The remote controls used on televisions, VCRs, and stereos all use infrared communication. They are relatively directional, cheap, and easy to build, but have a major drawback: they do not pass through solid objects (try standing between your remote control and your television and see if it still works). In general, as we go from long-wave radio toward visible light, the waves behave more and more like light and less and less like radio.

On the other hand, the fact that infrared waves do not pass through solid walls well is also a plus. It means that an infrared system in one room of a building will not interfere with a similar system in adjacent rooms. Furthermore, security of infrared systems against eavesdropping is better than that of radio systems precisely for this reason. For these reasons, no government license is needed to operate an infrared system, in contrast to radio systems, which must be licensed.

These properties have made infrared an interesting candidate for indoor wireless LANs. For example, the computers and offices in a building can be equipped with relatively unfocused (i.e., somewhat omnidirectional) infrared transmitters and receivers. In this way, portable computers with infrared capability can be on the local LAN without having to physically connect to it. When several people show up for a meeting with their portables, they can just sit down in the conference room and be fully connected, without having to plug in. Infrared communication cannot be used outdoors because the sun shines as brightly in the infrared as in the visible spectrum. For more information about infrared communication, see (Adams et al., 1993; and Bantz and Bauchot, 1994).

2.3.5. Lightwave Transmission

Unguided optical signaling has been in use for centuries. Paul Revere used binary optical signaling from the Old North Church just prior to his famous ride. A more modern application is to connect the LANs in two buildings via lasers mounted on their rooftops. Coherent optical signaling using lasers is inherently unidirectional, so each building needs its own laser and its own photodetector. This scheme offers very high bandwidth and very low cost. It is also relatively easy to install, and, unlike microwave, does not require an FCC license.

The laser's strength, a very narrow beam, is also its weakness here. Aiming a laser beam 1 mm wide at a target 1 mm wide 500 meters away requires the marksmanship of a latter-day Annie Oakley. Usually, lenses are put into the system to defocus the beam slightly.

A disadvantage is that laser beams cannot penetrate rain or thick fog, but they normally work well on sunny days. However, the author once attended a conference at a modern hotel in Europe at which the conference organizers thoughtfully provided a room full of terminals for the attendees to read their email during boring presentations. Since the local PTT was unwilling to install a large number of telephone lines for just 3 days, the organizers put a laser on the roof and aimed it at their university's computer science building a few kilometers away. They tested it the night before the conference and it worked perfectly. At 9 a.m. the next morning, on a bright sunny day, the link failed completely and stayed down all day. That evening, the organizers tested it again very carefully, and once again it worked absolutely perfectly. The pattern repeated itself for two more days consistently.

After the conference, the organizers discovered the problem. Heat from the sun during the daytime caused convection currents to rise up from the roof of the building, as shown in Fig. 2-13. This turbulent air diverted the beam and made it dance around the detector. Atmospheric "seeing" like this makes the stars twinkle (which is why astronomers put their telescopes on the tops of mountains—to get above as much of the atmosphere as possible). It is also responsible for shimmering roads on a hot day and the wavy images when looking out above a hot radiator.

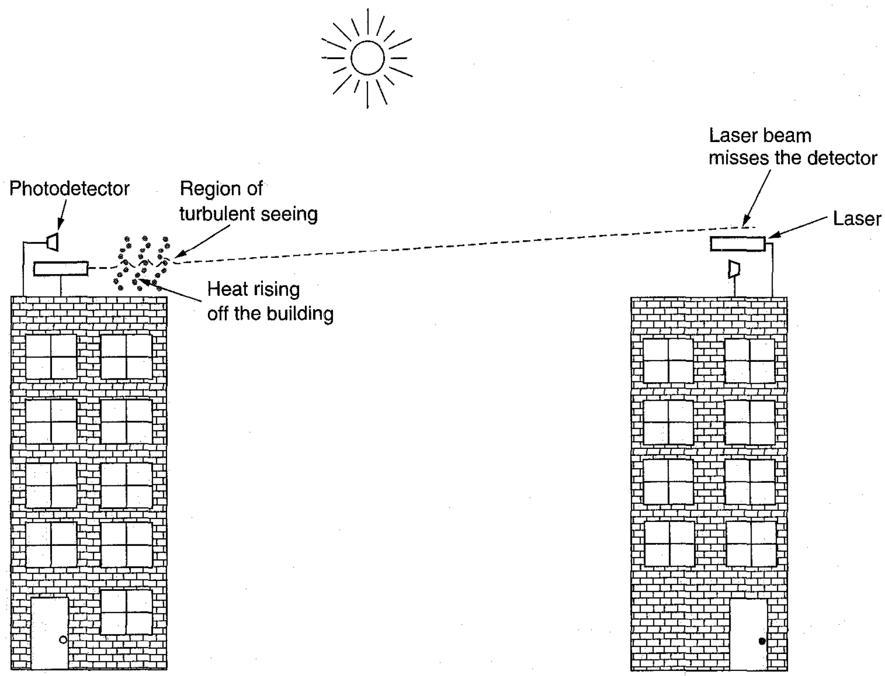

Fig. 2-13. Convection currents can interfere with laser communication systems. A bidirectional system, with two lasers, is pictured here.

2.4. THE TELEPHONE SYSTEM

When two computers owned by the same company or organization and located close to each other need to communicate, it is often easiest just to run a cable between them. LANs work this way. However, when the distances are large, or there are many computers, or the cables would have to pass through a public road or other public right of way, the costs of running private cables are usually prohibitive. Furthermore, in just about every country in the world, stringing private transmission lines across (or underneath) public property is also illegal. Consequently, the network designers must rely upon the existing telecommunication facilities.

These facilities, especially the **PSTN**, (**Public Switched Telephone Network**), were usually designed many years ago, with a completely different goal in mind: transmitting the human voice in a more or less recognizable form. Their suitability for use in computer-computer communication is often marginal at best, but the situation is rapidly changing with the introduction of fiber optics and digital technology. In any event, the telephone system is so tightly intertwined with (wide area) computer networks, that it is worth devoting considerable time studying it.

To see the order of magnitude of the problem, let us make a rough but illustrative comparison of the properties of a typical computer-computer connection via a local cable and via a dial-up telephone line. A cable running between two computers can transfer data at memory speeds, typically 10^7 to 10^8 bps. The error rate is usually so low that it is hard to measure, but one error per day would be considered poor at most installations. One error per day at these speeds is equivalent to one error per 10^{12} or 10^{13} bits sent.

In contrast, a dial-up line has a maximum data rate on the order of 10^4 bps and an error rate of roughly 1 per 10^5 bits sent, varying somewhat with the age of the telephone switching equipment involved. The combined bit rate times error rate performance of a local cable is thus 11 orders of magnitude better than a voice-grade telephone line. To make an analogy in the field of transportation, the ratio of the cost of the entire Apollo project, which landed men on the moon, to the cost of a bus ride downtown is about 11 orders of magnitude (in 1965 dollars: 40 billion to 0.40).

The trouble, of course, is that computer systems designers are used to working with computer systems, and when suddenly confronted with another system whose performance (from their point of view) is 11 orders of magnitude worse, it is not surprising that much time and effort have been devoted to trying to figure out how to use it efficiently. On the other hand, the telephone companies have made massive strides in the past decade in upgrading equipment and improving service in certain areas. In the following sections we will describe the telephone system and show what it used to be and where it is going. For additional information about the innards of the telephone system see (Bellamy, 1991).

2.4.1. Structure of the Telephone System

When Alexander Graham Bell patented the telephone in 1876 (just a few hours ahead of his rival, Elisha Gray), there was an enormous demand for his new invention. The initial market was for the sale of telephones, which came in pairs. It was up to the customer to string a single wire between them. The electrons returned through the earth. If a telephone owner wanted to talk to n other telephone owners, separate wires had to be strung to all n houses. Within a year, the cities were covered with wires passing over houses and trees in a wild jumble. It became immediately obvious that the model of connecting every telephone to every other telephone, as shown in Fig. 2-14(a) was not going to work.

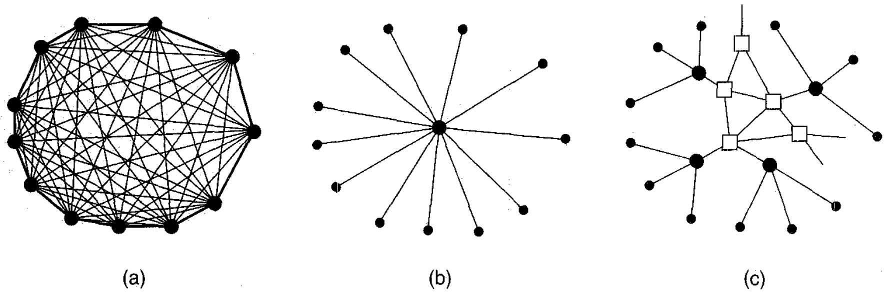

Fig. 2-14. (a) Fully interconnected network. (b) Centralized switch. (c) Two-level hierarchy.

To his credit, Bell saw this and formed the Bell Telephone Company, which opened its first switching office (in New Haven, Connecticut) in 1878. The company ran a wire to each customer's house or office. To make a call, the customer would crank the phone to make a ringing sound in the telephone company office to attract the attention of an operator, who would then manually connect the caller to the callee using a jumper cable. The model of a single switching office is illustrated in Fig. 2-14(b).

Pretty soon, Bell System switching offices were springing up everywhere and people wanted to make long-distance calls between cities, so the Bell system began to connect the switching offices. The original problem soon returned: to connect every switching office to every other switching office by means of a wire between them quickly became unmanageable, so second-level switching offices were invented, After a while, multiple second-level offices were needed, as shown in Fig. 2-14(c). Eventually, the hierarchy grew to five levels.

By 1890, the three major parts of the telephone system were in place: the switching offices, the wires between the customers and the switching offices (by now balanced, insulated, twisted pairs instead of open wires with an earth return), and the long-distance connections between the switching offices. While there have been improvements in all three areas since then, the basic Bell System model has remained essentially intact for over 100 years. For a short technical history of the telephone system, see (Hawley, 1991).

At present, the telephone system is organized as a highly redundant, multilevel hierarchy. The following description is highly simplified but gives the essential flavor nevertheless. Each telephone has two copper wires coming out of it that go directly to the telephone company's nearest **end office** (also called a **local central office**). The distance is typically 1 to 10 km, being smaller in cities than in rural areas.

In the United States alone there are about 19,000 end offices. The concatenation of the area code and the first three digits of the telephone number uniquely specify an end office, which is why the rate structure uses this information. The two-wire connections between each subscriber's telephone and the end office are known in the trade as the **local loop**. If the world's local loops were stretched out end to end, they would extend to the moon and back 1000 times.

At one time, 80 percent of AT&T's capital value was the copper in the local loops. AT&T was then, in effect, the world's largest copper mine. Fortunately, this fact was not widely known in the investment community. Had it been known, some corporate raider might have bought AT&T, terminated all telephone service in the United States, ripped out all the wire, and sold the wire to a copper refiner to get a quick payback.

If a subscriber attached to a given end office calls another subscriber attached to the same end office, the switching mechanism within the office sets up a direct electrical connection between the two local loops. This connection remains intact for the duration of the call.

If the called telephone is attached to another end office, a different procedure has to be used. Each end office has a number of outgoing lines to one or more nearby switching centers, called **toll offices** (or if they are within the same local area, **tandem offices**). These lines are called **toll connecting trunks**. If both the caller's and callee's end offices happen to have a toll connecting trunk to the same toll office (a likely occurrence if they are relatively close by), the connection may be established within the toll office. A telephone network consisting only of telephones (the small dots), end offices (the large dots) and toll offices (the squares) is shown in Fig. 2-14(c).

If the caller and callee do not have a toll office in common, the path will have to be established somewhere higher up in the hierarchy. There are primary, sectional, and regional offices that form a network by which the toll offices are connected. The toll, primary, sectional, and regional exchanges communicate with each other via high bandwidth **intertoll trunks** (also called **interoffice trunks**). The number of different kinds of switching centers and their topology (e.g., may two sectional offices have a direct connection or must they go through a regional office?) varies from country to country depending on its telephone density. Figure 2-15 shows how a medium-distance connection might be routed.

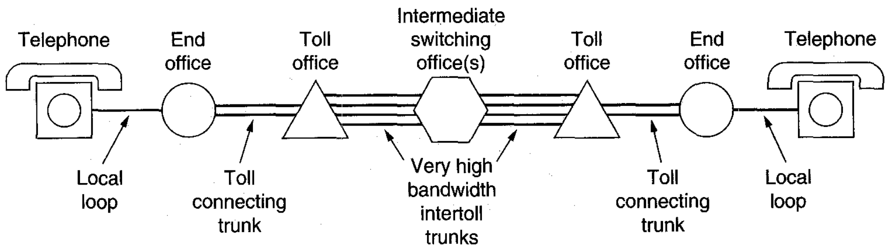

Fig. 2-15. Typical circuit route for a medium-distance call.

A variety of transmission media are used for telecommunication. Local loops consist of twisted pairs nowadays, although in the early days of telephony, uninsulated wires spaced 25 cm apart on telephone poles were common. Between switching offices, coaxial cables, microwaves, and especially fiber optics are widely used.

In the past, signaling throughout the telephone system was analog, with the actual voice signal being transmitted as an electrical voltage from source to destination. With the advent of digital electronics and computers, digital signaling has become possible. In this system, only two voltages are allowed, for example −5 volts and +5 volts.

This scheme has a number of advantages over analog signaling. First is that although the attenuation and distortion are more severe when sending two-level signals than when using modems, it is easy to calculate how far a signal can propagate and still be recognizable. A digital regenerator can be inserted into the line there, to restore the signal to its original value, since there are only two possibilities. A digital signal can pass through an arbitrary number of regenerators with no loss in signal and thus travel long distances with no information loss. In contrast, analog signals always suffer some information loss when amplified, and this loss is cumulative. The net result is that digital transmission can be made to have a low error rate.

A second advantage of digital transmission is that voice, data, music, and images (e.g., television, fax, and video) can be interspersed to make more efficient use of the circuits and equipment. Another advantage is that much higher data rates are possible using existing lines.

A third advantage is that digital transmission is much cheaper than analog transmission, since it is not necessary to accurately reproduce an analog waveform after it has passed through potentially hundreds of amplifiers on a transcontinental call. Being able to correctly distinguish a 0 from a 1 is enough.

Finally, maintenance of a digital system is easier than maintenance of an analog one. A transmitted bit is either received correctly or not, making it simpler to track down problems.

Consequently, all the long-distance trunks within the telephone system are rapidly being converted to digital. The old system used analog transmission over copper wires; the new one uses digital transmission over optical fibers.

In summary, the telephone system consists of three major components:

1. Local loops (twisted pairs, analog signaling).
2. Trunks (fiber optics or microwave, mostly digital).
3. Switching offices.

After a short digression on the politics of telephones, we will come back to each of these three components in some detail. For the local loop, we will be concerned with how to send digital data over it (quick answer: use a modem). For the long-haul trunks, the main issue is how to collect multiple calls together and send them together. This subject is called multiplexing, and we will study three different ways to do it. Finally, there are two fundamentally different ways of doing switching, so we will look at both of these.

2.4.2. The Politics of Telephones

For decades prior to 1984, the Bell System provided both local and long distance service throughout most of the United States. In the 1970s, the U.S. government came to believe that this was an illegal monopoly and sued to break it up. The government won, and on Jan. 1, 1984, AT&T was broken up into AT&T Long Lines, 23 **BOC**s (**Bell Operating Companies**), and a few other pieces. The 23 BOCs were grouped together into seven regional BOCs (RBOCs) to make them economically viable. The entire nature of telecommunication in the United States was changed overnight by court order (*not* by an act of Congress).

The exact details of the divestiture were described in the so-called **MFJ** (**Modified Final Judgment**), an oxymoron if ever there was one (if the judgment could be modified, it clearly was not final). This event led to increased competition, better service, and lower prices to consumers and businesses. Many other countries are now considering introducing competition along similar lines.

To make it clear who could do what, the United States was divided up into about 160 **LATA**s (**Local Access and Transport Areas**). Very roughly, a LATA is about as big as the area covered by one area code. Within a LATA, there is normally one **LEC** (**Local Exchange Carrier**) that has a monopoly on traditional telephone service within the LATA. The most important LECs are the BOCs, although some LATAs contain one or more of the 1500 independent telephone companies operating as LECs. In geographically large LATAs (mostly in the West), the LEC may handle long distance calls within its own LATA but may not handle calls going to a different LATA.

All inter-LATA traffic is handled by a different kind of company, an **IXC** (**IntereXchange Carrier**). Originally, AT&T Long Lines was the only serious IXC, but now MCI and Sprint are well-established competitors in the IXC

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business. One of the concerns at the breakup was to ensure that all the IXCs would be treated equally in terms of line quality, tariffs, and the number of digits their customers would have to dial to use them. The way this is handled is illustrated in Fig. 2-16. Here we see three example LATAs, each with several end offices. LATAs 2 and 3 also have a small hierarchy with tandem offices (intra-LATA toll offices).

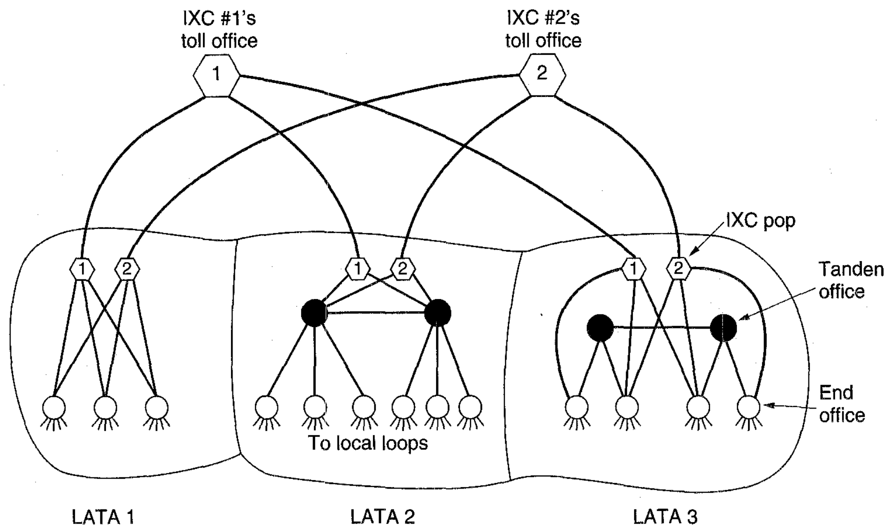

Fig. 2-16. The relationship of LATAs, LECs, and IXCs. All the circles are LEC switching offices. Each hexagon belongs to the IXC whose number is in it.

Any IXC that wishes to handle calls originating in a LATA can build a switching office called a **POP** (**Point of Presence**) there. The LEC is required to connect each IXC to every end office, either directly, as in LATAs 1 and 3, or indirectly, as in LATA 2. Furthermore, the terms of the connection, both technical and financial, must be identical for all IXCs. In this way, a subscriber in, say, LATA 1, can choose which IXC to use for calling subscribers in LATA 3.

As part of the MFJ, the IXCs were forbidden to offer local telephone service and the LECs were forbidden to offer inter-LATA telephone service, although both were free to enter other businesses, such as operating fried chicken restaurants. In 1984, that was a fairly unambiguous statement. Unfortunately, technology has a way of making the law obsolete. Neither cable television nor cellular phones were covered by the agreement. As cable television went from one way to two way, and cellular phones exploded in popularity, both LECs and IXCs began buying up or merging with cable and cellular operators.

By 1995, Congress saw that trying to maintain a distinction between the various kinds of companies was no longer tenable and drafted a bill to allow cable TV companies, local telephone companies, long distance carriers, and cellular operators to enter one another's businesses. The idea was that any company could then offer its customers a single integrated package containing cable TV, telephone, and information services, and that different companies would compete on service and price. The bill was enacted into law in February 1996. As a result, the U.S. telecommunications landscape is currently undergoing a radical restructuring.

2.4.3. The Local Loop

For the past 100 years, analog transmission has dominated all communication. In particular, the telephone system was originally based entirely on analog signaling. While the long-distance trunks are now largely digital in the more advanced countries, the local loops are still analog and are likely to remain so for at least a decade or two, due to the enormous cost of converting them. Consequently, when a computer wishes to send digital data over a dial-up line, the data must first be converted to analog form by a modem for transmission over the local loop, then converted to digital form for transmission over the long-haul trunks, then back to analog over the local loop at the receiving end, and finally back to digital by another modem for storage in the destination computer. This arrangement is shown in Fig. 2-17.

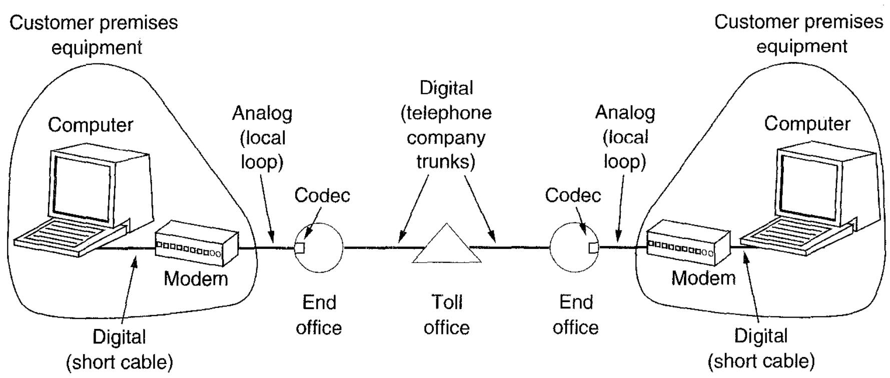

Fig. 2-17. The use of both analog and digital transmission for a computer to computer call. Conversion is done by the modems and codecs.

While this situation is not exactly ideal, such is life for the time being, and students of networking should have some understanding of both analog and digital transmission, as well as how the conversions back and forth work. For leased lines it is possible to go digital from start to finish, but these are expensive and are only useful for building intracompany private networks.

In the following sections we will look briefly at what is wrong with analog transmission and examine how modems make it possible to transmit digital data over analog circuits. We will also look at two common modem interfaces, RS-232-C and RS-449.

Transmission Impairments

Analog signaling consists of varying a voltage with time to represent an information stream. If transmission media were perfect, the receiver would receive exactly the same signal that the transmitter sent. Unfortunately, media are not perfect, so the received signal is not the same as the transmitted signal. For digital data, this difference can lead to errors.

Transmission lines suffer from three major problems: attenuation, delay distortion, and noise. **Attenuation** is the loss of energy as the signal propagates outward. On guided media (e.g., wires and optical fibers), the signal falls off logarithmically with the distance. The loss is expressed in decibels per kilometer. The amount of energy lost depends on the frequency. To see the effect of this frequency dependence, imagine a signal not as a simple waveform, but as a series of Fourier components. Each component is attenuated by a different amount, which results in a different Fourier spectrum at the receiver, and hence a different signal.

If the attenuation is too much, the receiver may not be able to detect the signal at all, or the signal may fall below the noise level. In many cases, the attenuation properties of a medium are known, so amplifiers can be put in to try to compensate for the frequency-dependent attenuation. The approach helps but can never restore the signal exactly back to its original shape.

The second transmission impairment is **delay distortion**. It is caused by the fact that different Fourier components travel at different speeds. For digital data, fast components from one bit may catch up and overtake slow components from the bit ahead, mixing the two bits and increasing the probability of incorrect reception.

The third impairment is **noise**, which is unwanted energy from sources other than the transmitter. Thermal noise is caused by the random motion of the electrons in a wire and is unavoidable. Cross talk is caused by inductive coupling between two wires that are close to each other. Sometimes when talking on the telephone, you can hear another conversation in the background. That is cross talk. Finally, there is impulse noise, caused by spikes on the power line or other causes. For digital data, impulse noise can wipe out one or more bits.

Modems

Due to the problems just discussed, especially the fact that both attenuation and propagation speed are frequency dependent, it is undesirable to have a wide range of frequencies in the signal. Unfortunately, square waves, as in digital data, have a wide spectrum and thus are subject to strong attenuation and delay distortion. These effects make baseband (DC) signaling unsuitable except at slow speeds and over short distances.

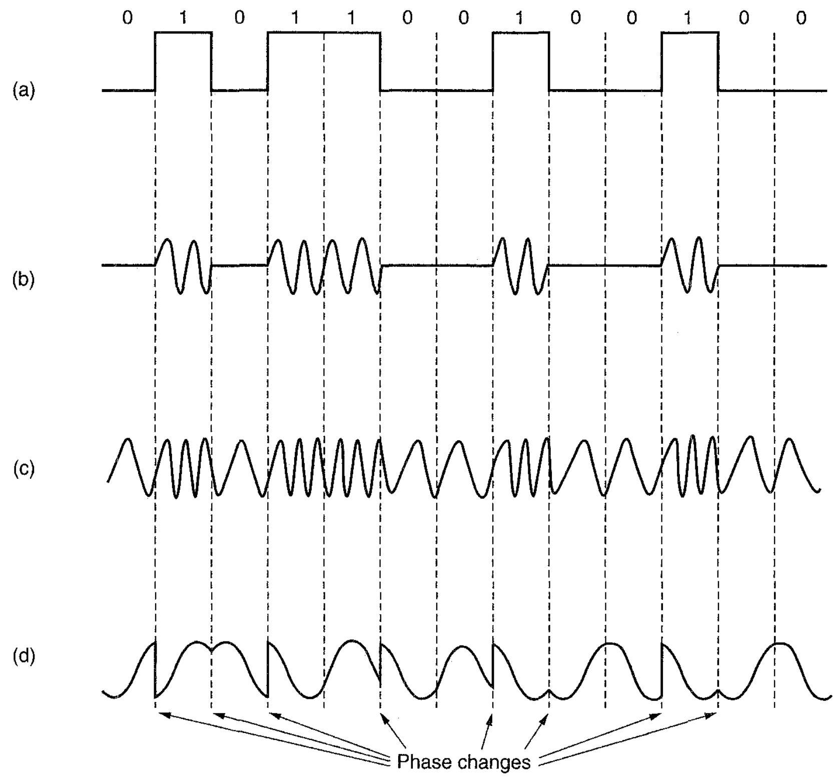

Fig. 2-18. (a) A binary signal. (b) Amplitude modulation. (c) Frequency modulation. (d) Phase modulation.

To get around the problems associated with DC signaling, especially on telephone lines, AC signaling is used. A continuous tone in the 1000- to 2000-Hz range, called a **sine wave carrier** is introduced. Its amplitude, frequency, or phase can be modulated to transmit information. In **amplitude modulation**, two different voltage levels are used to represent 0 and 1, respectively. In **frequency modulation**, also known as **frequency shift keying**, two (or more) different tones are used. In the simplest form of **phase modulation**, the carrier wave is systematically shifted 45, 135, 225, or 315 degrees at uniformly spaced intervals. Each phase shift transmits 2 bits of information. Figure 2-18 illustrates the three forms of modulation. A device that accepts a serial stream of bits as input and produces a modulated carrier as output (or vice versa) is called a **modem** (for modulator-demodulator). The modem is inserted between the (digital) computer and the (analog) telephone system.

To go to higher and higher speeds, it is not possible to just keep increasing the sampling rate. The Nyquist theorem says that even with a perfect 3000-Hz line (which a dial-up telephone is decidedly not), there is no point in sampling faster than 6000 Hz. Thus all research on faster modems is focused on getting more bits per sample (i.e., per baud).

Most advanced modems use a combination of modulation techniques to transmit multiple bits per baud. In Fig. 2-19(a), we see dots at 0, 90, 180, and 270 degrees, with two amplitude levels per phase shift. Amplitude is indicated by the distance from the origin. In Fig. 2-19(b) we see a different modulation scheme, in which 16 different combinations of amplitude and phase shift are used. Thus Fig. 2-19(a) has eight valid combinations and can be used to transmit 3 bits per baud. In contrast, Fig. 2-19(b) has 16 valid combinations and can thus be used to transmit 4 bits per baud. The scheme of Fig. 2-19(b) when used to transmit 9600 bps over a 2400-baud line is called **QAM** (**Quadrature Amplitude Modulation**).

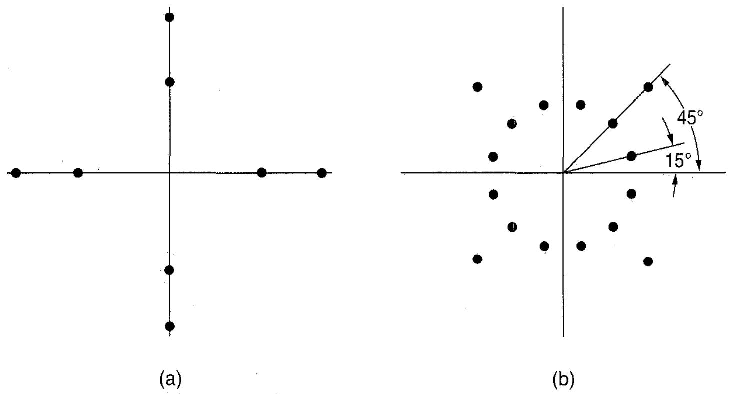

Fig. 2-19. (a) 3 bits/baud modulation. (b) 4 bits/baud modulation.

Diagrams such as those of Fig. 2-19, which show the legal combinations of amplitude and phase, are called **constellation patterns**. Each high-speed modem standard has its own constellation pattern and can talk only to other modems that use the same one (although most modems can emulate all the slower ones). The ITU V.32 9600 bps modem standard uses the constellation pattern of Fig. 2-19(b), for example.

The next step above 9600 bps is 14,400 bps. It is called **V.32 bis**. This speed is achieved by transmitting 6 bits per sample at 2400 baud. Its constellation pattern has 64 points. Fax modems use this speed to transmit pages that have been scanned in as bit maps. After V.32 bis comes **V.34**, which runs at 28,800 bps.

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With so many points in the constellation pattern, even a small amout of noise in the detected amplitude or phase can result in an error, and potentially 6 bad bits. To reduce the chance of getting an error, many modems add a parity bit, giving 128 points in the constellation pattern. The coding of the points is carefully done to maximize the chance of detecting errors. The coding that does this is called **trellis coding**.

A completely different approach to high-speed transmission is to divide the available 3000-Hz spectrum into 512 tiny bands and transmit at, say, 20 bps in each one. This scheme requires a substantial processor inside the modem, but has the advantage of being able to disable frequency bands that are too noisy. Modems that use this approach normally have V.32 or V.34 capability as well, so they can talk to standard modems.

Many modems now have compression and error correction built into the modems. The big advantage of this approach is that these features improve the effective data rate without requiring any changes to existing software. One popular compression scheme is **MNP 5**, which uses run-length encoding to squeeze out runs of identical bytes. Fax modems also use run-length encoding, since runs of 0s (blank paper) are very common. Another scheme is **V.42 bis**, which uses a Ziv-Lempel compression algorithm also used in Compress and other programs (Ziv and Lempel, 1977).

Even when modems are used, another problem can occur on telephone lines: echoes. On a long line, when the signal gets to the final destination, some of the energy may be reflected back, analogous to acoustic echos in the mountains. As an illustration of electromagnetic echoes, try shining a flashlight from a darkened room through a closed window at night. You will see a reflection of the flashlight in the window (i.e., some of the energy has been reflected at the air-glass junction and sent back toward you). The same thing happens on transmission lines, especially at the point where the local loop terminates in the end office.

The effect of the echo is that a person speaking on the telephone hears his own words after a short delay. Psychological studies have shown that this is annoying to many people, often making them stutter or become confused. To eliminate the problem of echoes, echo suppressors are installed on lines longer than 2000 km. (On short lines the echoes come back so fast that people are not bothered by them.) An **echo suppressor** is a device that detects human speech coming from one end of the connection and suppresses all signals going the other way. It is basically an amplifier than can be switched on and off by a control signal produced by a speech detection circuit.

When the first person stops talking and the second begins, the echo suppressor switches directions. A good echo suppressor can reverse in 2 to 5 msec. While it is functioning, however, information can only travel in one direction; echoes cannot get back to the sender. Figure 2-20(a) shows the state of the echo suppressors while *A* is talking to *B*. Figure 2-20(b) shows the state after *B* has started talking.

The echo suppressors have several properties that are undesirable for data

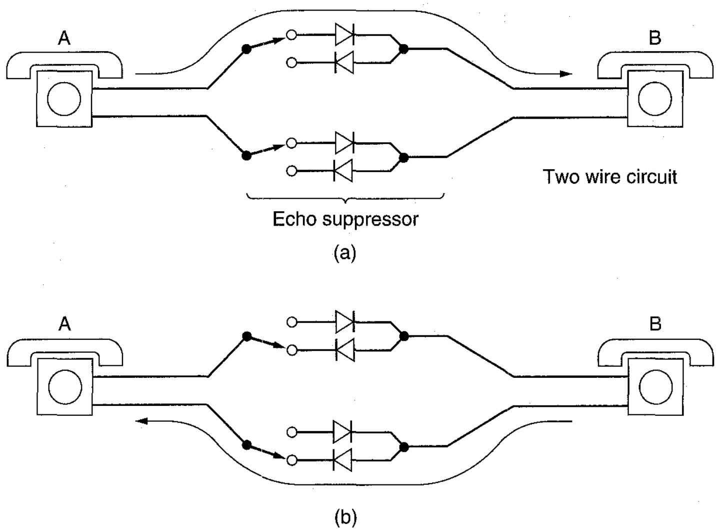

Fig. 2-20. (a) *A* talking to *B*. (b) *B* talking to *A*.

communication. First, it they were not present, it would be possible to transmit in both directions at the same time by using a different frequency band for each direction. This approach is called **full-duplex** transmission. With echo suppressors, full-duplex transmission is impossible. The alternative is **half-duplex** transmission, in which communication can go either way, but only one at a time. A single railroad track is half-duplex. Even if half-duplex transmission is adequate, it is a nuisance because the time required to switch directions can be substantial. Furthermore, the echo suppressors are designed to reverse upon detecting human speech, not digital data.

To alleviate these problems, an escape hatch has been provided on telephone circuits with echo suppressors. When the echo suppressors hear a pure tone at 2100 Hz, they shut down and remain shut down as long as a carrier is present. This arrangement is one of the many examples of **in-band signaling**, so called because the control signals that activate and deactivate internal control functions lie within the band accessible to the user. In general the trend is away from in-band signaling, to prevent users from interfering with the operation of the system itself. In the United States, most of the in-band signaling is gone, but in other countries it still exists.

An alternative to echo suppressors are **echo cancelers.** These are circuits that simulate the echo, estimate how much it is, and subtract it from the signal delivered, without the need for mechanical relays. When echo cancelers are used, full-duplex operation is possible. For this reason, echo cancelers are rapidly replacing echo suppressors in the United States and other large countries.

RS-232-C and RS-449

The interface between the computer or terminal and the modem is an example of a physical layer protocol. It must specify in detail the mechanical, electrical, functional, and procedural interface. We will now look closely at two well-known physical layer standards: RS-232-C and its successor, RS-449.

Let us start with **RS-232-C**, the third revision of the original RS-232 standard. The standard was drawn up by the Electronic Industries Association, a trade organization of electronics manufacturers, and is properly referred to as EIA RS-232-C. The international version is given in CCITT recommendation **V.24**, which is similar but differs slightly on some of the rarely used circuits. In the standards, the terminal or computer is officially called a **DTE** (**Data Terminal Equipment**) and the modem is officially called a **DCE** (**Data Circuit-Terminating Equipment**).

The mechanical specification is for a 25-pin connector 47.04 ± .13 mm wide (screw center to screw center), with all the other dimensions equally well specified. The top row has pins numbered 1 to 13 (left to right); the bottom row has pins numbered 14 to 25 (also left to right).

The electrical specification for RS-232-C is that a voltage more negative than −3 volts is a binary 1 and a voltage more positive than +4 volts is a binary 0. Data rates up to 20 kbps are permitted, as are cables up to 15 meters.

The functional specification tells which circuits are connected to each of the 25 pins, and what they mean. Figure 2-21 shows 9 pins that are nearly always implemented. The remaining ones are frequently omitted. When the terminal or computer is powered up, it asserts (i.e., sets to a logical 1) Data Terminal Ready (pin 20). When the modem is powered up, it asserts Data Set Ready (pin 6). When the modem detects a carrier on the telephone line, it asserts Carrier Detect (pin 8). Request to Send (pin 4) indicates that the terminal wants to send data. Clear to Send (pin 5) means that the modem is prepared to accept data. Data are transmitted on the Transmit circuit (pin 2) and received on the Receive circuit (pin 3).

Other circuits are provided for selecting the data rate, testing the modem, clocking the data, detecting ringing signals, and sending data in the reverse direction on a secondary channel. They are hardly ever used in practice.

The procedural specification is the protocol, that is, the legal sequence of events. The protocol is based on action-reaction pairs. When the terminal asserts Request to Send, for example, the modem replies with Clear to Send, if it is able to accept data. Similar action-reaction pairs exist for other circuits as well.

It commonly occurs that two computers must be connected using RS-232-C. Since neither one is a modem, there is an interface problem. This problem is solved by connecting them with a device called a **null modem**, which connects the transmit line of one machine to the receive line of the other. It also crosses some of the other lines in a similar way. A null modem looks like a short cable.

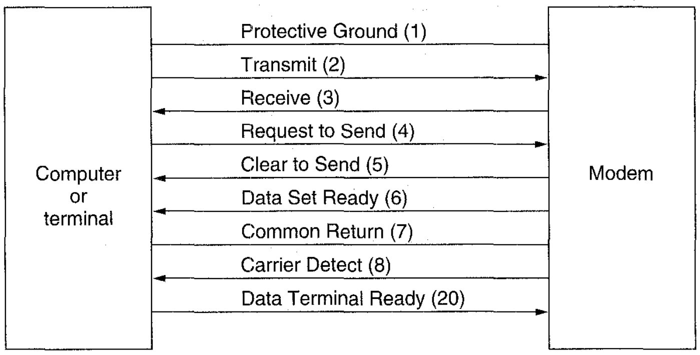

Fig. 2-21. Some of the principal RS-232-C circuits. The pin numbers are given in parentheses.

RS-232-C has been around for years. Gradually, the limitation of the data rate to not more than 20 kbps and the 15-meter maximum cable length have become increasingly annoying. EIA had a long debate about whether to try to have a new standard that was compatible with the old one (but technically not very advanced) or a new and incompatible one that would meet all needs for years to come. They eventually compromised by choosing both.

The new standard, called **RS-449**, is actually three standards in one. The mechanical, functional, and procedural interfaces are given in RS-449, but the electrical interface is given by two different standards. The first of these, **RS-423-A**, is similar to RS-232-C in that all its circuits share a common ground. This technique is called **unbalanced transmission**. The second electrical standard, **RS-422-A**, in contrast, uses **balanced transmission**, in which each of the main circuits requires two wires, with no common ground. As a result, RS-422-A can be used at speeds up to 2 Mbps over 60-meter cables.

The circuits used in RS-449 are shown in Fig. 2-22. Several new circuits not present in RS-232-C have been added. In particular, circuits for testing the modem both locally and remotely were included. Due to the inclusion of a number of two-wire circuits (when RS-422-A is used), more pins are needed in the new standard, so the familiar 25-pin connector was dropped. In its place is a 37-pin connector and a 9-pin connector. The 9-pin connector is required only if the second (reverse) channel is being used.

Fiber in the Local Loop

For advanced future services, such as video on demand, the 3-kHz channel currently used will not do. Discussions about what to do about this tend to focus on two solutions. The straightforward one—running a fiber from the end office

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	RS-232-C			CCITT V .24			RS-449		
	Code	Pin	Circuit	Code	Pin	Circuit	Code	Pin	Circuit
	AA	1	Protective ground	101	1	Protective ground	_	1	
	AB	7	Signal ground	102	7	Signal ground	SG	19	Signal ground
							SC	37	Send common
							RC	20	Receive common
	BA	2	Transmitted data	103	2	Transmitted data	SD	4, 22	Send data
	BB	3	Received data	104	3	Received data	RD	6, 24	Receive data
	CA	4	Request to send	105	4	Request to send	RS	7, 25	Request to send
	CB	5	Clear to send	106	5	Ready for sending	CS	9, 27	Clear to send
	CC	6	Data set ready	107	6	Data set ready	DM	11, 29	Data mode
	CD	20	Data terminal ready	108	20	Data terminal ready	TR	12, 30	Terminal ready
	CE	22	Ring indicator	125	22	Calling indicator	IC	15	Incoming call
	CF	8	Line detector	109	8	Line detector	RR	13, 31	Receiver ready
	CG	21	Signal quality	110	21	Signal quality	SQ	33	Signal quality
	CH	23	DTE rate	111	23	DTE rate	SR	16	Signaling rate
	CI	18	DCE rate	112	18	DCE rate	SI	2	Signaling indicators
							IS	28	Terminal in service
				136		New signal	NS	34	New signal
				126	11	Select frequency	SF	16	Select frequency
	DA	24	DTE timing	113	24	DTE timing	TT	17, 25	Terminal timing
	DB	15	DCE timing	114	15	DCE timing	ST	5, 23	Send timing
	DD	17	Receiver timing	115	17	Receiver timing	RT	8, 26	Receive timing
Secondary Channel	SBA	14	Transmitted data	118	14	Transmitted data	SSD	3	Send data
	SBB	16	Received data	119	16	Received data	SRD	4	Receive data
	SCA	19	Request to send	120	19	Line signal	SRS	7	Request to send
	SCB	13	Clear to send	121	13	Channel ready	SCS	8	Clear to send
	SCF	12	Line detector	122	12	Line detector	SRR	2	Receiver ready
							LL	10	Local loopback
							RL	14	Remote loopback
							TM	18	Test mode
							SS	32	Select standby
							SB	36	Standby indicator

Fig. 2-22. Comparison of RS-232-C, V.24, and RS-449.

into everyone's house is called **FTTH** (**Fiber To The Home**). This solution fits in well with the current system but will not be economically feasible for decades. It is simply too expensive.

An alternative solution that is much cheaper is **FTTC** (**Fiber To The Curb**). In this model, the telephone company runs an optical fiber from each end office into each neighborhood (the curb) that it serves (Paff, 1995). The fiber is

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terminated in a junction box that all the local loops enter. Since the local loops are now much shorter (perhaps 100 meters instead of 3 km), they can be run at higher speeds, probably around 1 Mbps, which is just enough for compressed video. This design is shown in Fig. 2-23(a).

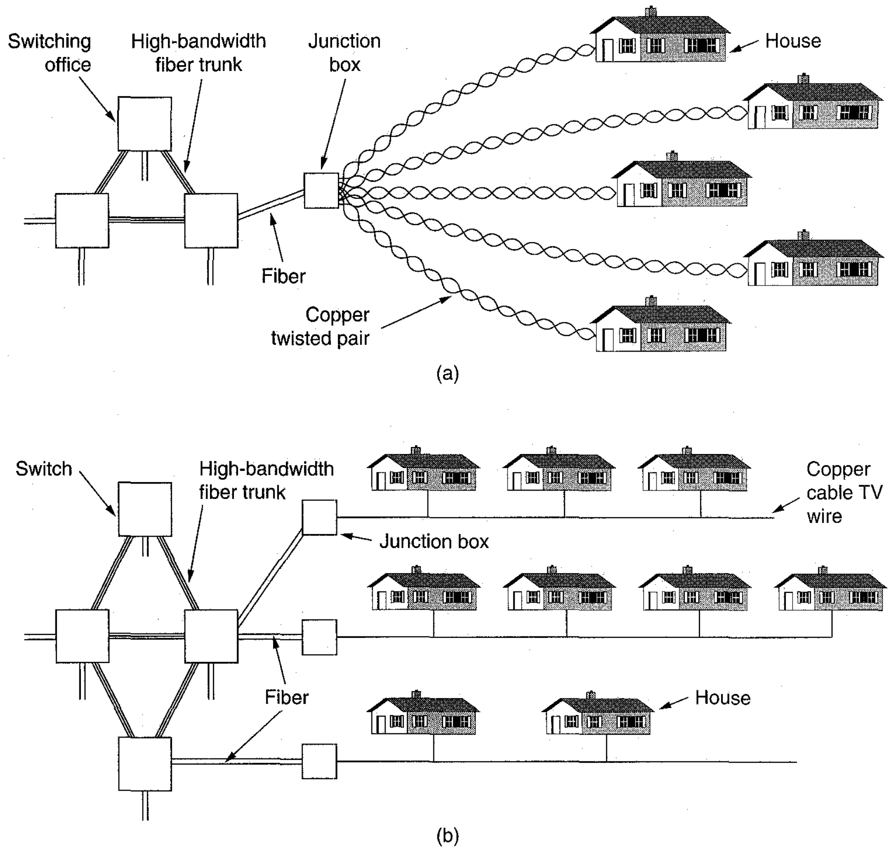

Fig. 2-23. Fiber to the curb. (a) Using the telephone network. (b) Using the cable TV network.

In this manner, multiple videos (or other information channels) can pour down the fiber at high speed and be split over the twisted pairs at the end. By sharing a 1-Gbps fiber over 100 to 1000 customers, the cost per customer can be reduced, and considerably higher bandwidth can be provided than now. Going appreciably above 1 Mbps for long distances with the existing twisted pairs is impossible. Thus in the long term, all the twisted pairs will have to be replaced by fiber. Whether the intermediate solution of FTTC should be used for the time being or

FTTH should be the goal from the beginning is a matter of some debate within the telephone industry.

An alternative design using the existing cable TV infrastructure is shown in Fig. 2-23(b). Here a multidrop cable is used instead of the point-to-point system characteristic of the telephone system. It is likely that both Fig. 2-23(a) and Fig. 2-23(b) will coexist in the future, as telephone companies and cable TV operators become direct competitors for voice, data, and possibly even television service. For more information about this topic, see (Cook and Stern, 1994; Miki, 1994b; and Mochida, 1994).

2.4.4. Trunks and Multiplexing

Economies of scale play an important role in the telephone system. It costs essentially the same amount of money to install and maintain a high-bandwidth trunk as a low-bandwidth trunk between two switching offices (i.e., the costs come from having to dig the trench and not from the copper wire or optical fiber). Consequently, telephone companies have developed elaborate schemes for multiplexing many conversations over a single physical trunk. These multiplexing schemes can be divided into two basic categories: **FDM** (**Frequency Division Multiplexing**), and **TDM** (**Time Division Multiplexing**). In FDM the frequency spectrum is divided among the logical channels, with each user having exclusive possession of some frequency band. In TDM the users take turns (in a round robin), each one periodically getting the entire bandwidth for a little burst of time.

AM radio broadcasting provides illustrations of both kinds of multiplexing. The allocated spectrum is about 1 MHz, roughly 500 to 1500 kHz. Different frequencies are allocated to different logical channels (stations), each operating in a portion of the spectrum, with the interchannel separation great enough to prevent interference. This system is an example of frequency division multiplexing. In addition (in some countries), the individual stations have two logical subchannels: music and advertising. These two alternate in time on the same frequency, first a burst of music, then a burst of advertising, then more music, and so on. This situation is time division multiplexing.

Below we will examine frequency division multiplexing. After that we will see how FDM can be applied to fiber optics (wavelength division multiplexing). Then we will turn to TDM, and end with an advanced TDM system used for fiber optics (SONET).

Frequency Division Multiplexing

Figure 2-24 shows how three voice-grade telephone channels are multiplexed using FDM. Filters limit the usable bandwidth to about 3000 Hz per voice-grade channel. When many channels are multiplexed together, 4000 Hz is allocated to each channel to keep them well separated. First the voice channels are raised in frequency, each by a different amount. Then they can be combined, because no two channels now occupy the same portion of the spectrum. Notice that even though there are gaps (guard bands) between the channels, there is some overlap between adjacent channels, because the filters do not have sharp edges. This overlap means that a strong spike at the edge of one channel will be felt in the adjacent one as nonthermal noise.

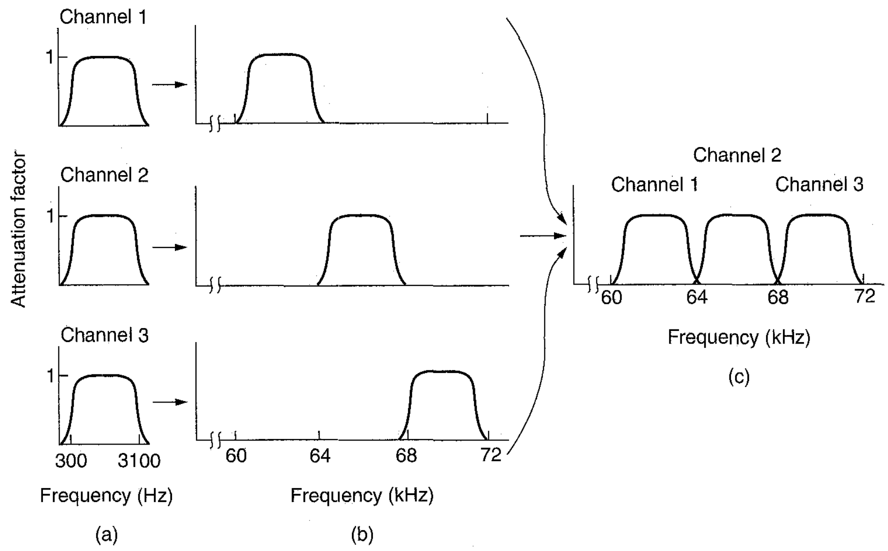

Fig. 2-24. Frequency division multiplexing. (a) The original bandwidths. (b) The bandwidths raised in frequency. (c) The multiplexed channel.

The FDM schemes used around the world are to some degree standardized. A widespread standard is 12 4000-Hz voice channels (3000 Hz for the user, plus two guard bands of 500 Hz each) multiplexed into the 60 to 108 kHz band. This unit is called a **group**. The 12- to 60-kHz band is sometimes used for another group. Many carriers offer a 48- to 56-kbps leased line service to customers, based on the group. Five groups (60 voice channels) can be multiplexed to form a **supergroup**. The next unit is the **mastergroup**, which is five supergroups (CCITT standard) or ten supergroups (Bell system). Other standards up to 230,000 voice channels also exist.

Wavelength Division Multiplexing

For fiber optic channels, a variation of frequency division multiplexing is used. It is called **WDM** (**Wavelength Division Multiplexing**). A simple way of achieving FDM on fibers is depicted in Fig. 2-25. Here two fibers come together at a prism (or more likely, a diffraction grating), each with its energy in a different band. The two beams are passed through the prism or grating, and combined onto a single shared fiber for transmission to a distant destination, where they are split again.

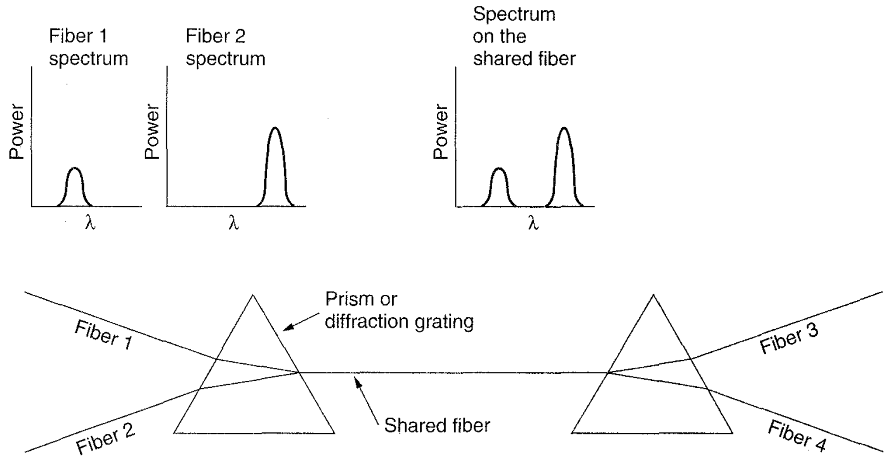

Fig. 2-25. Wavelength division multiplexing.

There is really nothing new here. As long as each channel has its own frequency range, and all the ranges are disjoint, they can be multiplexed together on the long-haul fiber. The only difference with electrical FDM is that an optical system using a diffraction grating is completely passive, and thus highly reliable.

It should be noted that the reason WDM is popular is that the energy on a single fiber is typically only a few gigahertz wide because it is currently impossible to convert between electrical and optical media any faster. Since the bandwidth of a single fiber band is about 25,000 GHz (see Fig. 2-6), there is great potential for multiplexing many channels together over long-haul routes. A necessary condition, however, is that the incoming channels use different frequencies.

A potential application of WDM is in the FTTC systems described earlier. Initially, a telephone company could run a single fiber from an end office to a neighborhood junction box where it met up with twisted pairs from the houses. Years later, when the cost of fiber is lower and the demand for it is higher, the twisted pairs can be replaced by fiber and all the local loops joined onto the fiber running to the end office using WDM.

In the example of Fig. 2-25, we have a fixed wavelength system. Bits from fiber 1 go to fiber 3, and bits from fiber 2 go to fiber 4. It is not possible to have bits go from fiber 1 to fiber 4. However, it is also possible to build WDM systems that are switched. In such a device, there are many input fibers and many output

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fibers, and the data from any input fiber can go to any output fiber. Typically, the coupler is a passive star, with the light from every input fiber illuminating the star. Although spreading the energy over n outputs dilutes it by a factor n, such systems are practical for hundreds of channels.

Of course, if the light from one of the incoming fibers is at 1.50206 microns and potentially might have to go to any output fiber, all the output fibers need tunable filters so the selected one can set itself to 1.50206 microns. Such optical tunable filters can be built from Fabry-Perot or Mach-Zehnder interferometers. Alternatively, the input fibers could be tunable and the output ones fixed. Having both be tunable is an unnecessary expense and is rarely worth it.

Time Division Multiplexing

Although FDM is still used over copper wires or microwave channels, it requires analog circuitry and is not amenable to being done by a computer. In contrast, TDM can be handled entirely by digital electronics, so it has become far more widespread in recent years. Unfortunately, it can only be used for digital data. Since the local loops produce analog signals, a conversion is needed from analog to digital in the end office, where all the individual local loops come together to be combined onto outgoing trunks. We will now look at how multiple analog voice signals are digitized and combined onto a single outgoing digital trunk. (Remember that computer data sent over a modem are also analog when they get to the end office.)

The analog signals are digitized in the end office by a device called a **codec** (coder-decoder), producing a 7- or 8-bit number (see Fig. 2-17). The codec makes 8000 samples per second (125 μsec/sample) because the Nyquist theorem says that this is sufficient to capture all the information from the 4-kHz telephone channel bandwidth. At a lower sampling rate, information would be lost; at a higher one, no extra information would be gained. This technique is called **PCM** (**Pulse Code Modulation**). PCM forms the heart of the modern telephone system. As a consequence, virtually all time intervals within the telephone system are multiples of 125 μsec.

When digital transmission began emerging as a feasible technology, CCITT was unable to reach agreement on an international standard for PCM. Consequently, there are now a variety of incompatible schemes in use in different countries around the world. International hookups between incompatible countries require (often expensive) "black boxes" to convert the originating country's system to that of the receiving country.

One method that is in widespread use in North America and Japan is the T1 carrier, depicted in Fig. 2-26. (Technically speaking, the format is called DS1 and the carrier is called T1, but we will not make that subtle distinction here.) The T1 carrier consists of 24 voice channels multiplexed together. Usually, the analog signals are sampled on a round-robin basis with the resulting analog stream being fed to the codec rather than having 24 separate codecs and then merging the digital output. Each of the 24 channels, in turn, gets to insert 8 bits into the output stream. Seven bits are data, and one is for control, yielding $7 \times 8000 = 56{,}000$ bps of data, and $1 \times 8000 = 8000$ bps of signaling information per channel.

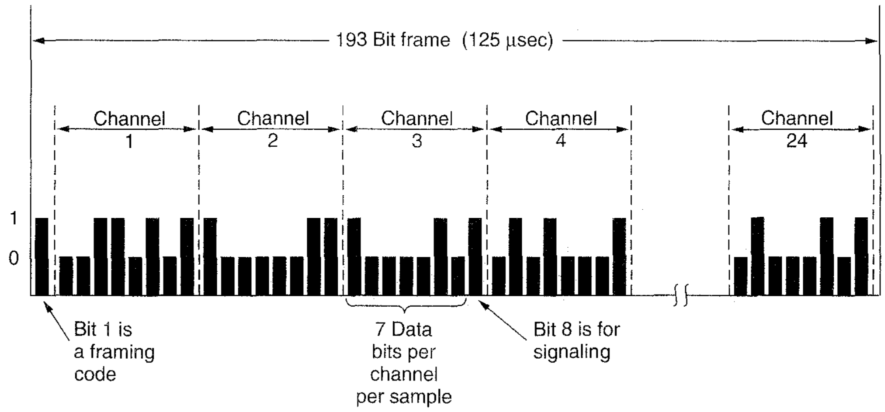

Fig. 2-26. The T1 carrier (1.544 Mbps).

A frame consists of $24 \times 8 = 192$ bits, plus one extra bit for framing, yielding 193 bits every 125 μsec. This gives a gross data rate of 1.544 Mbps. The 193rd bit is used for frame synchronization. It takes on the pattern 0101010101 . . . . Normally, the receiver keeps checking this bit to make sure that it has not lost synchronization. If it does get out of sync, the receiver can scan for this pattern to get resynchronized. Analog customers cannot generate the bit pattern at all, because it corresponds to a sine wave at 4000 Hz, which would be filtered out. Digital customers can, of course, generate this pattern, but the odds are against its being present when the frame slips. When a T1 system is being used entirely for data, only 23 of the channels are used for data. The 24th one is used for a special synchronization pattern, to allow faster recovery in the event that the frame slips.

When CCITT finally did reach agreement, they felt that 8000 bps of signaling information was far too much, so its 1.544-Mbps standard is based upon an 8- rather than a 7-bit data item; that is, the analog signal is quantized into 256 rather than 128 discrete levels. Two (incompatible) variations are provided. In **common-channel signaling**, the extra bit (which is attached onto the rear rather than the front of the 193 bit frame) takes on the values 10101010 . . . in the odd frames and contains signaling information for all the channels in the even frames.

In the other variation, **channel associated signaling**, each channel has its own private signaling subchannel. A private subchannel is arranged by allocating one of the eight user bits in every sixth frame for signaling purposes, so five out of six samples are 8 bits wide, and the other one is only 7 bits wide. CCITT also has a recommendation for a PCM carrier at 2.048 Mbps called **E1**. This carrier has 32 8-bit data samples packed into the basic 125-μsec frame. Thirty of the channels are used for information and two are used for signaling. Each group of four frames provides 64 signaling bits, half of which are used for channel associated signaling and half of which are used for frame synchronization or are reserved for each country to use as it wishes. Outside North America and Japan, the 2.048-Mbps carrier is in widespread use.

Once the voice signal has been digitized, it is tempting to try to use statistical techniques to reduce the number of bits needed per channel. These techniques are appropriate not only to encoding speech, but to the digitization of any analog signal. All of the compaction methods are based upon the principle that the signal changes relatively slowly compared to the sampling frequency, so that much of the information in the 7- or 8-bit digital level is redundant.

One method, called **differential pulse code modulation**, consists of outputting not the digitized amplitude, but the difference between the current value and the previous one. Since jumps of ±16 or more on a scale of 128 are unlikely, 5 bits should suffice instead of 7. If the signal does occasionally jump wildly, the encoding logic may require several sampling periods to "catch up." For speech, the error introduced can be ignored.

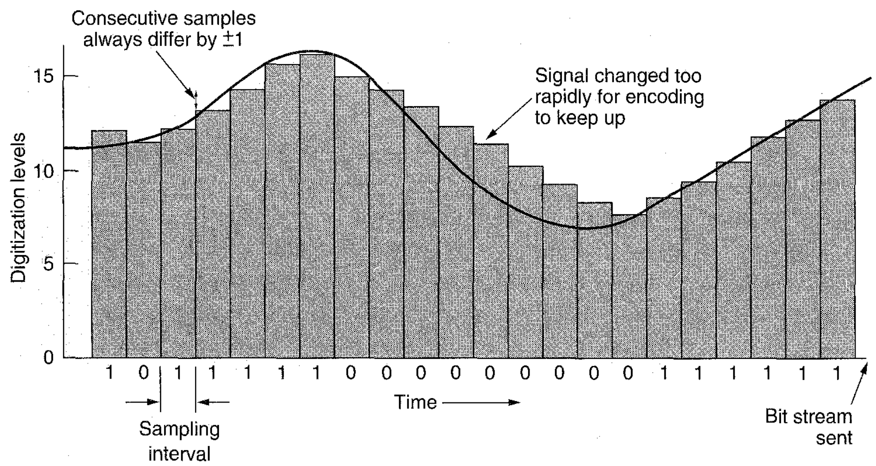

Fig. 2-27. Delta modulation.

A variation of this compaction method requires each sampled value to differ from its predecessor by either +1 or −1. A single bit is transmitted, telling whether the new sample is above or below the previous one. This technique, called **delta modulation**, is illustrated in Fig. 2-27. Like all compaction techniques that assume small level changes between consecutive samples, delta encoding can get into trouble if the signal changes too fast, as shown in the figure. When this happens, information is lost.

An improvement to differential PCM is to extrapolate the previous few values to predict the next value and then to encode the difference between the actual signal and the predicted one. The transmitter and receiver must use the same prediction algorithm, of course. Such schemes are called **predictive encoding**. They are useful because they reduce the size of the numbers to be encoded, hence the number of bits to be sent.

Although PCM is widely used on interoffice trunks, the computer user gets relatively little benefit from it if all data must be sent to the end office in the form of a modulated analog sine wave at 28.8 kbps. It would be nice if the carrier would attach the local loop directly to the PCM trunk system, so that the computer could output digital data directly onto the local loop at 1.544 or 2.048 Mbps. Unfortunately, the local loops cannot run at these speeds for very far.

Time division multiplexing allows multiple T1 carriers to be multiplexed into higher-order carriers. Figure 2-28 shows how this can be done. At the left we see four T1 channels being multiplexed onto one T2 channel. The multiplexing at T2 and above is done bit for bit, rather than byte for byte with the 24 voice channels that make up a T1 frame. Four T1 streams at 1.544 Mbps should generate 6.176 Mbps, but T2 is actually 6.312 Mbps. The extra bits are used for framing and recovery, in case the carrier slips.

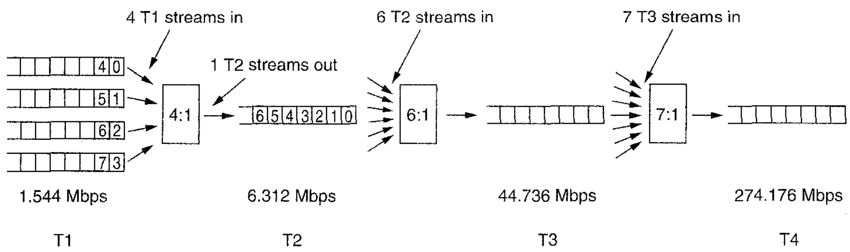

Fig. 2-28. Multiplexing T1 streams onto higher carriers.

At the next level, six T2 streams are combined bitwise to form a T3 stream. Then seven T3 streams are joined to form a T4 stream. At each step a small amount of overhead is added for framing and recovery.

Just as there is little agreement on the basic carrier between the United States and the rest of the world, there is equally little agreement on how it is to be multiplexed into higher bandwidth carriers. The U.S. scheme of stepping up by 4, 6, and 7 did not strike everyone else as the way to go, so the CCITT standard calls for multiplexing four streams onto one stream at each level. Also, the framing and recovery data are different. The CCITT hierarchy for 32, 128, 512, 2048, and 8192 channels runs at speeds of 2.048, 8.848, 34.304, 139.264, and 565.148 Mbps.

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SONET/SDH

In the early days of fiber optics, every telephone company had its own proprietary optical TDM system. After AT&T was broken up in 1984, local telephone companies had to connect to multiple long-distance carriers, all with different optical TDM systems, so the need for standardization became obvious. In 1985, Bellcore, the RBOCs research arm, began working on a standard, called **SONET (Synchronous Optical NETwork)**. Later, CCITT joined the effort, which resulted in a SONET standard and a set of parallel CCITT recommendations (G.707, G.708, and G.709) in 1989. The CCITT recommendations are called **SDH (Synchronous Digital Hierarchy)** but differ from SONET only in minor ways. Virtually all the long-distance telephone traffic in the United States, and much of it elsewhere now uses trunks running SONET in the physical layer. As SONET chips become cheaper, SONET interface boards for computers may become more widespread, so it may become easier for companies to plug their computers directly into the heart of the telephone network over specially conditioned leased lines. Below we will discuss the goals and design of SONET briefly. For additional information see (Bellamy, 1991; and Omidyar and Aldridge, 1993).

The SONET design had four major goals. First and foremost, SONET had to make it possible for different carriers to interwork. Achieving this goal required defining a common signaling standard with respect to wavelength, timing, framing structure, and other issues.

Second, some means was needed to unify the U.S., European, and Japanese digital systems, all of which were based on 64-kbps PCM channels, but all of which combined them in different (and incompatible) ways.

Third, SONET had to provide a way to multiplex multiple digital channels together. At the time SONET was devised, the highest speed digital carrier actually used widely in the United States was T3, at 44.736 Mbps. T4 was defined, but not used much, and nothing was even defined above T4 speed. Part of SONET's mission was to continue the hierarchy to gigabits/sec and beyond. A standard way to multiplex slower channels into one SONET channel was also needed.

Fourth, SONET had to provide support for operations, administration, and maintenance (OAM). Previous systems did not do this very well.

An early decision was to make SONET a traditional TDM system, with the entire bandwidth of the fiber devoted to one channel containing time slots for the various subchannels. As such, SONET is a synchronous system. It is controlled by a master clock with an accuracy of about 1 part in 10^9. Bits on a SONET line are sent out at extremely precise intervals, controlled by the master clock.

When cell switching was later proposed to be the basis of broadband ISDN, the fact that it permitted irregular cell arrivals got it labeled as *asynchronous* transfer mode (i.e., ATM) to contrast it to the synchronous operation of SONET.

A SONET system consists of switches, multiplexers, and repeaters, all connected by fiber. A path from a source to destination with one intermediate multiplexer and one intermediate repeater is shown in Fig. 2-29. In SONET terminology, a fiber going directly from any device to any other device, with nothing in between, is called a **section**. A run between two multiplexers (possibly with one or more repeaters in the middle) is called a **line**. Finally, the connection between the source and destination (possibly with one or more multiplexers and repeaters) is called a **path**. The SONET topology can be a mesh, but is often a dual ring.

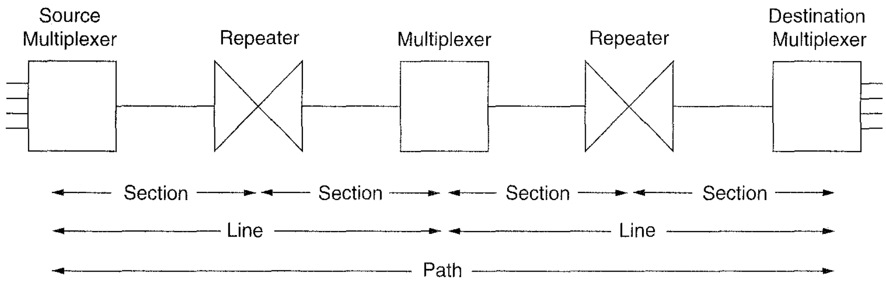

Fig. 2-29. A SONET path.

The basic SONET frame is a block of 810 bytes put out every 125 μsec. Since SONET is synchronous, frames are emitted whether or not there are any useful data to send. Having 8000 frames/sec exactly matches the sampling rate of the PCM channels used in all digital telephony systems.

The 810-byte SONET frames are best described as a rectangle of bytes, 90 columns wide by 9 rows high. Thus $8 \times 810 = 6480$ bits are transmitted 8000 times per second, for a gross data rate of 51.84 Mbps. This is the basic SONET channel and is called **STS-1** (**Synchronous Transport Signal-1**). All SONET trunks are a multiple of STS-1.

The first three columns of each frame are reserved for system management information, as illustrated in Fig. 2-30. The first three rows contain the section overhead; the next six contain the line overhead. The section overhead is generated and checked at the start and end of each section, whereas the line overhead is generated and checked at the start and end of each line.

The remaining 87 columns hold $87 \times 9 \times 8 \times 8000 = 50.112$ Mbps of user data. However, the user data, called the **SPE** (**Synchronous Payload Envelope**) do not always begin in row 1, column 4. The SPE can begin anywhere within the frame. A pointer to the first byte is contained in the first row of the line overhead. The first column of the SPE is the path overhead (i.e., header for the end-to-end path sublayer protocol).

The ability to allow the SPE to begin anywhere within the SONET frame, and even to span two frames, as shown in Fig. 2-30, gives added flexibility to the

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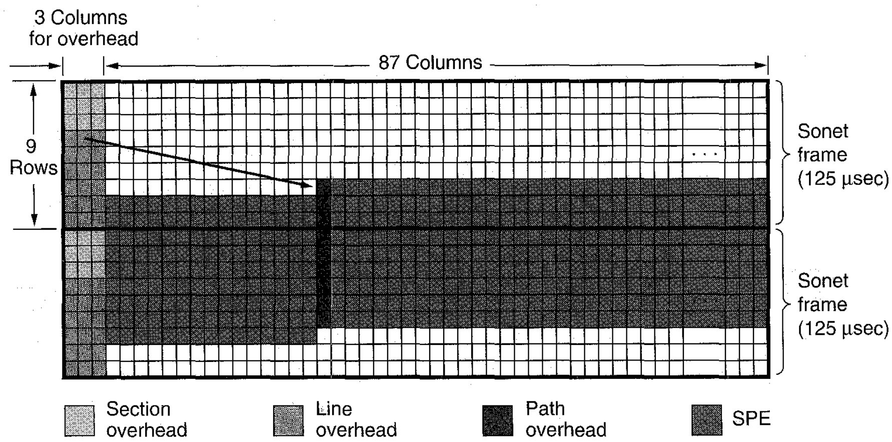

Fig. 2-30. Two back-to-back SONET frames.

system. For example, if a payload arrives at the source while a dummy SONET frame is being constructed, it can be inserted into the current frame, instead of being held until the start of the next one. This feature is also useful when the payload does not fit exactly in one frame, as in the case of a sequence of 53-byte ATM cells. The first row of the line overhead can then point to the start of the first full cell, to provide synchronization.

The section, line, and path overheads contain a profusion of bytes used for operations, administration, and maintenance. Since each byte occurs 8000 times per second, it represents a PCM channel. Three of these are, in fact, used to provide voice channels for section, line, and path maintenance personnel. Other bytes are used for framing, parity, error monitoring, IDs, clocking, synchronization, and other functions. Bellamy (1991) describes all the fields in detail.

The multiplexing of multiple data streams, called **tributaries**, plays an important role in SONET. Multiplexing is illustrated in Fig. 2-31. On the left, we start with various low-speed input streams, which are converted to the basic STS-1 SONET rate, in most cases by adding filler to round up to 51.84 Mbps. Next, three STS-1 tributaries are multiplexed onto one 155.52-Mbps STS-3 output stream. This stream, in turn, is multiplexed with three others onto a final output stream having 12 times the capacity of the STS-1 stream. At this point the signal is scrambled, to prevent long runs of 0s or 1s from interfering with the clocking, and converted from an electrical to an optical signal.

Multiplexing is done byte for byte. For example, when three STS-1 tributaries at 51.84 Mbps are merged into one STS-3 stream at 155.52 Mbps, the multiplexer first outputs 1 byte from tributary 1, then 1 from tributary 2, and finally 1 from tributary 3, before going back to 1. The STS-3 figure analogous to Fig. 2-30

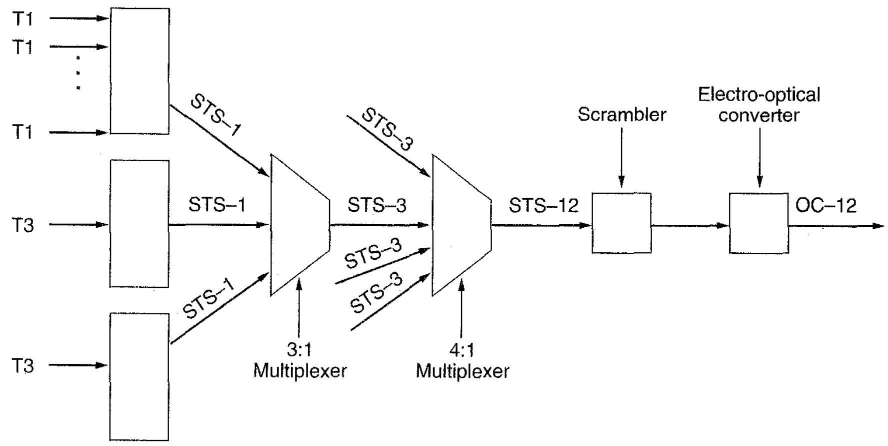

Fig. 2-31. Multiplexing in SONET.

shows (from left to right) columns from tributaries 1, 2, and 3, in that order, then another triple, and so on, out to column 270. One of these 270×9 byte frames is sent every 125 μsec, giving the 155.52-Mbps data rate.

The SONET multiplexing hierarchy is shown in Fig. 2-32. Rates from STS-1 to STS-48 have been defined. The optical carrier corresponding to STS-*n* is called OC-*n* but is bit-for-bit the same except for the scrambling shown in Fig. 2-31. The SDH names are different, and they start at OC-3 because CCITT-based systems do not have a rate near 51.84 Mbps. The OC-9 carrier is present because it closely matches the speed of a major high-speed trunk used in Japan. OC-18 and OC-36 will be used in Japan in the future. The gross data rate includes all the overhead. The SPE data rate excludes the line and section overhead. The user data rate excludes all overhead and only counts the 86 columns available for the payload.

As an aside, when a carrier, such as OC-3, is not multiplexed, but carries the data from only a single source, the letter *c* (for concatenated) is appended to the designation, so OC-3 indicates a 155.52-Mbps carrier consisting of three separate OC-1 carriers, but OC-3c indicates a data stream from a single source at 155.52 Mbps. The three OC-1 streams within an OC-3c stream are interleaved by column, first column 1 from stream 1, then column 1 from stream 2, then column 1 from stream 3, followed by column 2 from stream 1, and so on, leading to a frame 270 columns wide and 9 rows deep.

The amount of actual user data in an OC-3c stream is slightly higher than in an OC-3 stream (149.760 Mbps versus 148.608 Mbps) because the path overhead column is included inside the SPE only once, instead of the three times it would be with three independent OC-1 streams. In other words, 260 of the 270 columns

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SONET		SDH	Data rate (Mbps)		
Electrical	Optical	Optical	Gross	SPE	User
STS-1	OC-1		51.84	50.112	49.536
STS-3	OC-3	STM-1	155.52	150.336	148.608
STS-9	OC-9	STM-3	466.56	451.008	445.824
STS-12	OC-12	STM-4	622.08	601.344	594.432
STS-18	OC-18	STM-6	933.12	902.016	891.648
STS-24	OC-24	STM-8	1244.16	1202.688	1188.864
STS-36	OC-36	STM-12	1866.24	1804.032	1783.296
STS-48	OC-48	STM-16	2488.32	2405.376	2377.728

Fig. 2-32. SONET and SDH multiplex rates.

are available for user data in OC-3c, whereas only 258 columns are available for user data in OC-3. Higher-order concatenated frames (e.g., OC-12c) also exist.

By now it should be clear why ATM runs at 155 Mbps: the intention is to carry ATM cells over SONET OC-3c trunks. It should also be clear that the widely quoted 155-Mbps figure is the gross rate, including the SONET overhead. Furthermore, somewhere along the way somebody incorrectly rounded 155.52 Mbps to 155 Mbps instead of 156 Mbps, and now everyone else does it wrong, too.

The SONET physical layer is divided up into four sublayers, as shown in Fig. 2-33. The lowest sublayer is the **photonic sublayer**. It is concerned with specifying the physical properties of the light and fiber to be used.

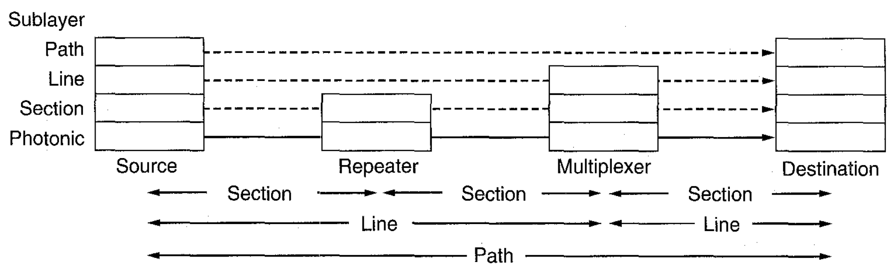

Fig. 2-33. The SONET architecture.

The three remaining sublayers correspond to the sections, lines, and paths. The section sublayer handles a single point-to-point fiber run, generating a standard frame at one end and processing it at the other. Sections can start and end at repeaters, which just amplify and regenerate the bits, but do not change or process them in any way.

The line sublayer is concerned with multiplexing multiple tributaries onto a single line and demultiplexing them at the other end. To the line sublayer, the repeaters are transparent. When a multiplexer puts out bits on a fiber, it expects them to arrive at the next multiplexer unchanged, no matter how many repeaters are used in between. The protocol in the line sublayer is thus between two multiplexers and deals with issues such as how many inputs are being multiplexed together and how. In contrast, the path sublayer and protocol deal with end-to-end issues.

2.4.5. Switching

From the point of view of the average telephone engineer, the phone system is divided into two parts: outside plant (the local loops and trunks, since they are outside the switching offices), and inside plant (the switches). We have just looked at outside plant. Now it is time to examine inside plant.

Two different switching techniques are used inside the telephone system: circuit switching and packet switching. We will give a brief introduction to each of them below. Then we will go into circuit switching in detail, because that is how the current telephone system works. Later in the chapter we will go into packet switching in detail in the context of the next generation telephone system, broadband ISDN.

Circuit Switching

When you or your computer places a telephone call, the switching equipment within the telephone system seeks out a physical "copper" (including fiber and radio) path all the way from your telephone to the receiver's telephone. This technique is called **circuit switching** and is shown schematically in Fig. 2-34(a). Each of the six rectangles represents a carrier switching office (end office, toll office, etc.). In this example, each office has three incoming lines and three outgoing lines. When a call passes through a switching office, a physical connection is (conceptually) established between the line on which the call came in and one of the output lines, as shown by the dotted lines.

In the early days of the telephone, the connection was made by having the operator plug a jumper cable into the input and output sockets. In fact, there is a surprising little story associated with the invention of automatic circuit switching equipment. It was invented by a 19th Century undertaker named Almon B. Strowger. Shortly after the telephone was invented, when someone died, one of the survivors would call the town operator and say: "Please connect me to an undertaker." Unfortunately for Mr. Strowger, there were two undertakers in his town, and the other one's wife was the town telephone operator. He quickly saw that either he was going to have to invent automatic telephone switching equipment or he was going to go out of business. He chose the first option. For nearly 100 years, the circuit switching equipment used worldwide was known as Strowger gear. (History does not record whether the now-unemployed switchboard operator got a job as an information operator, answering questions such as: What is the phone number of an undertaker?

The model shown in Fig. 2-34(a) is highly simplified of course, because parts of the "copper" path between the two telephones may, in fact, be microwave links onto which thousands of calls are multiplexed. Nevertheless, the basic idea is valid: once a call has been set up, a dedicated path between both ends exists and will continue to exist until the call is finished.

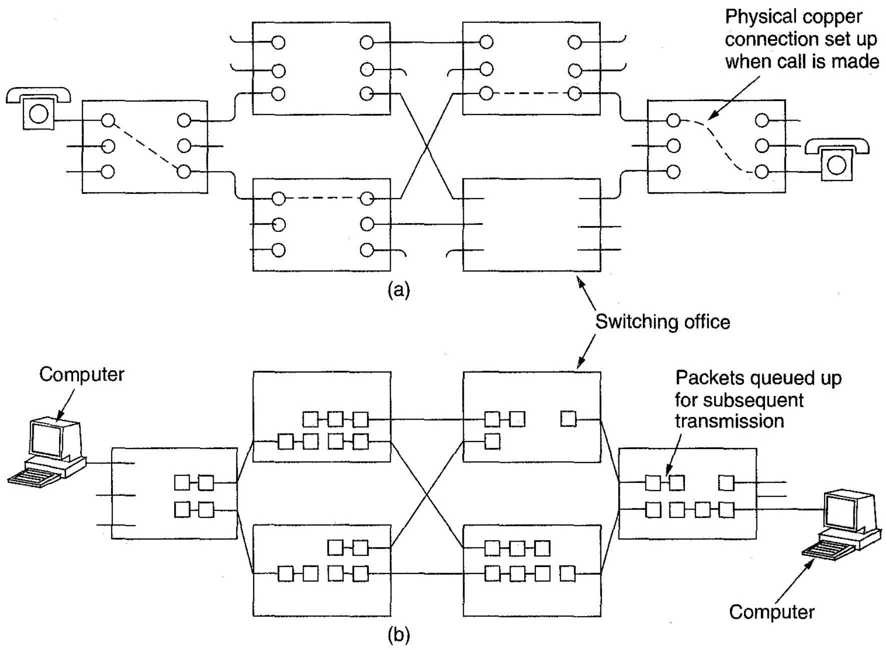

Fig. 2-34. (a) Circuit switching. (b) Packet switching.

An important property of circuit switching is the need to set up an end-to-end path *before* any data can be sent. The elapsed time between the end of dialing and the start of ringing can easily be 10 sec, more on long-distance or international calls. During this time interval, the telephone system is hunting for a copper path, as shown in Fig. 2-35(a). Note that before data transmission can even begin, the call request signal must propagate all the way to the destination, and be

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acknowledged. For many computer applications (e.g., point-of-sale credit verification), long setup times are undesirable.

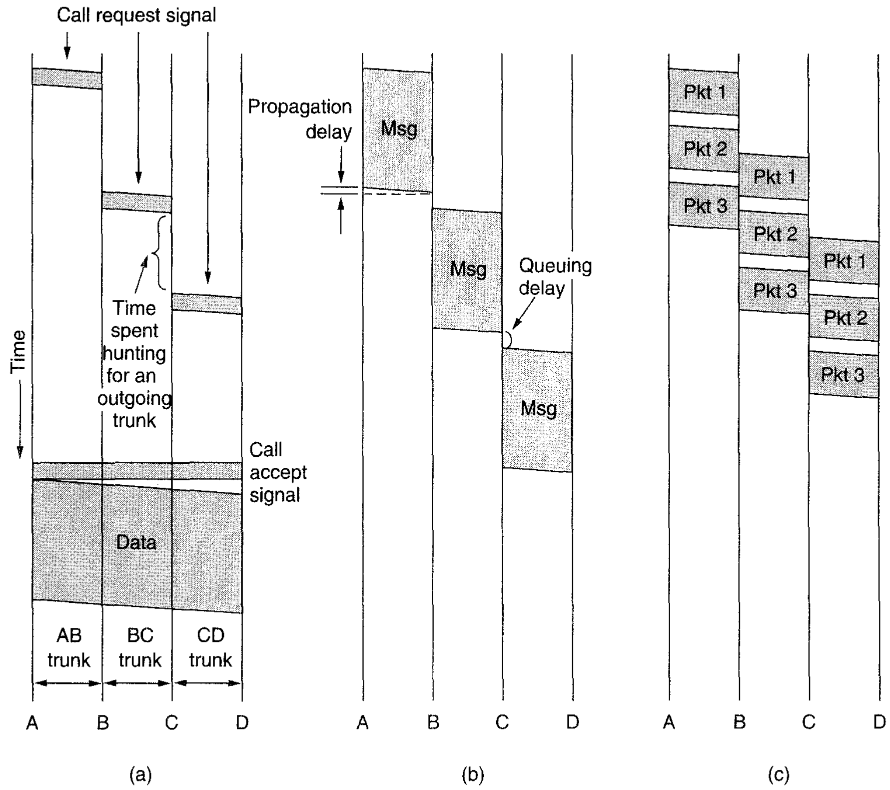

Fig. 2-35. Timing of events in (a) circuit switching, (b) message switching, (c) packet switching.

As a consequence of the copper path between the calling parties, once the setup has been completed, the only delay for data is the propagation time for the electromagnetic signal, about 5 msec per 1000 km. Also as a consequence of the established path, there is no danger of congestion—that is, once the call has been put through, you never get busy signals, although you might get one before the connection has been established due to lack of switching or trunk capacity.

An alternative switching strategy is **message switching**, shown in Fig. 2-35(b). When this form of switching is used, no physical copper path is established in advance between sender and receiver. Instead, when the sender has a block of data to be sent, it is stored in the first switching office (i.e., router) and then forwarded later, one hop at a time. Each block is received in its entirety, inspected for errors, and then retransmitted. A network using this technique is called a **store-and-forward** network, as mentioned in Chap. 1.

The first electromechanical telecommunication systems used message switching, namely for telegrams. The message was punched on paper tape off-line at the sending office, and then read in and transmitted over a communication line to the next office along the way, where it was punched out on paper tape. An operator there tore the tape off and read it in on one of the many tape readers, one per outgoing trunk. Such a switching office was called a **torn tape office**.

With message switching, there is no limit on block size, which means that routers (in a modern system) must have disks to buffer long blocks. It also means that a single block may tie up a router-router line for minutes, rendering message switching useless for interactive traffic. To get around these problems, **packet switching** was invented. Packet-switching networks place a tight upper limit on block size, allowing packets to be buffered in router main memory instead of on disk. By making sure that no user can monopolize any transmission line very long (milliseconds), packet-switching networks are well suited to handling interactive traffic. A further advantage of packet switching over message switching is shown in Fig. 2-35(b) and (c): the first packet of a multipacket message can be forwarded before the second one has fully arrived, reducing delay and improving throughput. For these reasons, computer networks are usually packet switched, occasionally circuit switched, but never message switched.

Circuit switching and packet switching differ in many respects. The key difference is that circuit switching statically reserves the required bandwidth in advance, whereas packet switching acquires and releases it as it is needed. With circuit switching, any unused bandwidth on an allocated circuit is just wasted. With packet switching it may be utilized by other packets from unrelated sources going to unrelated destinations, because circuits are never dedicated. However, just because no circuits are dedicated, a sudden surge of input traffic may overwhelm a router, exceeding its storage capacity and causing it to lose packets.

In contrast, with circuit switching, when packet switching is used, it is straightforward for the routers to provide speed and code conversion. Also, they can provide error correction to some extent. In some packet-switched networks, however, packets may be delivered in the wrong order to the destination. Reordering of packets can never happen with circuit switching.

Another difference is that circuit switching is completely transparent. The sender and receiver can use any bit rate, format, or framing method they want to. The carrier does not know or care. With packet switching, the carrier determines the basic parameters. A rough analogy is a road versus a railroad. In the former, the user determines the size, speed, and nature of the vehicle; in the latter, the carrier does. It is this transparency that allows voice, data, and fax to coexist within the phone system.

A final difference between circuit and packet switching is the charging algorithm. Packet carriers usually base their charge on both the number of bytes (or packets) carried and the connect time. Furthermore, transmission distance usually does not matter, except perhaps internationally. With circuit switching, the charge is based on the distance and time only, not the traffic. The differences are summarized in Fig. 2-36.

Item	Circuit-switched	Packet-switched
Dedicated "copper" path	Yes	No
Bandwidth available	Fixed	Dynamic
Potentially wasted bandwidth	Yes	No
Store-and-forward transmission	No	Yes
Each packet follows the same route	Yes	No
Call setup	Required	Not needed
When can congestion occur	At setup time	On every packet
Charging	Per minute	Per packet

Fig. 2-36. A comparison of circuit-switched and packet-switched networks.

Both circuit switching and packet switching are so important, we will come back to them shortly and describe the various technologies used in detail.

The Switch Hierarchy

It is worth saying a few words about how the routing between switches is done within the current circuit-switched telephone system. We will describe the AT&T system here, but other companies and countries use the same general principles. The telephone system has five classes of switching offices, as illustrated in Fig. 2-37. There are 10 regional switching offices, and these are fully interconnected by 45 high-bandwidth fiber optic trunks. Below the regional offices are 67 sectional offices, 230 primary offices, 1300 toll offices, and 19,000 end offices. The lower four levels were originally connected as a tree.

Calls are generally connected at the lowest possible level. Thus if a subscriber connected to end office 1 calls another subscriber connected to end office 1, the call will be completed in that office. However, a call from a customer attached to end office 1 in Fig. 2-37 to a customer attached to end office 2 will have to go toll office 1. However, a call from end office 1 to end office 4 will have to go up to primary office 1, and so on. With a pure tree, there is only one minimal route, and that would normally be taken.

During years of operation, the telephone companies noticed that some routes were busier than others. For example, there were many calls from New York to Los Angeles. Rather than go all the way up the hierarchy, they simply installed **direct trunks** for the busy routes. A few of these are shown in Fig. 2-37 as

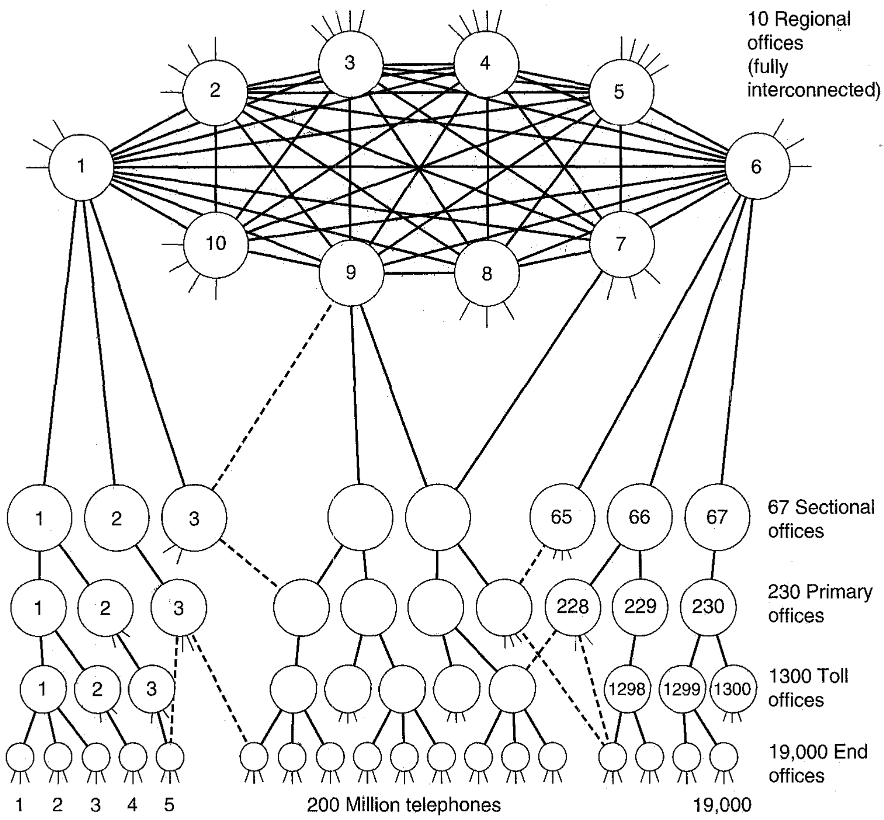

Fig. 2-37. The AT&T telephone hierarchy. The dashed lines are direct trunks.

dashed lines. As a consequence, many calls can now be routed along many paths. The actual route chosen is generally the most direct one, but if the necessary trunks along it are full, an alternative is chosen. This complex routing is now possible because a switching machine, like the AT&T 5 ESS, is in fact just a general purpose computer with a large amount of very specialized I/O equipment.

Crossbar Switches

Let us now turn from how calls are routed among switches to how individual switches actually work inside. Several kinds of switches are (or were) common within the telephone system. The simplest kind is the **crossbar switch** (also called a **crosspoint switch**), shown in Fig. 2-38. In a switch with n input lines and n output lines (i.e., n full duplex lines), the crossbar switch has n^2

intersections, called **crosspoints**, where an input and an output line may be connected by a semiconductor switch, as shown in Fig. 2-38(a). In Fig. 2-38(b) we see an example in which line 0 is connected to line 4, line 1 is connected to line 7, and line 2 is connected to line 6. Lines 3 and 5 are not connected. All the bits that arrive at the switch from line 4, for example, are immediately sent out of the switch on line 0. Thus the crossbar switch implements circuit switching by making a direct electrical connection, just like the jumper cables in the first-generation switches, only automatically and within microseconds.

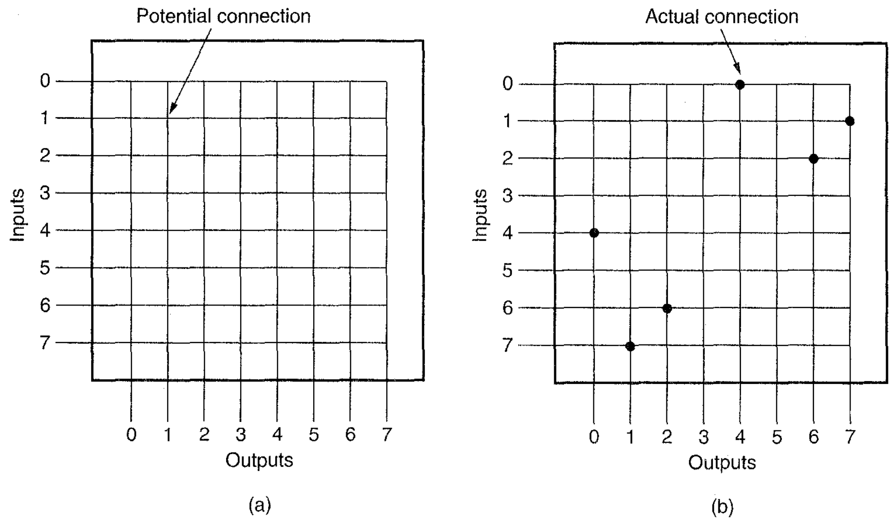

Fig. 2-38. (a) A crossbar switch with no connections. (b) A crossbar switch with three connections set up: 0 with 4, 1 with 7, and 2 with 6.

The problem with a crossbar switch is that the number of crossbars grows as the square of the number of lines into the switch. If we assume that all lines are full duplex and that there are no self-connections, only the crosspoints above the diagonal are needed. Still, $n(n - 1)/2$ crosspoints are needed. For $n = 1000$, we need 499,500 crosspoints. While building a VLSI chip with this number of transistor switches is possible, having 1000 pins on the chip is not. Thus a single crossbar switch is only useful for relatively small end offices.

Space Division Switches

By splitting the crossbar switch into small chunks and interconnecting them, it is possible to build multistage switches with many fewer crosspoints. These are called **space division switches**. Two configurations are illustrated in Fig. 2-39.

To keep our example simple, we will consider only three-stage switches, but

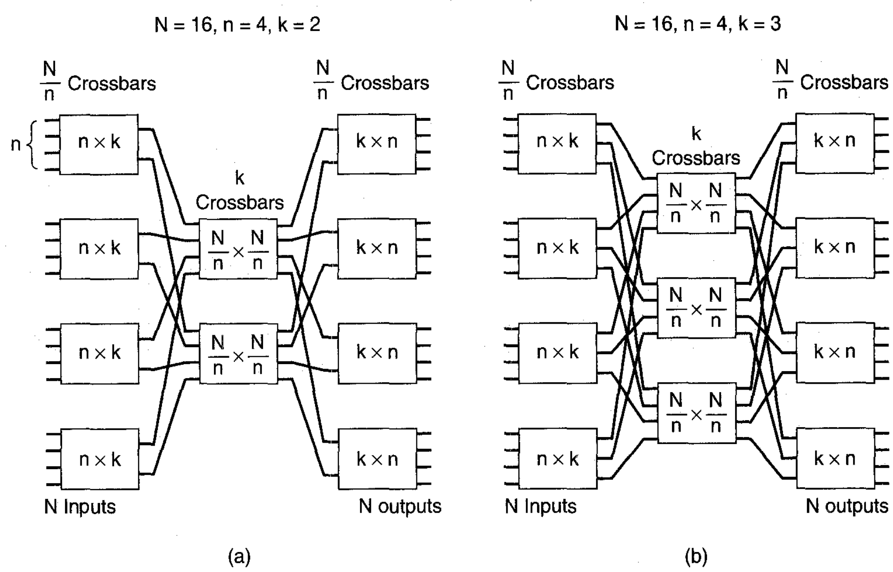

Fig. 2-39. Two space division switches with different parameters.

switches with more stages are also possible. In these examples, we have a total of N inputs and N outputs. Instead of building a single $N \times N$ crossbar, we build the switch out of smaller rectangular crossbars. In the first stage, each crossbar has n inputs, so we need N/n of them to handle all N incoming lines.

The second stage has k crossbars, each with N/n inputs and N/n outputs. The third stage is a repeat of the first stage, but reversed left to right. Each intermediate crossbar is connected to each input crossbar and each output crossbar. Consequently, it is possible to connect every input to every output using either the first intermediate crossbar in Fig. 2-39(a) or using the second one. In fact, there are two disjoint paths from each input to each output, depending which intermediate crossbar is chosen. In Fig. 2-39(b) there are three paths for each input/output pair. With k intermediate stages (k is a design parameter), there are k disjoint paths.

Let us now compute the number of crosspoints needed for a three-stage switch. In the first stage, there are N/n crossbars, each with nk crosspoints, for a total of Nk. In the second stage, there are k crossbars, each with $(N/n)^2$ crosspoints. The third stage is the same as the first. Adding up the three stages, we get

$$\text{Number of crosspoints} = 2kN + k(N/n)^2$$

For $N = 1000$, $n = 50$ and $k = 10$, we need only 24,000 crosspoints instead of the 499,500 required by a 1000×1000 single-stage crossbar.

Unfortunately, as usual, there is no free lunch. The switch can block. Consider Fig. 2-39(a) again. Stage 2 has eight inputs, so a maximum of eight calls can be connected at once. When call nine comes by, it will have to get a busy signal, even though the destination is available. The switch of Fig. 2-39(b) is better, handling a maximum of 12 calls instead of 8, but it uses more crosspoints. Sometimes when making a phone call you may have gotten a busy signal before you finished dialing. This was probably caused by blocking part way through the network.

It should be obvious that the larger k is, the more expensive the switch and the lower the blocking probability. In 1953, Clos showed that when $k = 2n - 1$, the switch will never block (Clos, 1953). Other researchers have analyzed calling patterns in great detail to construct switches that theoretically can block but do so only rarely in practice.

Time Division Switches

A completely different kind of switch is the **time division switch**, shown in Fig. 2-40. With time division switching, the n input lines are scanned in sequence to build up an input frame with n slots. Each slot has k bits. For T1 switches, the slots are 8 bits, with 8000 frames processed per second.

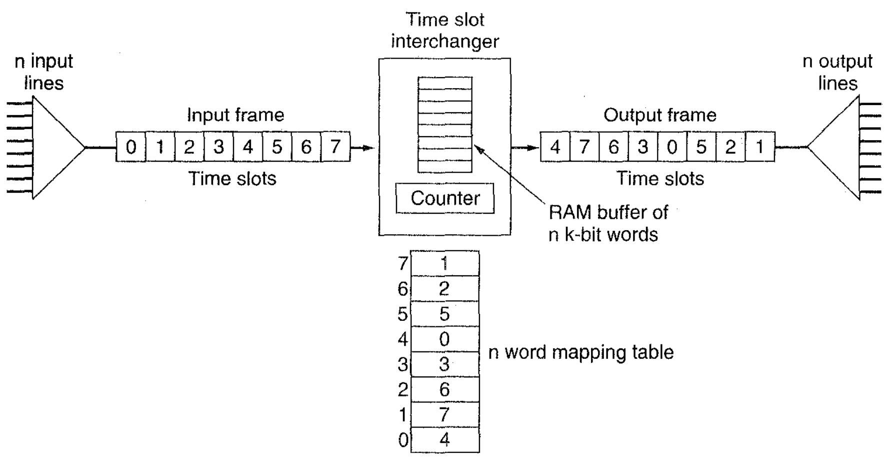

Fig. 2-40. A time division switch.

The heart of the time division switch is the **time slot interchanger**, which accepts input frames and produces output frames in which the time slots have been reordered. In Fig. 2-40, input slot 4 is output first, then slot 7, and so on. Finally, the output frame is demultiplexed, with output slot 0 (input slot 4) going to line 0, and so on. In essence, the switch has moved a byte from input line 4 to output line 0, another byte from input line 7 to output line 1, and so on. Viewed from the outside, the whole arrangement is a circuit switch, even though there are no physical connections.

The time slot interchanger works as follows: When an input frame is ready to be processed, each slot (i.e., each byte in the input frame) is written into a RAM buffer inside the interchanger. The slots are written in order, so buffer word i contains slot i.

After all the slots of the input frame have been stored in the buffer, the output frame is constructed by reading out the words again, but in a different order. A counter goes from 0 to $n - 1$. At step j, the contents of word j of a mapping table is read out and used to address the RAM table. Thus if word 0 of the mapping table contains a 4, word 4 of the RAM buffer will be read out first, and the first slot of the output frame will be slot 4 of the input frame. Thus the contents of the mapping table determine which permutation of the input frame will be generated as the output frame, and thus which input line is connected to which output line.

Time division switches use tables that are linear in the number of lines, rather than quadratic, but they have another limitation. It is necessary to store n slots in the buffer RAM and then read them out again within one frame period of 125 μsec. If each of these memory accesses takes T microsec, the time needed to process a frame is $2nT$ microsec, so we have $2nT = 125$ or $n = 125/2T$. For a memory with 100-nsec cycle time, we can support at most 625 lines. We can also turn this relation around and use it to determine the required memory cycle to support a given number of lines. As with a crossbar switch, it is possible to devise multistage switches that split the work up into several parts and then combine the results in order to handle larger numbers of lines.

2.5. NARROWBAND ISDN

For more than a century, the primary international telecommunication infrastructure has been the public circuit-switched telephone system. This system was designed for analog voice transmission and is inadequate for modern communication needs. Anticipating considerable user demand for an end-to-end digital service (i.e., not like Fig. 2-17 which is part digital and part analog), the world's telephone companies and PTTs got together in 1984 under the auspices of CCITT and agreed to build a new, fully digital, circuit-switched telephone system by the early part of the 21st Century. This new system, called **ISDN** (**Integrated Services Digital Network**), has as its primary goal the integration of voice and nonvoice services. It is already available in many locations and its use is growing slowly. In the following sections we will describe what it does and how it works. For further information, see (Dagdeviren et al., 1994; and Kessler, 1993).

2.5.1. ISDN Services

The key ISDN service will continue to be voice, although many enhanced features will be added. For example, many corporate managers have an intercom button on their telephone that rings their secretaries instantly (no call setup time). One ISDN feature is telephones with multiple buttons for instant call setup to arbitrary telephones anywhere in the world. Another feature is telephones that display the caller's telephone number, name, and address on a display while ringing. A more sophisticated version of this feature allows the telephone to be connected to a computer, so that the caller's database record is displayed on the screen as the call comes in. For example, a stockbroker could arrange that when she answers the telephone, the caller's portfolio is already on the screen along with the current prices of all the caller's stocks. Other advanced voice services include call forwarding and conference calls worldwide.

Advanced nonvoice services are remote electricity meter reading, and on-line medical, burglar, and smoke alarms that automatically call the hospital, police, or fire department, respectively, and give their address to speed up response.

2.5.2. ISDN System Architecture

It is now time to look at the ISDN architecture in detail, particularly the customer's equipment and the interface between the customer and the telephone company or PTT. The key idea behind ISDN is that of the **digital bit pipe**, a conceptual pipe between the customer and the carrier through which bits flow. Whether the bits originated from a digital telephone, a digital terminal, a digital facsimile machine, or some other device is irrelevant. All that matters is that bits can flow through the pipe in both directions.

The digital bit pipe can, and normally does, support multiple independent channels by time division multiplexing of the bit stream. The exact format of the bit stream and its multiplexing is a carefully defined part of the interface specification for the digital bit pipe. Two principal standards for the bit pipe have been developed, a low bandwidth standard for home use and a higher bandwidth standard for business use that supports multiple channels that are identical to the home use channel. Furthermore, businesses may have multiple bit pipes if they need additional capacity beyond what the standard business pipe can provide.

In Fig. 2-41(a) we see the normal configuration for a home or small business. The carrier places a network terminating device, **NT1**, on the customer's premises and connects it to the ISDN exchange in the carrier's office, several kilometers away, using the twisted pair that was previously used to connect to the telephone. The NT1 box has a connector on it into which a passive bus cable can be inserted. Up to eight ISDN telephones, terminals, alarms, and other devices can be connected to the cable, similar to the way devices are connected to a LAN. From the customer's point of view, the network boundary is the connector on NT1.

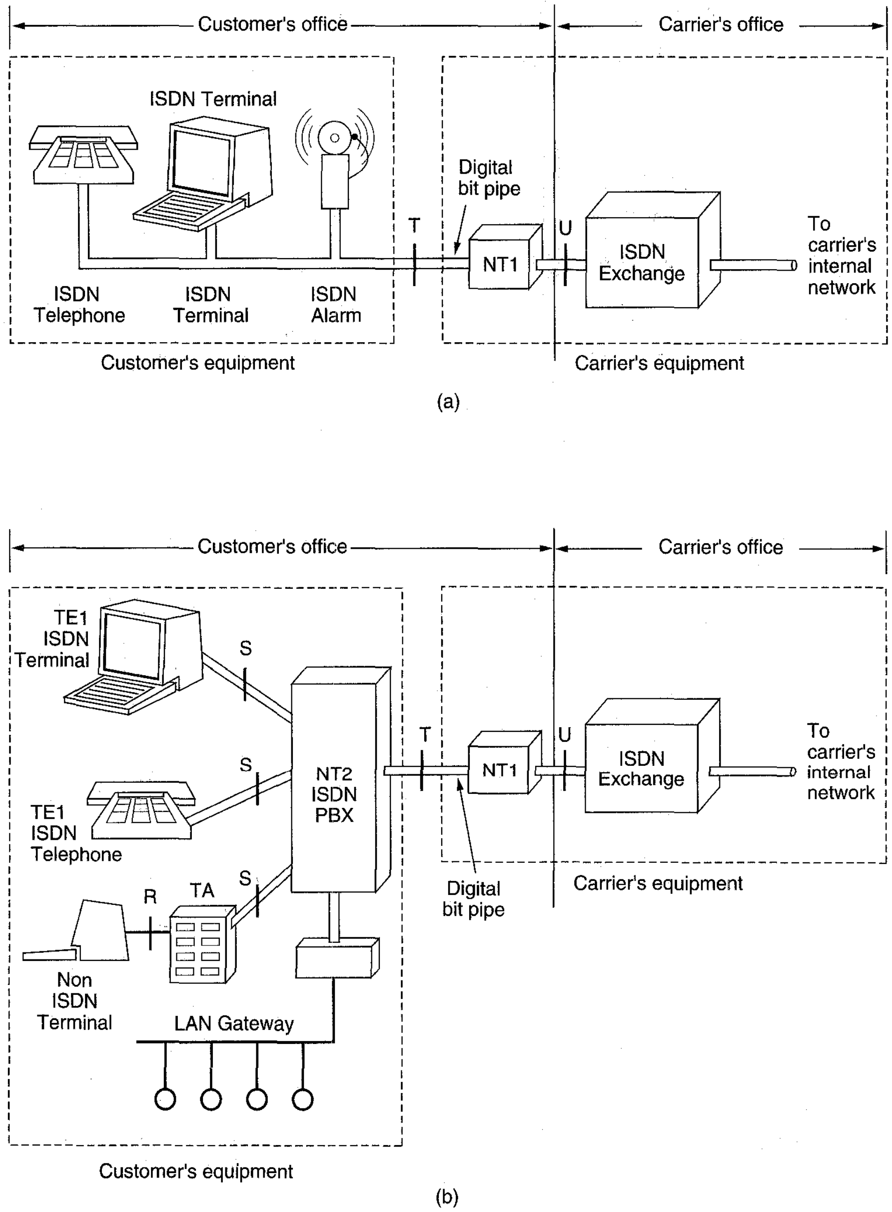

Fig. 2-41. (a) Example ISDN system for home use. (b) Example ISDN system with a PBX for use in large businesses.

For large businesses, the model of Fig. 2-41(a) is inadequate because it is common to have more telephone conversations going on simultaneously than the bus can handle. Therefore, the model of Fig. 2-41(b) is used. In this model we find a device, **NT2**, called a **PBX** (**Private Branch eXchange**), connected to NT1 and providing the real interface for telephones, terminals and other equipment. An ISDN PBX is not very different conceptually from an ISDN switch, although it is usually smaller and cannot handle as many conversations at the same time.

CCITT defined four **reference points**, called **R**, **S**, **T**, and **U**, between the various devices. These are marked in Fig. 2-41. The U reference point is the connection between the ISDN exchange in the carrier's office and NT1. At present it is a two-wire copper twisted pair, but at some time in the future it may be replaced by fiber optics. The T reference point is what the connector on NT1 provides to the customer. The S reference point is the interface between the ISDN PBX and the ISDN terminals. The R reference point is the connection between the terminal adapter and non-ISDN terminals. Many different kinds of interfaces will be used at R.

2.5.3. The ISDN Interface

The ISDN bit pipe supports multiple channels interleaved by time division multiplexing. Several channel types have been standardized:

A - 4-kHz analog telephone channel
B - 64-kbps digital PCM channel for voice or data
C - 8- or-16 kbps digital channel
D - 16-kbps digital channel for out-of-band signaling
E - 64-kbps digital channel for internal ISDN signaling
H - 384-, 1536-, or 1920-kbps digital channel

It was not CCITT's intention to allow an arbitrary combination of channels on the digital bit pipe. Three combinations have been standardized so far:

1. **Basic rate**: 2B + 1D
2. **Primary rate**: 23B + 1D (U.S. and Japan) or 30B + 1D (Europe)
3. **Hybrid**: 1A + 1C

The basic rate and primary rate channels are illustrated in Fig. 2-42.

The basic rate should be viewed as a replacement for **POTS** (**Plain Old Telephone Service**) for home or small business use. Each of the 64-kbps B channels can handle a single PCM voice channel with 8-bit samples made 8000 times a second (note that 64 kbps means 64,000 here, not 65,536). Signaling is on a separate 16-kbps D channel, so the full 64 kbps are available to the user (as in the CCITT 2.048-Mbps system and unlike the U.S. and Japanese T1 system).

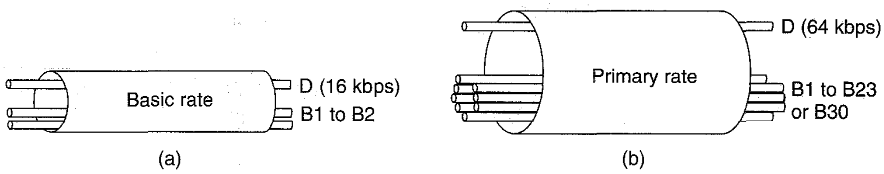

Fig. 2-42. (a) Basic rate digital pipe. (b) Primary rate digital pipe.

Because ISDN is so focused on 64-kbps channels, we refer to it as **N-ISDN** (**Narrowband ISDN**), to contrast it with broadband ISDN (ATM) to be discussed later.

The primary rate interface is intended for use at the T reference point for businesses with a PBX. It has 23 B channels and 1 D channel (at 64 kbps) in the United States and Japan and 30 B channels and 1 D channel (at 64 kbps) in Europe. The 23B + 1D choice was made to allow an ISDN frame fit nicely on AT&T's T1 system. The 30B + 1D choice was made to allow an ISDN frame fit nicely in CCITT's 2.048 Mbps system. The 32nd time slot in the CCITT system is used for framing and general network maintenance. Note that the amount of D channel per B channel in the primary rate is much less than in the basic rate, as it is not expected that there will be much telemetry or low bandwidth packet data there.

2.5.4. Perspective on N-ISDN

N-ISDN was a massive attempt to replace the analog telephone system with a digital one suitable for both voice and nonvoice traffic. Achieving worldwide agreement on the interface standard for the basic rate was supposed to lead to a large user demand for ISDN equipment, thus leading to mass production, economies of scale, and inexpensive VLSI ISDN chips. Unfortunately, the standardization process took years and the technology in this area moved very rapidly, so that once the standard was finally agreed upon, it was obsolete.

For home use, the largest demand for new services will undoubtedly be for video on demand. Unfortunately, the ISDN basic rate lacks the necessary bandwidth by two orders of magnitude. For business use, the situation is even bleaker. Currently available LANs offer at least 10 Mbps and are now being replaced by 100-Mbps LANs. Offering 64-kbps service to businesses in the 1980s was a serious proposition. In the 1990s, it is a joke.

Oddly enough, ISDN may yet be saved, but by a totally unexpected application: Internet access. Various companies now sell ISDN adaptors that combine the 2B + D channels into a single 144-kbps digital channel. Many Internet service providers also support these adaptors. The result is that people can access the

Internet over a 144-kbps fully digital link, instead of a 28.8-kbps analog modem link. For many Internet users, gaining a factor of five for downloading World Wide Web pages full of graphics is a service worth having. While B-ISDN at 155 Mbps is even better, N-ISDN at 144 kbps is here now for an affordable price, and that may be its main niche for the next few years.

2.6. BROADBAND ISDN AND ATM

When CCITT finally figured out that narrowband ISDN was not going to set the world on fire, it tried to think of a new service that might. The result was **broadband ISDN** (**B-ISDN**), basically a digital virtual circuit for moving fixed-size packets (cells) from source to destination at 155 Mbps (really 156 Mbps, as mentioned earlier). Since this data rate is even enough for (uncompressed) HDTV, it is likely to satisfy even the biggest bandwidth hogs for at least a few years.

Whereas narrowband ISDN was a timid first step into the digital age, broadband ISDN is a bold leap into the unknown. The benefits are enormous, such as a bandwidth increase over narrowband ISDN by a factor of 2500, but the challenges are equally huge (Armbruster, 1995).

To start with, broadband ISDN is based on ATM technology, and as we discussed briefly in Chap. 1, ATM is fundamentally a packet-switching technology, not a circuit-switching technology (although it can emulate circuit switching fairly well). In contrast, both the existing PSTN and narrowband ISDN are circuit-switching technologies. An enormous amount of engineering experience in circuit switching will be rendered obsolete by this change. Going from circuit switching to packet switching is truly a paradigm shift.

As if that were not enough, broadband ISDN cannot be sent over existing twisted pair wiring for any substantial distance. This means that introducing it will require ripping out most of the local loops and putting in either category 5 twisted pair or fiber (Stephens and Banwell, 1995). Furthermore, space division and time division switches cannot be used for packet switching. They will all have to be replaced by new switches based on different principles and running at much higher speeds. The only things that can be salvaged are the wide area fiber trunks.

In short, throwing out 100 years' accumulated knowledge plus an investment in both inside plant and outside plant worth many hundreds of billions of dollars is not exactly a small step to be taken lightly. Nevertheless, it is clear to the telephone companies that if they do not do it, the cable television companies, thinking about video on demand, probably will. While it is likely that both the existing PSTN and narrowband ISDN will be around for a decade or perhaps even longer, the long-term future probably lies with ATM, so we will study it in great detail in this book, starting with the physical layer in this chapter.

2.6.1. Virtual Circuits versus Circuit Switching

The basic broadband ISDN service is a compromise between pure circuit switching and pure packet switching. The actual service offered is connection oriented, but it is implemented internally with packet switching, not circuit switching. Two kinds of connections are offered: permanent virtual circuits and switched virtual circuits. **Permanent virtual circuits** are requested by the customer manually (e.g., by sending a fax to the carrier) and typically remain in place for months or years. **Switched virtual circuits** are like telephone calls: they are set up dynamically as needed and potentially torn down immediately afterward.

In a circuit-switching network, making a connection actually means a physical path is established from the source to the destination through the network, certainly when space division switches are used. (With time division switches, the concept of "a physical path" is already getting a little fuzzy around the edges.) In a virtual circuit network, like ATM, when a circuit is established, what really happens is that the route is chosen from source to destination, and all the switches (i.e., routers) along the way make table entries so they can route any packets on that virtual circuit. They also have the opportunity to reserve resources for the new circuit. Figure 2-43 shows a single virtual circuit from host H_1 to host H_5 via switches (routers) A, E, C, and D.

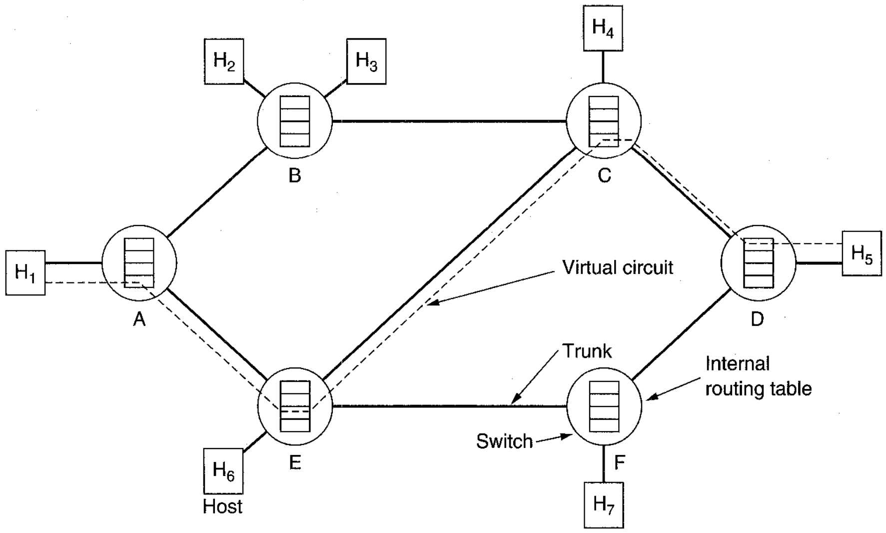

Fig. 2-43. The dotted line shows a virtual circuit. It is simply defined by table entries inside the switches.

When a packet comes along, the switch inspects the packet's header to find out which virtual circuit it belongs to. Then it looks up that virtual circuit in its tables to determine which communication line to send on. We will examine this process in more detail in Chap. 5.

The meaning of the permanent virtual circuit between H_1 and H_5 in Fig. 2-43 should now be clear. It is an agreement between the customer and the carrier that the switches will always hold table entries for a particular destination, even if there has been no traffic for months. Clearly, such an agreement costs resources (certainly table space inside the switches and possibly reserved bandwidth and buffers as well) so there is always a monthly charge per permanent virtual circuit. The advantage over a switched virtual circuit is that there is no setup time. Packets can move instantly. For some applications, such as credit card verification, saving a few seconds on each transaction may easily be worth the cost.

In contrast, a leased line from H_1 to H_5 in a circuit-switched network with the topology of Fig. 2-43 and space division switches would actually hold the crosspoints closed for months and would actually reserve bandwidth on the trunks permanently, either as FDM bands or as time slots (a leased "line" can be multihop if no direct line is available). Such an arrangement is obviously far more wasteful of resources when it is idle than a virtual circuit.

2.6.2. Transmission in ATM Networks

As we have pointed out before, ATM stands for *Asynchronous* Transfer Mode. This mode can be contrasted with the synchronous T1 carrier illustrated in Fig. 2-44(a). One T1 frame is generated precisely every 125 μsec. This rate is governed by a master clock. Slot k of each frame contains 1 byte of data from the same source. T1 is synchronous.

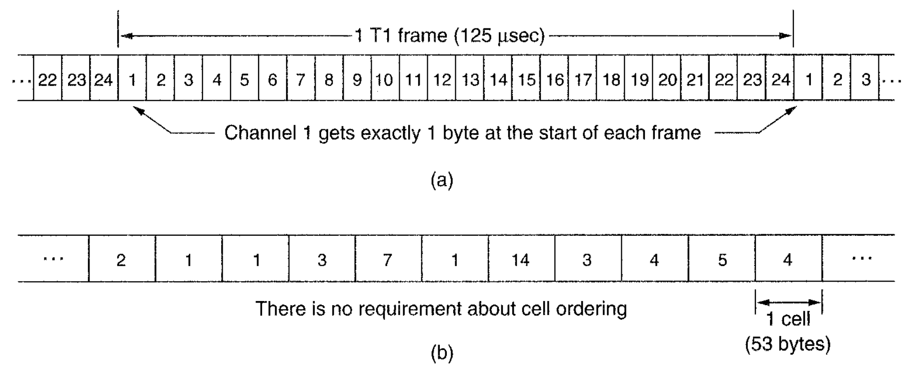

Fig. 2-44. (a) Synchronous transmission mode. (b) Asynchronous transmission mode.

ATM, in contrast, has no requirement that cells rigidly alternate among the various sources. Fig. 2-44(b) shows cells on a line from various sources, with no particular pattern. Cells arrive randomly from different sources.

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Furthermore, it is not even required that the stream of cells coming out of a computer be continuous. Gaps between the data cells are possible. Such gaps are filled by special idle cells.

ATM does not standardize the format for transmitting cells. Rather, it specifies that just sending individual cells is allowed but also specifies that cells may be encased in a carrier such as T1, T3, SONET, or FDDI (a fiber optic LAN). For these examples, standards exist telling how cells are packed into the frames these systems provide.

In the original ATM standard, the primary rate was 155.52 Mbps, with an additional rate at four times that speed (622.08 Mbps). These rates were chosen to be compatible with SONET, the framing standard used on fiber optic links throughout the telephone system. ATM over T3 (44.736 Mbps) and FDDI (100 Mbps) is also foreseen.

The transmission medium for ATM is normally fiber optics, but for runs under 100 meters, coax or category 5 twisted pair are also acceptable. Fiber runs can be many kilometers. Each link goes between a computer and an ATM switch, or between two ATM switches. In other words, all ATM links are point-to-point (unlike LANs, which have many senders and receivers on the same cable). Multicasting is achieved by having a cell enter a switch on one line and exit it on multiple lines. Each point-to-point link is unidirectional. For full-duplex operation, two parallel links are needed, one for traffic each way.

The ATM **Physical Medium Dependent** sublayer is concerned with getting the bits on and off the wire. Different hardware is needed for different cables and fibers, depending on the speed and line encoding. The purpose of the transmission convergence sublayer is to provide a uniform interface to the ATM layer in both directions. Outbound, the ATM layer provides a sequence of cells, and the PMD sublayer encodes them as necessary and pushes them out the door as a bit stream.

Inbound, the PMD sublayer takes the incoming bits from the network and delivers a bit stream to the TC sublayer. The cell boundaries are not marked in any way. It is up to the TC sublayer to somehow figure out how to tell where one cell ends and the next one begins. This job is not only difficult, it is theoretically impossible. Thus the TC sublayer clearly has its work cut out for it. Because the TC sublayer is doing cell framing, it is a data link function, so we will discuss it in Chap. 3. For additional information about the ATM physical layer, see (Rao and Hatamian, 1995).

2.6.3. ATM Switches

Many ATM cell switch designs have been described in the literature. Some of these have been implemented and tested. In this section we will give a brief introduction to the principles of ATM cell switch design and illustrate these with a few examples. For more information, see (De Prycker 1993; Garcia-Haro and

Jajszczyk, 1994; Handel et al., 1994; and Partridge, 1994). For an ATM switch optimized for running IP over ATM, see (Parulkar et al., 1995).

The general model for an ATM cell switch is shown in Fig. 2-45. It has some number of input lines and some number of output lines, almost always the same number (because the lines are bidirectional). ATM switches are generally synchronous in the sense of during a cycle, one cell is taken from each input line (if one is present), passed into the internal **switching fabric**, and eventually transmitted on the appropriate output line.

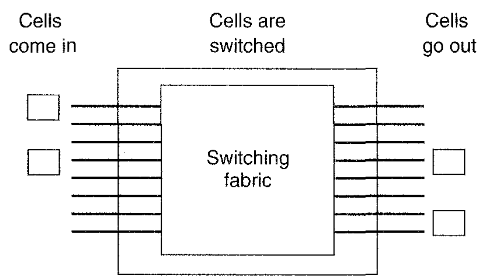

Fig. 2-45. A generic ATM switch.

Switches may be pipelined, that is, it may take several cycles before an incoming cell appears on its output line. Cells actually arrive on the input lines asynchronously, so there is a master clock that marks the beginning of a cycle. Any cell fully arrived when the clock ticks is eligible for switching during that cycle. A cell not fully arrived has to wait until the next cycle.

Cells arrive at ATM speed, normally about 150 Mbps. This works out to slightly over 360,000 cells/sec, which means that the cycle time of the switch has to be about 2.7 μsec. A commercial switch might have anywhere from 16 to 1024 input lines, which means that it must be prepared to accept and start switching a batch of 16 to 1024 cells every 2.7 μsec. At 622 Mbps, a new batch of cells is injected into the switching fabric about every 700 nsec. The fact that the cells are fixed length and short (53 bytes) makes it possible to build such switches. With longer variable-length packets, high-speed switching would be more complex, which is why ATM uses short fixed-length cells.

All ATM switches have two common goals:

1. Switch all cells with as low a discard rate as possible.

2. Never reorder the cells on a virtual circuit.

Goal 1 says that it is permitted to drop cells in emergencies, but that the loss rate should be as small as possible. A loss rate of 1 cell in 10^{12} is probably acceptable. On a large switch, this loss rate is about 1 or 2 cells per hour. Goal 2 says that cells arriving on a virtual circuit in a certain order must also depart in that order, with no exceptions, ever. This constraint makes switch design considerably more difficult, but it is required by the ATM standard.

A problem that occurs in all ATM switches is what to do if the cells arriving at two or more input lines want to go to the same output port in the same cycle. Solving this problem is one of the key issues in the design of all ATM switches. One nonsolution is to pick one cell to deliver and discard the rest. Since this algorithm violates goal 1, we cannot use it.

Our next attempt is to provide a queue for each input line. If two or more cells conflict, one of them is chosen for delivery, and the rest are held for the next cycle. The choice can be made at random, or cyclically, but should not exhibit systematic bias in favor of, for example, the lowest-numbered line to avoid giving lines with low numbers better service than lines with high numbers. Figure 2-46(a) depicts the situation at the start of cycle 1, in which cells have arrived on all four input lines, destined for output lines 2, 0, 2, and 1, respectively. Because there is a conflict for line 2, only one of the cells can be chosen. Suppose that it is the one on input line 0. At the start of cycle 1, shown in Fig. 2-46(b), three cells have been output, but the cell on line 2 has been held, and two more cells have arrived. Only at the start of cycle 4, shown in Fig. 2-46(d), have all the cells cleared the switch.

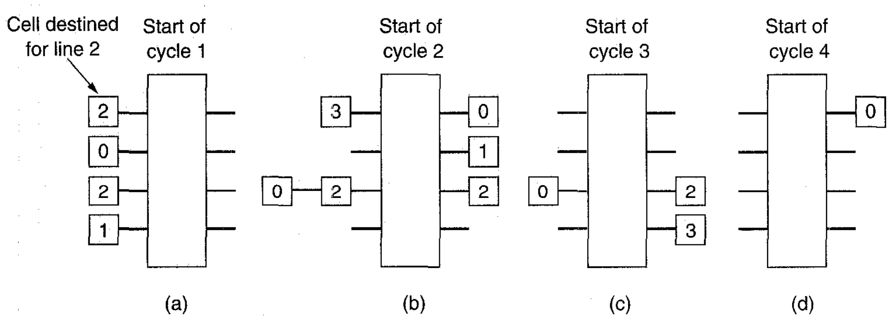

Fig. 2-46. Input queueing at an ATM switch.

The problem with input queueing is that when a cell has to be held up, it blocks the progress of any cells behind it, even if they could otherwise be switched. This effect is called **head-of-line blocking**. It is somewhat more complicated than shown here, since in a switch with 1024 input lines, conflicts may not be noticed until the cells are actually through the switch and fighting over the output line. Keeping a cell on its input queue until a signal comes back saying it made it through the switch requires extra logic, a reverse signaling path, and more delay. What is sometimes done is to put the losing cells on a recirculating bus that sends them back to the input side, but the switch has to be careful where it puts them, to avoid delivering cells from the same virtual circuit out of order.

An alternative design, one that does not exhibit head-of-line blocking does the queueing on the output side, as shown in Fig. 2-47. Here we have the same cell arrival pattern, but now when two cells want to go to the same output line in the same cycle, both are passed through the switch. One of them is put on the output line, and the other is queued on the output line, as shown in Fig. 2-47(b). Here it takes only three cycles, instead of four, to switch all the packets. Karol et al. (1987) have shown that output queueing is generally more efficient than input queueing.

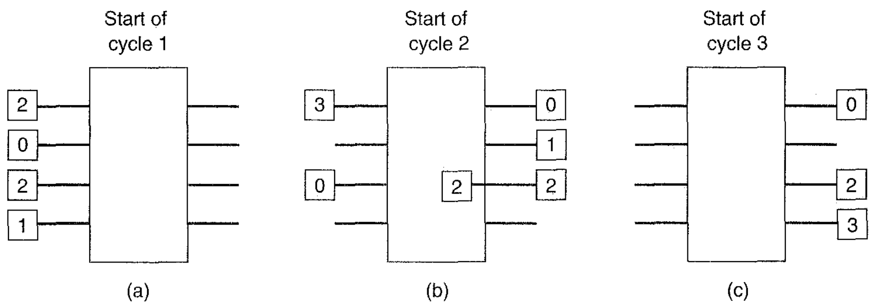

Fig. 2-47. Output queueing at an ATM switch.

The Knockout Switch

Let us now look more carefully at one ATM switch design that uses output queueing. It is called the **knockout switch** (Yeh et al., 1987) and is illustrated in Fig. 2-48 for eight input lines and eight output lines. Each input line is connected to a bus on which incoming cells are broadcast in the cycle they arrive. Having only one bus driver per bus simplifies the design and timing considerably.

For each arriving cell, hardware inspects the cell's header to find its virtual circuit information, looks that up in the routing tables (see Fig. 2-43), and enables the correct crosspoint. The cell then travels along its bus until it gets to the enabled crosspoint, at which time it heads south toward its output line. It is possible for multiple cells, in fact even all of them, to go to the same output line. It is also possible for a cell to be multicast to several output lines by just enabling several crosspoints on its broadcast bus.

The simplest way to handle collisions would be to simply buffer all cells at the output side. However, for a switch with 1024 input lines, in the worst case 1024 output buffers would be needed. In practice, this situation is very unlikely to occur, so a reasonable optimization is to provide far fewer output buffers, say n.

In the unlikely event that more cells arrive in one cycle than can be handled, the concentrator on each line selects out n cells for queueing, discarding the rest. The concentrator is a clever circuit for making this selection in a fair way, using

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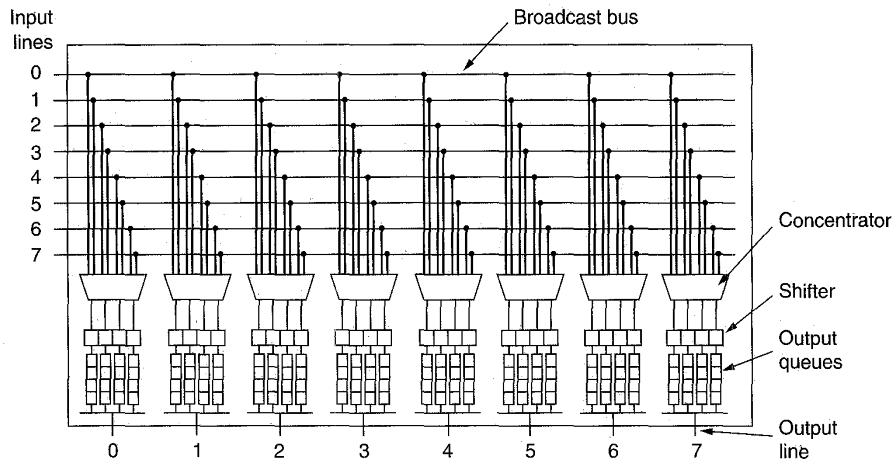

Fig. 2-48. A simplified diagram of the knockout switch.

an elimination (knockout) tournament similar to the quarter finals, semifinals, and finals in many sports tournaments.

Conceptually, all the selected cells go into a single output queue (unless it is full, in which case cells are discarded). However, actually getting all the cells into a single queue in the allotted time is not feasible, so the output queue is simulated by multiple queues. The selected cells go into a shifter, which then distributes them uniformly over n output queues using a token to keep track of which queue goes next, in order to maintain sequencing within each virtual circuit. By varying n, the designers can trade switch cost off against expected cell loss rate.

The Batcher-Banyan Switch

The problem with the knockout switch is that it is basically a crossbar switch, so the number of crosspoints is quadratic in the number of lines. Just as this factor proved to be a problem with circuit switching, it is also a problem with packet switching. The solution for circuit switching was space division switching, which vastly reduced the number of crosspoints, at the cost of requiring a multistage switch. A similar solution is available for packet switching.

This solution is called the **Batcher-banyan switch**. Like knockout switches, Batcher-banyan switches are synchronous, accepting a set of cells (zero or one per input line) on each cycle. Even a simple Batcher-banyan is more complicated than the space division switches of Fig. 2-39, so we will introduce it step by step. In Fig. 2-49(a) we have an 8×8 three-stage banyan switch, so called because its wiring is said to resemble the roots of a banyan tree. In all banyan switches, only

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one path exists from each input line to each output line. Routing is done by looking up the output line for each cell (based on the virtual circuit information and tables). This 3-bit binary number is then put in front of the cell, as it will be used for routing through the switch.

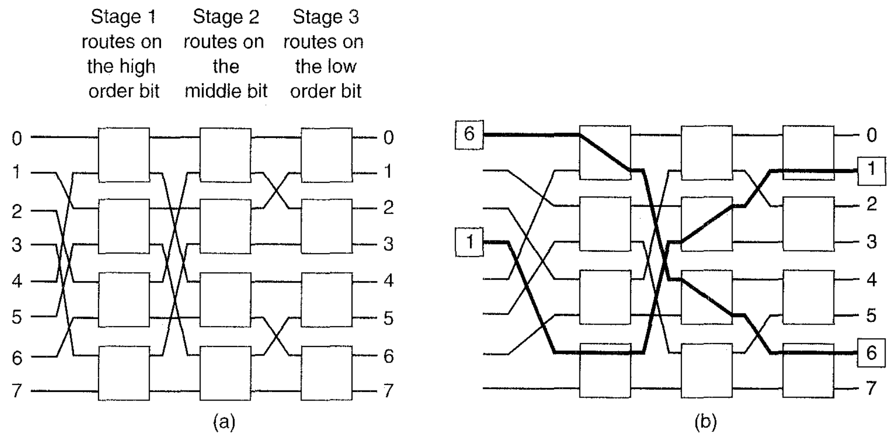

Fig. 2-49. (a) A banyan switch with eight input lines and eight output lines. (b) The routes that two cells take through the banyan switch.

Each of the 12 switching elements in the banyan switch has two inputs and two outputs. When a cell arrives at a switching element, 1 bit of the output line number is inspected, and based on that, the cell is routed either to port 0 (the upper one) or port 1 (the lower one). In the event of a collision, one cell is routed and one is discarded.

A banyan switch parses the output line number from left to right, so stage 1 examines the leftmost (i.e., high-order) bit, stage 2 examines the middle bit, and stage 3 examines the rightmost (i.e., low-order) bit. In Fig. 2-49(b) we have two cells present: a cell on input line 0 heading for output line 6, and a cell on input line 3 heading for output line 1. For the first cell, the binary output address is 110, so it passes through the three stages using the lower, lower, and upper ports, respectively, as shown. Similarly, the other cell, labeled 001 in binary, uses the upper, upper, and lower ports, respectively.

Unfortunately, a collision occurs in a banyan switch when two incoming cells want to exit a switching element via the same port at the same time. A series of such collisions is illustrated in Fig. 2-50(a). In stage 1, the collisions involve the cells heading for the following pairs of output lines: (5, 7), (0, 3), (6, 4), and (2, 1). Suppose that these collisions are resolved in favor of 5, 0, 4, and 1. In the second stage we get collisions between (0, 1) and (5, 4). Here we let 1 and 5 win, and they are then routed to the correct output lines.

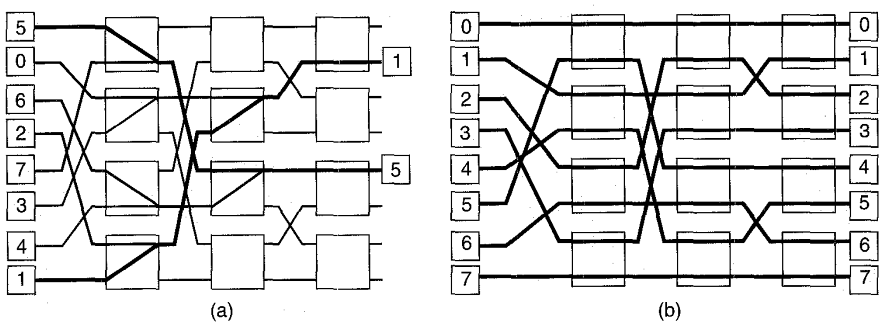

Fig. 2-50. (a) Cells colliding in a banyan switch. (b) Collision-free routing through a banyan switch.

Now look at Fig. 2-50(b). All eight cells get through with no collisions. The conclusion is: depending on the input, the banyan switch can do a good job or a bad job of routing.

The idea behind the Batcher-banyan switch is to put a switch in front of the banyan switch to permute the cells into a configuration that the banyan switch can handle without loss. For example, if the incoming cells are sorted by destination and presented on input lines 0, 2, 4, 6, 1, 3, 5, and 7, in that order as far as necessary (depending on how many cells there are), then the banyan switch does not lose cells.

To sort the incoming cells we can use a Batcher switch, invented by K.E. Batcher (1968). Like the banyan and knockout switches, it too is synchronous and works with discrete cycles. A Batcher switch is built up of 2×2 switching elements, but these work differently than those in the banyan switch. When a switching element receives two cells, it compares their output addresses numerically (thus not just 1 bit) and routes the higher one on the port in the direction of the arrow, and the lower one the other way. If there is only one cell, it goes to the port opposite the way the arrow is pointing.

A Batcher switch for eight lines is depicted on the left in Fig. 2-51. Stage 1 sorts the incoming cells pairwise. The next two stages do a four-way merge. The final three stages do an eight-way merge. In general, for n lines, the complexity of a Batcher switch grows like $n\log^2 n$. When k cells are present on the input lines, the Batcher switch puts the cells in sort order on the first k output lines.

After exiting the Batcher switch, the cells undergo a shuffle and are then injected into a banyan switch. The final result is that every cell appears on the correct output line at the far end of the banyan switch.

An example of how the combined Batcher-banyan switching fabric works is given in Fig. 2-52. Here cells are present on input lines 2, 3, 4, and 5, headed for output lines 6, 5, 1, and 4, respectively. Initially, cells for 5 and 6 enter the same

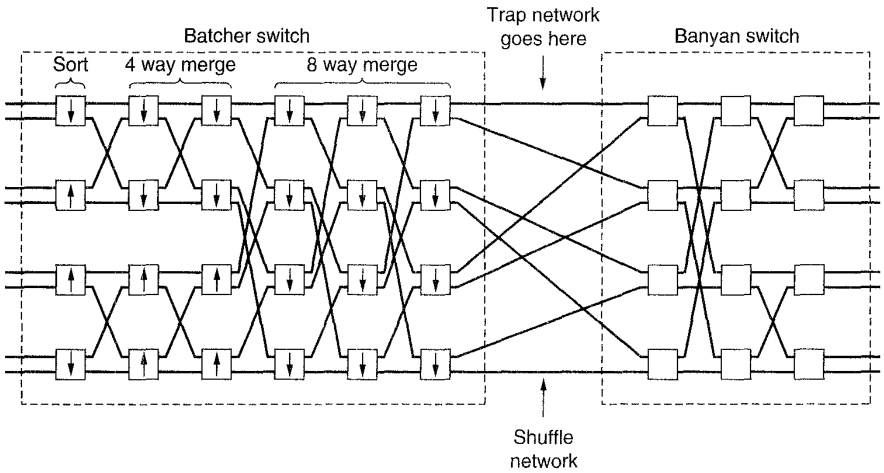

Fig. 2-51. The switching fabric for a Batcher-banyan switch.

switching element. Cell 6 has a higher address, so it exits in the direction of the arrow; cell 5 goes the other way. Here no exchange occurs. With cells 1 and 4, an exchange occurs, with cell 4 entering from the bottom but leaving from the top. The heavy lines show the paths all the way through to the end.

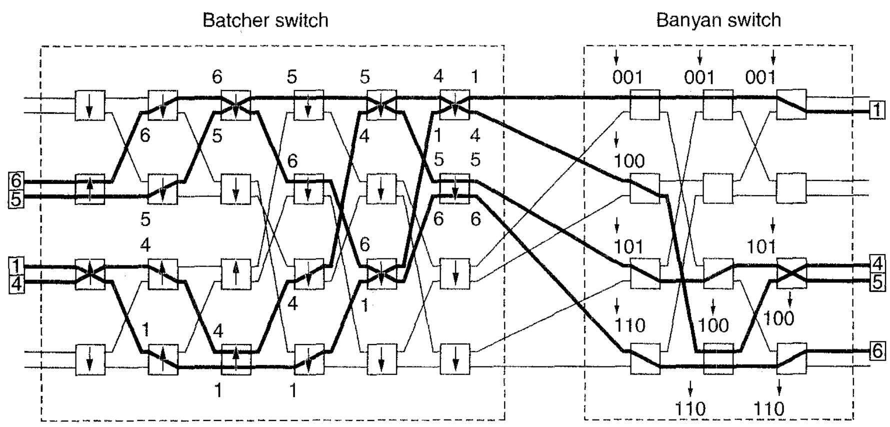

Fig. 2-52. An example with four cells using the Batcher-banyan switch.

Note that at the end of the Batcher switch, the four cells are stacked up at the top, in order. They are then run through a shuffle network and injected into the banyan switch, which is able to process them without collisions.

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In principle, the Batcher-banyan switch makes a fine ATM switch, but there are two complications that we have ignored: output line collisions and multicasting. If two or more cells are aimed at the same output line, the Batcher-banyan switch cannot handle them, so we have to go back to some kind of buffering. One way to solve this problem is by inserting a trap network between the Batcher switch and the banyan switch. The job of the trap network is to filter out duplicates and recirculate them for subsequent cycles, all the while maintaining the order of cells on a virtual circuit. (By now it should be clear that the ordering requirement is much more of a problem than it might have at first appeared.) Commercial switches also have to handle multicast.

The first Batcher-banyan ATM switch was designed by Huang and Knauer (1984). It was called Starlite. Then came Moonshine (Hui, 1987) and Sunshine (Giacopelli et al., 1991). You have to admit that these folks have a sense of humor. Starlite, Moonshine, and Sunshine differ primarily in the trap circuit and how they handle multicast.

2.7. CELLULAR RADIO

The traditional telephone system (even when broadband ISDN is fully operational) will still not be able to satisfy a growing group of users: people on the go. Consequently, there is increasing competition from a system that uses radio waves instead of wires and fibers for communication. This system will play an increasingly important role in the networking of notebook computers, shirt-pocket telephones, and personal digital assistants in the coming years. In the following sections we will examine satellite paging, cordless telephones, cellular telephones, and similar technologies. These systems are now merging, producing portable computers capable of sending and receiving phone calls, faxes, and email, as well as looking up information in remote databases, and doing this anywhere on earth.

Such devices are already creating a huge market. Many companies in the computer, telephone, satellite, and other industries want a piece of the action. The result is a chaotic market, with numerous overlapping and incompatible products and services, all rapidly changing, and typically different in every country as well. Nevertheless, the descriptions given below should provide at least a basic knowledge of the underlying technologies. For more information, see (Bates, 1994; Goodman, 1991; Macario, 1993; Padgett et al., 1995; and Seybold, 1994).

2.7.1. Paging Systems

The first paging systems used loudspeakers within a single building. In a hospital it is common to hear announcements on the public address system like: Will Dr. Suzanne Johnson please call extension 4321? Nowadays, people who want to be paged wear small beepers, usually with tiny screens for displaying short incoming messages.

A person wanting to page a beeper wearer can then call the beeper company and enter a security code, the beeper number, and the number the beeper wearer is to call (or another short message). The computer receiving the request then transmits it over land lines to a hilltop antenna, which either broadcasts the page directly (for local paging), or sends it to an overhead satellite (for long-distance paging), which then rebroadcasts it. When the beeper detects its unique number in the incoming radio stream, it beeps and displays the number to be called. It is also possible to page a group of people simultaneously with a single phone call.

The most advanced beeper systems plug directly into a computer and can receive not just a telephone number, but a longer message. The computer can then process the data as they come in. For example, a company could keep the price lists in its salespeoples' portable computers up to date using this form of paging.

Most current paging systems have the property that they are one-way systems, from a single computer out to a large number of receivers. There is no problem about who will speak next, and no contention among many competing users for a small number of channels as there is only one sender in the whole system.

Paging systems require little bandwidth since each message requires only a single burst of perhaps 30 bytes. At this data rate, a 1-Mbps satellite channel can handle over 240,000 pages per minute. The older paging systems run at various frequencies in the 150–174 MHz band. Most of the modern ones run in the 930–932 MHz band. Figure 2-53(a) shows the one-way nature of a paging system, with all communication being outbound at a single frequency. We will later see how this mode contrasts with mobile telephones, which are two way and use two frequencies per call, with different frequency pairs used for different calls, as depicted in Fig. 2-53(b). These differences make the paging system much simpler and less expensive to operate.

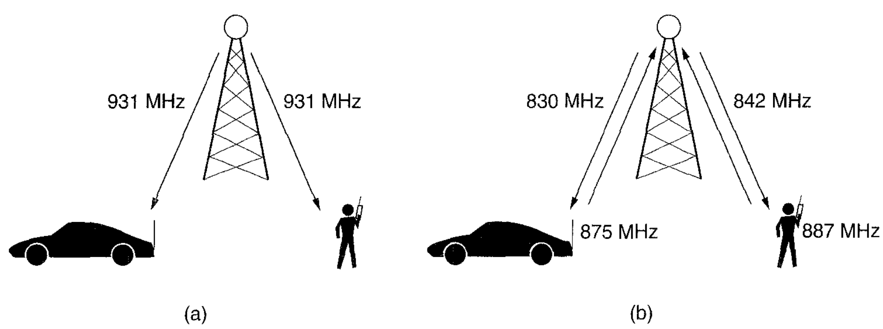

Fig. 2-53. (a) Paging systems are one way. (b) Mobile telephones are two way.

2.7.2. Cordless Telephones

Cordless telephones started as a way to allow people to walk around the house while on the phone. A cordless telephone consists of two parts: a base station and a telephone. These are always sold together. The base station has a standard phone jack at the back so it can be connected (by a wire) to the telephone system. The telephone communicates with the base station by low-power radio. The range is typically 100 to 300 meters.

Because early cordless telephones were only expected to communicate with their own base stations, there was no need for standardization. Some of the cheaper models used a fixed frequency, selected at the factory. If, by accident, your cordless phone happened to use the same frequency as your neighbor's, each of you could listen in on one another's calls. More expensive models avoided this problem by allowing the user to select the transmission frequency.

The first generation of cordless telephones, known as CT-1 in the United States and CEPT-1 in Europe, were entirely analog. They could, and often did, cause interference with radios and televisions. The poor reception and lack of security led the industry to develop a digital standard, CT-2, which originated in England. The first CT-2 devices could make calls, but not receive them, but as soon as the first one was sold, the manufacturer received some negative feedback and the system was quickly redesigned. Like the CT-1 version, each telephone had to be within a few hundred meters of its own base station, making it useful around the house or office, but useless in cars or when walking around town.

In 1992, a third generation, CT-3 or DECT, was introduced, which supported roaming over base stations. This technology is beginning to approach cellular telephones, which we will now describe.

2.7.3. Analog Cellular Telephones

Mobile radiotelephones were used sporadically for maritime and military communication during the early decades of the 20th Century. In 1946, the first system for car-based telephones was set up in St. Louis. This system used a single large transmitter on top of a tall building and had a single channel, used for both sending and receiving. To talk, the user had to push a button that enabled the transmitter and disabled the receiver. Such systems, known as **push-to-talk systems**, were installed in several cities beginning in the late 1950s. CB-radio, taxis, and police cars on television programs often use this technology.

In the 1960s, **IMTS** (**Improved Mobile Telephone System**) was installed. It, too, used a high-powered (200-watt) transmitter, on top of a hill, but now had two frequencies, one for sending and one for receiving, so the push-to-talk button was no longer needed. Since all communication from the mobile telephones went inbound on a different channel than the telephones listened to, the mobile users could not hear each other (unlike the push-to-talk system used in taxis).

IMTS supported 23 channels spread out from 150 MHz to 450 MHz. Due to the small number of channels, users often had to wait a long time before getting a dial tone. Also, due to the large power of the hilltop transmitter, adjacent systems had to be several hundred kilometers apart to avoid interference. All in all, the system was impractical due to the limited capacity.

Advanced Mobile Phone System

All that changed with **AMPS** (**Advanced Mobile Phone System**), invented by Bell Labs and first installed in the United States in 1982. It is also used in England, where it is called TACS, and in Japan, where it is called MCS-L1. In AMPS, a geographic region is divided up into **cells**, typically 10 to 20 km across, each using some set of frequencies. The key idea that gives AMPS far more capacity than all previous systems is using relatively small cells, and reusing transmission frequencies in nearby (but not adjacent) cells. Whereas an IMTS system 100 km across can have one call on each frequency, an AMPS system might have 100 10-km cells in the same area and be able to have 5 to 10 calls on each frequency, in widely separated cells. Furthermore, smaller cells mean less power is needed, which leads to smaller and cheaper devices. Hand-held telephones put out 0.6 watts; transmitters in cars are typically 3 watts, the maximum allowed by the FCC.

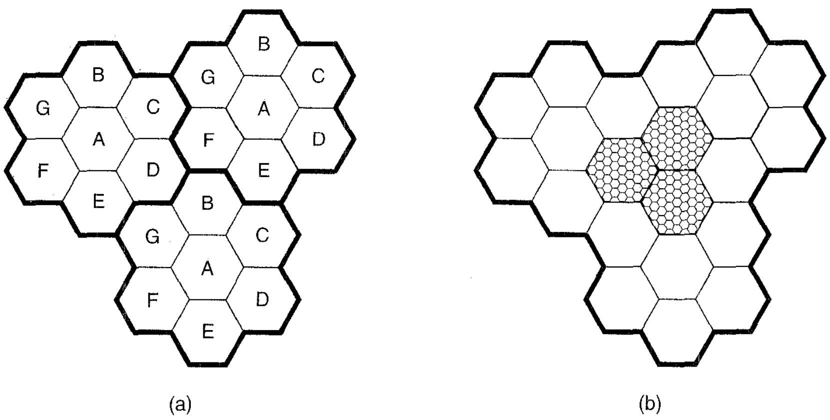

Fig. 2-54. (a) Frequencies are not reused in adjacent cells. (b) To add more users, smaller cells can be used.

The idea of frequency reuse is illustrated in Fig. 2-54(a). The cells are normally roughly circular, but they are easier to model as hexagons. In Fig. 2-54(a), the cells are all the same size. They are grouped together in units of seven cells. Each letter indicates a group of frequencies. Notice that for each frequency set,

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there is a buffer about two cells wide where that frequency is not reused, providing for good separation and low interference.

Finding locations high in the air to place base station antennas is a major issue. This problem has led some telecommunication carriers to forge alliances with the Roman Catholic Church, since the latter owns a substantial number of exalted potential antenna sites worldwide, all conveniently under a single management.

In an area where the number of users has grown to the point where the system is overloaded, the power is reduced and the overloaded cells are split into smaller cells to permit more frequency reuse, as shown in Fig. 2-54(b). How big the cells should be is a complex matter, and is treated in (Hac, 1995).

At the center of each cell is a base station to which all the telephones in the cell transmit. The base station consists of a computer and transmitter/receiver connected to an antenna. In a small system, all the base stations are connected to a single device called an **MTSO** (**Mobile Telephone Switching Office**) or **MSC** (**Mobile Switching Center**). In a larger one, several MTSOs may be needed, all of which are connected to a second-level MTSO, and so on. The MTSOs are essentially end offices as in the telephone system, and are, in fact, connected to at least one telephone system end office. The MTSOs communicate with the base stations, each other, and the PSTN using a packet switching network.

At any instant, each mobile telephone is logically in one specific cell and under the control of that cell's base station. When a mobile telephone leaves a cell, its base station notices the telephone's signal fading away and asks all the surrounding base stations how much power they are getting from it. The base station then transfers ownership to the cell getting the strongest signal, that is, the cell where the telephone is now located. The telephone is then informed of its new boss, and if a call is in progress, it will be asked to switch to a new channel (because the old one is not reused in any of the adjacent cells). This process is called **handoff** and takes about 300 msec. Channel assignment is done by the MTSO, which is the nerve center of the system. The base stations are really just radio relays.

Channels

The AMPS system uses 832 full-duplex channels, each consisting of a pair of simplex channels. There are 832 simplex transmission channels from 824 to 849 MHz, and 832 simplex receive channels from 869 to 894 MHz. Each of these simplex channels is 30 kHz wide. Thus AMPS uses FDM to separate the channels.

In the 800-MHz band, radio waves are about 40 cm long and travel in straight lines. They are absorbed by trees and plants and bounce off the ground and buildings. It is possible that a signal sent by a mobile telephone will reach the base station by the direct path, but also slightly later after bouncing off the ground or a building. This may lead to an echo effect or signal distortion. Sometimes, it is even possible to hear a distant conversation that has bounced several times.

In the United States, the 832 channels in each city are allocated by the FCC. Of these, half are assigned to the local telephone company, the **wireline carrier** or **B-side carrier**. The other half are assigned to a new entrant in the cellular business, the **A-side carrier**. The idea is to make sure there are at least two competing cellular suppliers, to promote competition and lower prices.

However, the distinction between a telephone company and a cellular phone company is now blurred, since most telephone companies have a cellular partner, and in 1994 AT&T merged with McCaw Cellular, the largest cellular operator. It frequently occurs that a company is an A-side carrier in some markets and a B-side carrier in others. Additional mixing occurs because a carrier may sell or trade any or all of its 416 channel licenses.

The 832 channels are divided into four categories:

1. Control (base to mobile) to manage the system.
2. Paging (base to mobile) to alert mobile users to calls for them.
3. Access (bidirectional) for call setup and channel assignment.
4. Data (bidirectional) for voice, fax, or data.

Twenty-one of the channels are reserved for control, and these are wired into a PROM in each telephone. Since the same frequencies cannot be reused in nearby cells, the actual number of voice channels available per cell is much smaller than 832, typically about 45.

Call Management

Each mobile telephone in AMPS has a 32-bit serial number and 10-digit telephone number in its PROM. The telephone number is represented as a 3-digit area code, in 10 bits, and a 7-digit subscriber number, in 24 bits. When a phone is switched on, it scans a preprogrammed list of 21 control channels to find the most powerful signal. Mobile phones are preset to scan for A-side only, B-side only, A-side preferred, or B-side preferred, depending on which service(s) the customer has subscribed to. From the control channel, it learns the numbers of the paging and access channels.

The phone then broadcasts its 32-bit serial number and 34-bit telephone number. Like all the control information in AMPS, this packet is sent in digital form, multiple times, and with an error-correcting code, even though the voice channels themselves are analog.

When the base station hears the announcement, it tells the MTSO, which records the existence of its new customer and also informs the customer's home

MTSO of his current location. During normal operation, the mobile telephone reregisters about once every 15 minutes.

To make a call, a mobile user switches on the phone, enters the number to be called on the keypad, and hits the SEND button. The phone then sends the number to be called and its own identity on the access channel. If a collision occurs there, it tries again later. When the base station gets the request, it informs the MTSO. If the caller is a customer of the MTSO's company (or one of its partners), the MTSO looks for an idle channel for the call. If one is found, the channel number is sent back on the control channel. The mobile phone then automatically switches to the selected voice channel and waits until the called party picks up the phone.

Incoming calls work differently. To start with, all idle phones continuously listen to the paging channel to detect messages directed at them. When a call is placed to a mobile phone (either from a fixed phone or another mobile phone), a packet is sent to the callee's home MTSO to find out where it is. A packet is then sent to the base station in its current cell, which then sends a broadcast on the paging channel of the form: "Unit 14, are you there?" The called phone then responds with "Yes" on the control channel. The base then says something like: "Unit 14, call for you on channel 3." At this point, the called phone switches to channel 3 and starts making ringing sounds.

Security Issues

Analog cellular phones are totally insecure. Anyone with an all-band radio receiver (scanner) can tune in and hear everything going on in a cell. Princess Di and her lover were once caught this way, which resulted in worldwide headlines. Since most cellular users do not realize how insecure the system is, they often give out credit card numbers and other once-confidential information this way.

Another major problem is theft of air time. With an all-band receiver attached to a computer, a thief can monitor the control channel and record the 32-bit serial number and 34-bit telephone numbers of all the mobile telephones it hears. By just driving around for a couple of hours, he can build up a large database. The thief can then pick a number and use it for his calls. This trick will work until the victim gets the bill, weeks later, at which time the thief just picks a new number.

Some thieves offer a low-cost telephone service by making calls for their customers using stolen numbers. Others reprogram mobile telephones with stolen numbers and sell them as phones that can make free calls.

Some of these problems could be solved by encryption, but then the police could not easily perform "wiretaps" on wireless criminals. This subject is very controversial and is discussed in more detail in Chap. 7.

Another issue in the general area of security is vandalism and damage to antennas and base stations. All these problems are quite severe and add up to hundreds of millions of dollars a year in losses for the cellular industry.

2.7.4. Digital Cellular Telephones

First generation cellular systems were analog. The second generation is digital. In the United States there was basically only one system: AMPS. When it was time for digital, three or four competitors emerged, and a struggle for survival began. It now appears that two systems will survive. The first one is backward compatible with the AMPS frequency allocation scheme and is specified in standards known as IS-54 and IS-135. The other is based on direct sequence spread spectrum and is specified in standard IS-95.

IS-54 is dual mode (analog and digital) and uses the same 30-kHz channels that AMPs does. It packs 48.6 kbps in each channel and shares it among three simultaneous users. Each user gets 13 kbps; the rest is control and timing overhead. Cells, base stations, and MTSOs work the same as in AMPS. Only the digital signaling and digital voice encoding is different. The IS-95 system is quite novel. We will discuss it when we get to channel allocation in Chap. 4.

In Europe, the reverse process happened. Five different analog systems were in use, in different countries, so someone with a British phone could not use it in France, and so on. This experience led the European PTTs to agree on a common digital system, called **GSM** (**Global Systems for Mobile communications**), which was deployed before any of the competing American systems. The Japanese system is different from all of the above.

Since the European systems were all different, it was simplest to make them pure digital operating in a new frequency band (1.8 GHz), in addition to retrofitting the 900-MHz band where possible. GSM uses both FDM and TDM. The available spectrum is broken up into 50 200-kHz bands. Within each band TDM is used to multiplex multiple users.

Some GSM telephones use smart cards, that is, credit card sized devices containing a CPU. The serial number and telephone number are contained there, not in the telephone, making for better physical security (stealing the phone without the card will not get you the number). Encryption is also used. We will discuss GSM in Chap. 4.

2.7.5. Personal Communications Services

The holy grail of the telephone world is a small cordless phone that you can use around the house and take with you anywhere in the world. It should respond to the same telephone number, no matter where it is, so people only have one telephone number (with AMPS, your home phone and your mobile phone have different numbers). This system is currently under vigorous development (Lipper and Rumsewicz, 1994). In the United States it is called **PCS** (**Personal Communications Services**). Everywhere else it is called **PCN** (**Personal Communications Network**). In the world of telephony, the United States has something of

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a tradition of marching to a different drummer than everyone else. Fortunately, most of the technical details are the same.

PCS will use cellular technology, but with microcells, perhaps 50 to 100 meters wide. This allows very low power (1/4 watt), which makes it possible to build very small, light phones. On the other hand, it requires many more cells than the 20-km AMPS cells. If we assume that a microcell is 1/200th the diameter of an AMPS cell, 40,000 times as many cells are required to cover the same area. Even if these microcells are much cheaper than AMPS cells, it is clear that building a complete PCS system from scratch will require a far more massive investment in infrastructure than did AMPS. Some telephone companies have realized that their telephone poles are excellent places to put the toaster-sized base stations, since the poles and wires already exist, thus greatly reducing the installation costs. These small base stations are sometimes called **telepoints**. How many to install and where to put them is a complicated issue (Steele et al., 1995a, 1995b).

The U.S. government (specifically, the FCC) is using PCS to make money out of thin air. In 1994–95 it auctioned off licenses to use the PCS spectrum (1.7 to 2.3 GHz). The auction raised 7.7 billion dollars for the government. This auction replaced the previous system of awarding frequency bands by lottery, thus eliminating the practice of companies with no interest in telecommunication entering the lottery. Any such company winning a frequency could instantly sell it to one of the losers for millions of dollars.

Unfortunately, there is no such thing as a free lunch, not even for the government. The 1.7- to 2.3-GHz band is already completely allocated to other users. These users will be given spectrum elsewhere and told to move there. The trouble is, antenna size depends on frequency, so this forced frequency reallocation will require a multibillion dollar investment in antennas, transmitters, etc. to be thrown away. Hordes of lobbyists are roaming around Washington with suggestions as to who should pay for all this. The net result is that PCS may not be widely deployed in this millenium. For a more rational way to deal with the spectrum, see (Youssef et al., 1995).

2.8. COMMUNICATION SATELLITES

In the 1950s and early 1960s, people tried to set up communication systems by bouncing signals off metallized weather balloons. Unfortunately, the received signals were too weak to be of any practical use. Then the U.S. Navy noticed a kind of permanent weather balloon in the sky—the moon—and built an operational system for ship-to-shore communication by bouncing signals off it.

Further progress in the celestial communication field had to wait until the first communication satellite was launched in 1962. The key difference between an artificial satellite and a real one is that the artificial one can amplify the signals before sending them back, turning a strange curiosity into a powerful communication system.

Communication satellites have some interesting properties that make them attractive for many applications. A communication satellite can be thought of as a big microwave repeater in the sky. It contains several **transponders**, each of which listens to some portion of the spectrum, amplifies the incoming signal, and then rebroadcasts it at another frequency, to avoid interference with the incoming signal. The downward beams can be broad, covering a substantial fraction of the earth's surface, or narrow, covering an area only hundreds of kilometers in diameter.

2.8.1. Geosynchronous Satellites

According to Kepler's law, the orbital period of a satellite varies as the orbital radius to the 3/2 power. Near the surface of the earth, the period is about 90 min. Communication satellites at such low altitudes are problematic because they are within sight of any given ground station for only a short time interval.

However, at an altitude of approximately 36,000 km above the equator, the satellite period is 24† hours, so it revolves at the same rate as the earth under it. An observer looking at a satellite in a circular equatorial orbit sees the satellite hang in a fixed spot in the sky, apparently motionless. Having the satellite be fixed in the sky is extremely desirable, because otherwise an expensive steerable antenna would be needed to track it.

With current technology, it is unwise to have satellites spaced much closer than 2 degrees in the 360-degree equatorial plane, to avoid interference. With a spacing of 2 degrees, there can only be 360/2 = 180 geosynchronous communication satellites in the sky at once. Some of these orbit slots are reserved for other classes of users (e.g., television broadcasting, government and military use, etc.).

Fortunately, satellites using different parts of the spectrum do not compete, so each of the 180 possible satellites could have several data streams going up and down simultaneously. Alternatively, two or more satellites could occupy one orbit slot if they operate at different frequencies.

To prevent total chaos in the sky, there have been international agreements about who may use which orbit slots and frequencies. The main commercial bands are listed in Fig. 2-55. The C band was the first to be designated for commercial satellite traffic. Two frequency ranges are assigned in it, the lower one for downlink traffic (from the satellite) and the upper one for uplink traffic (to the satellite). For a full-duplex connection one channel each way is required. These bands are already overcrowded because they are also used by the common carriers for terrestrial microwave links.

The next highest band available to commercial telecommunication carriers is

† For the purist, the rotation rate is the sidereal day: 23 hours 56 minutes 4.09 seconds.

Band	Frequencies	Downlink (GHz)	Uplink (GHz)	Problems
C	4/6	3.7–4.2	5.925–6.425	Terrestrial interference
Ku	11/14	11.7–12.2	14.0–14.5	Rain
Ka	20/30	17.7–21.7	27.5–30.5	Rain; equipment cost

Fig. 2-55. The principal satellite bands.

the Ku band. This band is not (yet) congested, and at these frequencies satellites can be spaced as close as 1 degree. However, another problem exists: rain. Water is an excellent absorber of these short microwaves. Fortunately, heavy storms are usually localized, so by using several widely separated ground stations instead of just one, the problem can be circumvented at the price of extra antennas, extra cables, and extra electronics to switch rapidly between stations. Bandwidth has also been allocated in the Ka band for commercial satellite traffic, but the equipment needed to use them is still expensive. In addition to these commercial bands, many government and military bands also exist.

A typical satellite has 12–20 transponders, each with a 36–50-MHz bandwidth. A 50-Mbps transponder can be used to encode a single 50-Mbps data stream, 800 64-kbps digital voice channels, or various other combinations. Furthermore, two transponders can use different polarizations of the signal, so they can use the same frequency range without interfering. In the earliest satellites, the division of the transponders into channels was static, by splitting the bandwidth up into fixed frequency bands (FDM). Nowadays, time division multiplexing is also used due to its greater flexibility.

The first satellites had a single spatial beam that illuminated the entire earth. With the enormous decline in the price, size, and power requirements of microelectronics, a much more sophisticated broadcasting strategy has become possible. Each satellite is equipped with multiple antennas and multiple transponders. Each downward beam can be focused on a small geographical area, so multiple upward and downward transmissions can take place simultaneously. These so-called **spot beams** are typically elliptically shaped, and can be as small as a few hundred km in diameter. A communication satellite for the United States would typically have one wide beam for the contiguous 48 states, plus spot beams for Alaska and Hawaii.

A new development in the communication satellite world is the development of low-cost microstations, sometimes called **VSATs** (**Very Small Aperture Terminals**) (Ivancic et al., 1994). These tiny terminals have 1-meter antennas and can put out about 1 watt of power. The uplink is generally good for 19.2 kbps, but the downlink is more, often 512 kbps. In many VSAT systems, the microstations do not have enough power to communicate directly with one another (via the satellite, of course). Instead, a special ground station, the **hub**, with a large, high-gain antenna is needed to relay traffic between VSATs, as shown in Fig. 2-56. In this mode of operation, either the sender or the receiver has a large antenna and a powerful amplifier. The trade-off is a longer delay in return for having cheaper end-user stations.

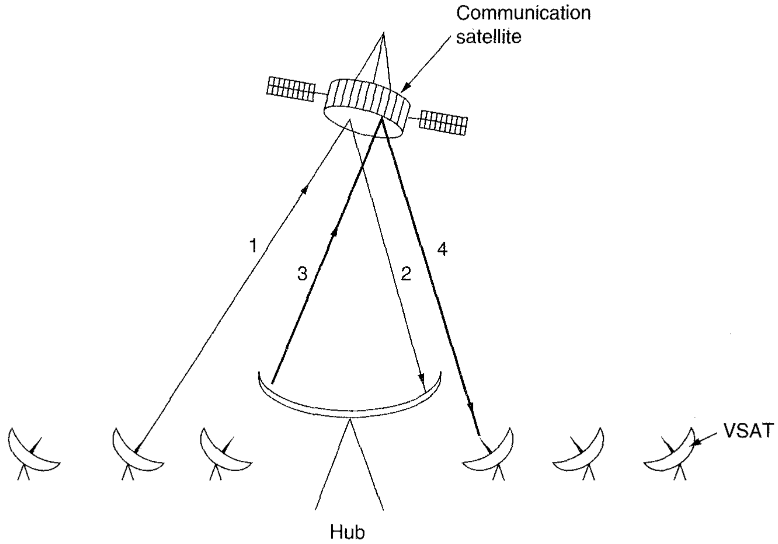

Fig. 2-56. VSATs using a hub.

Communication satellites have several properties that are radically different from terrestrial point-to-point links. To begin with, even though signals to and from a satellite travel at the speed of light (nearly 300,000 km/sec), the large round-trip distance introduces a substantial delay. Depending on the distance between the user and the ground station, and the elevation of the satellite above the horizon, the end-to-end transit time is between 250 and 300 msec. A typical value is 270 msec (540 msec for a VSAT system with a hub).

For comparison purposes, terrestrial microwave links have a propagation delay of roughly 3 μsec/km and coaxial cable or fiber optic links have a delay of approximately 5 μsec/km (electromagnetic signals travel faster in air than in solid materials).

Another important property of satellites is that they are inherently broadcast media. It does not cost more to send a message to thousands of stations within a transponder's footprint than it does to one. For some applications, this property is very useful. Even when broadcasting can be simulated using point-to-point line, satellite broadcasting may be much cheaper. On the other hand, from a security and privacy point of view, satellites are a complete disaster: everybody can hear everything. Encryption is essential when security is required.

Satellites also have the property that the cost of transmitting a message is independent of the distance traversed. A call across the ocean costs no more to service than a call across the street. Satellites also have excellent error rates and can be deployed almost instantly, a major consideration for military communication.

2.8.2. Low-Orbit Satellites

For the first 30 years of the satellite era, low-orbit satellites were rarely used for communication because they zip into and out of view so quickly. In 1990, Motorola broke new ground by filing an application with the FCC asking for permission to launch 77 low-orbit satellites for the Iridium project (element 77 is iridium). The plan was later revised to use only 66 satellites, so the project should have been renamed Dysprosium (element 66), but that probably sounded too much like a disease. The idea was that as soon as one satellite went out of view, another would replace it. This proposal set off a feeding frenzy among other communication companies. All of a sudden, everyone wanted to launch a chain of low-orbit satellites. We will briefly describe the Iridium system here, but the others are similar.

The basic goal of Iridium is to provide worldwide telecommunication service using hand-held devices that communicate directly with the Iridium satellites. It provides voice, data, paging, fax, and navigation service everywhere on earth. This service competes head-on with PCS/PCN and makes the latter unnecessary.

It uses ideas from cellular radio, but with a twist. Normally, the cells are fixed, but the users are mobile. Here, each satellite has a substantial number of spot beams that scan the earth as the satellite moves. Thus both the cells and the users are mobile in this system, but the handover techniques used for cellular radio are equally applicable to the case of the cell leaving the user as to the case of the user leaving the cell.

The satellites are to be positioned at an altitude of 750 km, in circular polar orbits. They would be arranged in north-south necklaces, with one satellite every 32 degrees of latitude. With six satellite necklaces, the entire earth would be covered, as suggested by Fig. 2-57(a). People not knowing much about chemistry can think of this arrangement as a very, very big dysprosium atom, with the earth as the nucleus and the satellites as the electrons.

Each satellite would have a maximum of 48 spot beams, with a total of 1628 cells over the surface of the earth, as shown in Fig. 2-57(b). Frequencies could be reused two cells away, as with conventional cellular radio. Each cell would have 174 full-duplex channels, for a total of 283,272 channels worldwide. Some of these would be for paging and navigation, which require hardly any bandwidth at all. (The paging devices envisioned would display two lines of alphanumeric text.)

The uplinks and downlinks would operate in the L band, at 1.6 GHz, making

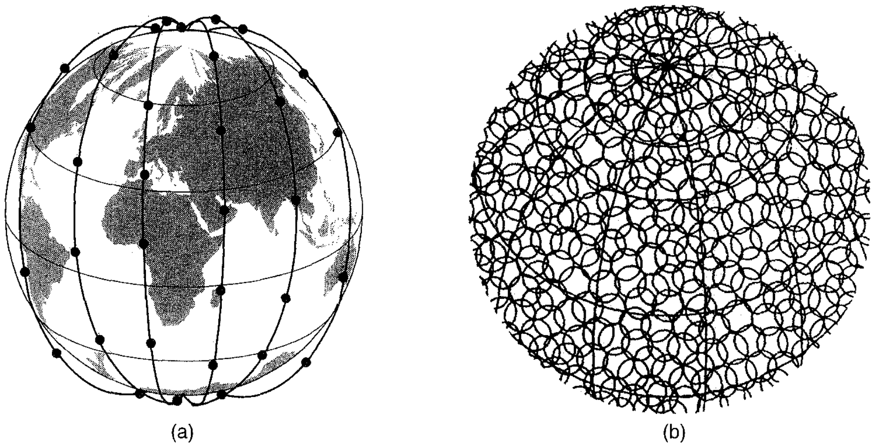

Fig. 2-57. (a) The Iridium satellites form six necklaces around the earth. (b) 1628 moving cells cover the earth.

it possible to communicate with a satellite using a small battery-powered device. Messages received by one satellite but destined for a remote one would be relayed among the satellites in the Ka band. Sufficient bandwidth is available in outer space for the intersatellite links. The limiting factor would be the uplink/downlink segments. Motorola estimates that 200 MHz would be sufficient for the whole system.

The projected cost to the end user is about 3 dollars per minute. If this technology can provide universal service anywhere on earth for that price, it is unlikely that the project will die for lack of customers. Business and other travelers who want to be in touch all the time, even in undeveloped areas, will sign up in droves. However, in developed areas, Iridium will face stiff competition from PCS/PCN with their toaster-on-a-pole telepoints.

2.8.3. Satellites versus Fiber

A comparison between satellite communication and terrestrial communication is instructive. As recently as 20 years ago, a case could be made that the future of communication lay with communication satellites. After all, the telephone system had changed little in the past 100 years and showed no signs of changing in the next 100 years. This glacial movement was caused in no small part by the regulatory environment in which the telephone companies were expected to provide good voice service at reasonable prices (which they did), and in return got a guaranteed profit on their investment. For people with data to transmit, 1200-bps modems were available. That was pretty much all there was.

The introduction of competition in 1984 in the United States and somewhat

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later in Europe changed all that radically. Telephone companies began replacing their long-haul networks with fiber and introduced high-bandwidth services like SMDS and B-ISDN. They also stopped their long-time practice of charging artificially high prices to long-distance users to subsidize local service.

All of a sudden, terrestrial fiber connections looked like the long-term winner. Nevertheless, communication satellites have some major niche markets that fiber does not (and sometimes, cannot) address. We will now look at a few of these.

While a single fiber has, in principle, more potential bandwidth than all the satellites ever launched, this bandwidth is not available to most users. The fibers that are now being installed are used within the telephone system to handle many long distance calls at once, not to provide individual users with high bandwidth. Furthermore, few users even have access to a fiber channel because the trusty old twisted pair local loop is in the way. Calling up the local telephone company end office at 28.8 kbps will never give more bandwidth than 28.8 kbps, no matter how wide the intermediate link is. With satellites, it is practical for a user to erect an antenna on the roof of the building and completely bypass the telephone system. For many users, bypassing the local loop is a substantial motivation.

For users who (sometimes) need 40 or 50 Mbps, an option is leasing a (44.736-Mbps) T3 carrier. However, this is an expensive undertaking. If that bandwidth is only needed intermittently, SMDS may be a suitable solution, but it is not available everywhere, and satellite service is.

A second niche is for mobile communication. Many people nowadays want to communicate while jogging, driving, sailing, and flying. Terrestrial fiber optic links are of no use to them, but satellite links potentially are. It is possible, however, that a combination of cellular radio and fiber will do an adequate job for most users (but probably not for those airborne or at sea).

A third niche is for situations in which broadcasting is essential. A message sent by satellite can be received by thousands of ground stations at once. For example, an organization transmitting a stream of stock, bond, or commodity prices to thousands of dealers might find a satellite system much cheaper than simulating broadcasting on the ground.

A fourth niche is for communication in places with hostile terrain or a poorly developed terrestrial infrastructure. Indonesia, for example, has its own satellite for domestic telephone traffic. Launching one satellite was much easier than stringing thousands of undersea cables among all the islands in the archipelago.

A fifth niche market for satellites is where obtaining the right of way for laying fiber is difficult or unduly expensive. Sixth, when rapid deployment is critical, as in military communication systems in time of war, satellites win easily.

In short, it looks like the mainstream communication of the future will be terrestrial fiber optics combined with cellular radio, but for some specialized uses, satellites are better. However, there is one caveat that applies to all of this: economics. Although fiber offers more bandwidth, it is certainly possible that terrestrial and satellite communication will compete aggressively on price. If advances in technology radically reduce the cost of deploying a satellite (e.g., some future space shuttle can toss out dozens of satellites on one launch), or low-orbit satellites catch on, it is not certain that fiber will win in all markets.

2.9. SUMMARY

The physical layer is the basis of all networks. Nature imposes two fundamental limits on al channels, and these determine their bandwidth. These limits are the Nyquist limit, which deals with noiseless channels, and the Shannon limit, for noisy channels.

Transmission media can be guided or unguided. The principle guided media are twisted pair, coaxial cable, and fiber optics. Unguided media include radio, microwaves, infrared, and lasers through the air.

A key element in most wide area networks is the telephone system. Its main components are the local loops, trunks, and switches. Local loops are analog, twisted pair circuits, which require modems for transmitting digital data. Trunks are digital, and can be multiplexed in several ways, including FDM, TDM, and WDM. The switches include crossbars, space division switches, and time division switches. Both circuit switching and packet switching are important.

In the future, the telephone system will be digital from end to end and will carry both voice and nonvoice traffic over the same lines. Two variants of this new system, known as ISDN, are being introduced. Narrowband ISDN is a circuit-switched digital system that is an incremental improvement over the current system. In contrast, broadband ISDN represents a paradigm shift, since it is based on cell switching ATM technology. Various kinds of ATM switches exist, including the knockout switch and the Batcher-banyan switch.

For mobile applications, the hard-wired telephone system is not suitable. Alternatives to the telephone system include cellular radio and communication satellites. Cellular radio is now widely used for portable telephones but will soon be common for data traffic as well. The current generation of cellular systems (e.g., AMPS) are analog, but the next generation (e.g., PCS/PCN) will be fully digital. Traditional communication satellites are geosynchronous, but there is now much interest in low-orbit satellite systems such as Iridium.

PROBLEMS

1. Compute the Fourier coefficients for the function $f(t) = t \quad (0 \le t \le 1)$.

2. A noiseless 4-kHz channel is sampled every 1 msec. What is the maximum data rate?

3. Television channels are 6 MHz wide. How many bits/sec can be sent if four-level digital signals are used? Assume a noiseless channel.

4. If a binary signal is sent over a 3-kHz channel whose signal-to-noise ratio is 20 dB, what is the maximum achievable data rate?

5. What signal-to-noise ratio is needed to put a T1 carrier on a 50-kHz line?

6. What is the difference between a passive star and an active repeater in a fiber optic network?

7. How much bandwidth is there in 0.1 micron of spectrum at a wavelength of 1 micron?

8. It is desired to send a sequence of computer screen images over an optical fiber. The screen is 480×640 pixels, each pixel being 24 bits. There are 60 screen images per second. How much bandwidth is needed, and how many microns of wavelength are needed for this band at 1.30 microns?

9. Is the Nyquist theorem true for optical fiber, or only for copper wire?

10. In Fig. 2-6 the lefthand band is narrower than the others. Why?

11. Radio antennas often work best when the diameter of the antenna is equal to the wavelength of the radio wave. Reasonable antennas range from 1 cm to 5 meters in diameter. What frequency range does this cover?

12. Multipath fading is maximized when the two beams arrive 180 degrees out of phase. How much of a path difference is required to maximize the fading for a 50 km long 1 GHz microwave link?

13. A laser beam 1 mm wide is aimed at a detector 1 mm wide 100 m away on the roof of a building. How much of an angular diversion (in degrees) does the laser have to have before it misses the detector?

14. A simple telephone system consists of two end offices and a single toll office to which each end office is connected by a 1-MHz full-duplex trunk. The average telephone is used to make four calls per 8-hour workday. The mean call duration is 6 min. Ten percent of the calls are long-distance (i.e., pass through the toll office). What is the maximum number of telephones an end office can support? (Assume 4 kHz per circuit.)

15. A regional telephone company has 10 million subscribers. Each of their telephones is connected to a central office by a copper twisted pair. The average length of these twisted pairs is 10 km. How much is the copper in the local loops worth? Assume that the cross section of each strand is a circle 1 mm in diameter, the specific gravity of copper is 9.0, and that copper sells for 3 dollars per kilogram.

16. The cost of a powerful microprocessor has dropped to the point where it is now possible to include one in each modem. How does that affect the handling of telephone line errors?

17. A modem constellation diagram similar to Fig. 2-19 has data points at the following coordinates: (1, 1), (1, −1), (−1, 1), and (−1, −1). How many bps can a modem with these parameters achieve at 1200 baud?

18. A modem constellation diagram similar to Fig. 2-19 has data points at (0, 1) and (0, 2). Does the modem use phase modulation or amplitude modulation?

19. Does FTTH fit into the telephone company model of end offices, toll offices, and so on, or does the model have to be changed in a fundamental way? Explain your answer.

20. At the low end, the telephone system is star shaped, with all the local loops in a neighborhood converging on an end office. In contrast, cable television consists of a single long cable snaking its way past all the houses in the same neighborhood. Suppose that a future TV cable were 10 Gbps fiber instead of copper. Could it be used to simulate the telephone model of everybody having their own private line to the end office? If so, how many one-telephone houses could be hooked up to a single fiber?

21. A cable TV system has 100 commercial channels, all of them alternating programs with advertising. Is this more like TDM or like FDM?

22. Why has the PCM sampling time been set at 125 μsec?

23. What is the percent overhead on a T1 carrier; that is, what percent of the 1.544 Mbps are not delivered to the end user?

24. Compare the maximum data rate of a noiseless 4-kHz channel using
(a) Analog encoding with 2 bits per sample.
(b) The T1 PCM system.

25. If a T1 carrier system slips and loses track of where it is, it tries to resynchronize using the 1st bit in each frame. How many frames will have to be inspected on the average to resynchronize with a probability of 0.001 of being wrong?

26. What is the difference, if any, between the demodulator part of a modem and the coder part of a codec? (After all, both convert analog signals to digital ones.)

27. A signal is transmitted digitally over a 4-kHz noiseless channel with one sample every 125 μsec. How many bits per second are actually sent for each of these encoding methods?
(a) CCITT 2.048 Mbps standard.
(b) DPCM with a 4-bit relative signal value.
(c) Delta modulation.

28. A pure sine wave of amplitude A is encoded using delta modulation, with x samples/sec. An output of +1 corresponds to a signal change of $+A/8$, and an output signal of −1 corresponds to a signal change of $-A/8$. What is the highest frequency that can be tracked without cumulative error?

29. SONET clocks have a drift rate of about 1 part in 10^9. How long does it take for the drift to equal the width of 1 bit? What are the implications of this calculation?

30. In Fig. 2-32, the user data rate for OC-3 is stated to be 148.608 Mbps. Show how this number can be derived from the SONET OC-3 parameters.

31. What is the available user bandwidth in an OC-12c connection?

32. Three packet-switching networks each contain n nodes. The first network has a star topology with a central switch, the second is a (bidirectional) ring, and the third is fully interconnected, with a wire from every node to every other node. What are the best, average, and worst case transmission paths in hops?

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33. Compare the delay in sending an x-bit message over a k-hop path in a circuit-switched network and in a (lightly loaded) packet-switched network. The circuit setup time is s sec, the propagation delay is d sec per hop, the packet size is p bits, and the data rate is b bps. Under what conditions does the packet network have a lower delay?

34. Suppose that x bits of user data are to be transmitted over a k-hop path in a packet-switched network as a series of packets, each containing p data bits and h header bits, with $x \gg p + h$. The bit rate of the lines is b bps and the propagation delay is negligible. What value of p minimizes the total delay?

35. How many crosspoints do the switches of Fig. 2-39(a) and Fig. 2-39(b) have? Compare this to a full 16×16 single-stage crossbar switch.

36. In the space division switch of Fig. 2-39(a), what is the smallest number of existing connections that can block a new outgoing call?

37. An alternative design to that of Fig. 2-39(a) is one in which the 16 lines are divided into two blocks of eight, instead of four blocks of four (i.e., $n = 8$ instead of $n = 4$). Such a design would save on hardware costs, since only two concentrators would be needed on the input and output sides. What is the strongest argument against this alternative?

38. How many lines can a time division switch handle if the RAM access time is 50 nsec?

39. How many bits of RAM buffer does a time switch interchanger need if the input line samples are 10 bits and there are 80 input lines?

40. Does time division switching necessarily introduce a minimum delay at each switching stage? If so, what is it?

41. How long does it take to transmit an 8 inch by 10 inch image by facsimile over an ISDN B channel? The facsimile digitizes the image into 300 pixels per inch and assigns 4 bits per pixel. Current FAX machines go faster than this over ordinary telephone lines. How do you think they do it?

42. Give an advantage and a disadvantage of NT12 (as opposed to NT1 and NT2) in an ISDN network.

43. In Fig. 2-50(a) we saw collisions between cells traveling through a banyan switch. These collisions occurred in the first and second stages. Can collisions also occur in the third stage? If so, under what conditions?

44. For this problem you are to route some cells through a Batcher-banyan ATM switch step by step. Four cells are present on input lines 0 through 3, headed for 3, 5, 2, and 1 respectively. For each of the six stages in the Batcher switch and the four steps in the banyan switch (including the input and output), list which cells are there as an eight-tuple (cell on line 0, cell on line 1, and so on). Indicate lines with no cell by –.

45. Now repeat the previous problem starting from (7, –, 6, –, 5, – 4, –).

46. An ATM switch has 1024 input lines and 1024 output lines. The lines operate at the SONET rate of 622 Mbps, which gives a user rate of approximately 594 Mbps. What aggregate bandwidth does the switch need to handle the load? How many cells per second must it be able to process?

47. In a typical cellular telephone system with hexagonal cells, it is forbidden to reuse a frequency band in an adjacent cell. If a total of 840 frequencies are available, how many can be used in a given cell?

48. Make a rough estimate of the number of PCS microcells 100 m in diameter it would take to cover San Francisco (120 square km).

49. Sometimes when a cellular user crosses the boundary from one cell to another, the current call is abruptly terminated, even though all transmitters and receivers are functioning perfectly. Why?

50. The 66 low-orbit satellites in the Iridium project are divided into six necklaces around the earth. At the altitude they are using, the period is 90 minutes. What is the average interval for handoffs for a stationary transmitter?

3

THE DATA LINK LAYER

In this chapter we will study the design of layer 2, the data link layer. This study deals with the algorithms for achieving reliable, efficient communication between two adjacent machines at the data link layer. By adjacent, we mean that the two machines are physically connected by a communication channel that acts conceptually like a wire (e.g., a coaxial cable or a telephone line). The essential property of a channel that makes it "wire-like" is that the bits are delivered in exactly the same order in which they are sent.

At first you might think this problem is so trivial that there is no software to study—machine *A* just puts the bits on the wire, and machine *B* just takes them off. Unfortunately, communication circuits make errors occasionally. Furthermore, they have only a finite data rate, and there is a nonzero propagation delay between the time a bit is sent and the time it is received. These limitations have important implications for the efficiency of the data transfer. The protocols used for communications must take all these factors into consideration. These protocols are the subject of this chapter.

After an introduction to the key design issues present in the data link layer, we will start our study of its protocols by looking at the nature of errors, their causes, and how they can be detected and corrected. Then we will study a series of increasingly complex protocols, each one solving more and more of the problems present in this layer. Finally, we will conclude with an examination of protocol modeling and correctness and give some examples of data link protocols.

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3.1. DATA LINK LAYER DESIGN ISSUES

The data link layer has a number of specific functions to carry out. These functions include providing a well-defined service interface to the network layer, determining how the bits of the physical layer are grouped into frames, dealing with transmission errors, and regulating the flow of frames so that slow receivers are not swamped by fast senders. In the following sections we will examine each of these issues in turn.

3.1.1. Services Provided to the Network Layer

The function of the data link layer is to provide services to the network layer. The principal service is transferring data from the network layer on the source machine to the network layer on the destination machine. On the source machine there is an entity, call it a process, in the network layer that hands some bits to the data link layer for transmission to the destination. The job of the data link layer is to transmit the bits to the destination machine, so they can be handed over to the network layer there, as shown in Fig. 3-1(a). The actual transmission follows the path of Fig. 3-1(b), but it is easier to think in terms of two data link layer processes communicating using a data link protocol. For this reason, we will implicitly use the model of Fig. 3-1(a) throughout this chapter.

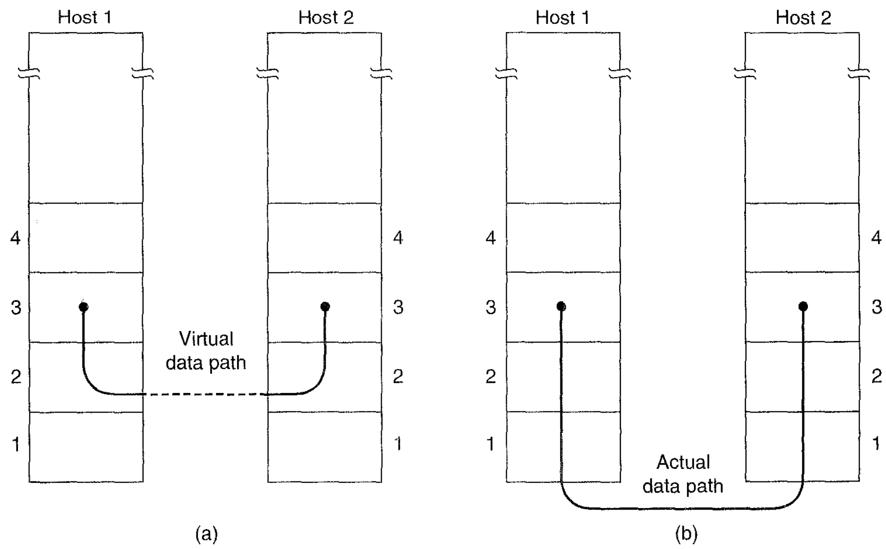

Fig. 3-1. (a) Virtual communication. (b) Actual communication.

The data link layer can be designed to offer various services. The actual

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services offered can vary from system to system. Three reasonable possibilities that are commonly provided are

1. Unacknowledged connectionless service.
2. Acknowledged connectionless service.
3. Acknowledged connection-oriented service.

Let us consider each of these in turn.

Unacknowledged connectionless service consists of having the source machine send independent frames to the destination machine without having the destination machine acknowledge them. No connection is established beforehand or released afterward. If a frame is lost due to noise on the line, no attempt is made to recover it in the data link layer. This class of service is appropriate when the error rate is very low so recovery is left to higher layers. It is also appropriate for real-time traffic, such as speech, in which late data are worse than bad data. Most LANs use unacknowledged connectionless service in the data link layer.

The next step up in terms of reliability is acknowledged connectionless service. When this service is offered, there are still no connections used, but each frame sent is individually acknowledged. In this way, the sender knows whether or not a frame has arrived safely. If it has not arrived within a specified time interval, it can be sent again. This service is useful over unreliable channels, such as wireless systems.

It is perhaps worth emphasizing that providing acknowledgements in the data link layer is just an optimization, never a requirement. The transport layer can always send a message and wait for it to be acknowledged. If the acknowledgement is not forthcoming before the timer goes off, the sender can just send the entire message again. The trouble with this strategy is that if the average message is broken up into, say, 10 frames, and 20 percent of all frames are lost, it may take a very long time for the message to get through. If individual frames are acknowledged and retransmitted, entire messages get through much faster. On reliable channels, such as fiber, the overhead of a heavyweight data link protocol may be unnecessary, but on wireless channels it is well worth the cost due to their inherent unreliability.

Getting back to our services, the most sophisticated service the data link layer can provide to the network layer is connection-oriented service. With this service, the source and destination machines establish a connection before any data are transferred. Each frame sent over the connection is numbered, and the data link layer guarantees that each frame sent is indeed received. Furthermore, it guarantees that each frame is received exactly once and that all frames are received in the right order. With connectionless service, in contrast, it is conceivable that a lost acknowledgement causes a frame to be sent several times and thus received several times. Connection-oriented service, in contrast, provides the network layer processes with the equivalent of a reliable bit stream.

When connection-oriented service is used, transfers have three distinct phases. In the first phase the connection is established by having both sides initialize variables and counters needed to keep track of which frames have been received and which ones have not. In the second phase, one or more frames are actually transmitted. In the third and final phase, the connection is released, freeing up the variables, buffers, and other resources used to maintain the connection.

Consider a typical example: a WAN subnet consisting of routers connected by point-to-point leased telephone lines. When a frame arrives at a router, the hardware verifies the checksum and passes the frame to the data link layer software (which might be embedded in a chip on the network adaptor board). The data link layer software checks to see if this is the frame expected, and if so, gives the packet contained in the payload field to the routing software. The routing software chooses the appropriate outgoing line and passes the packet back down to the data link layer software, which then transmits it. The flow over two routers is shown in Fig. 3-2.

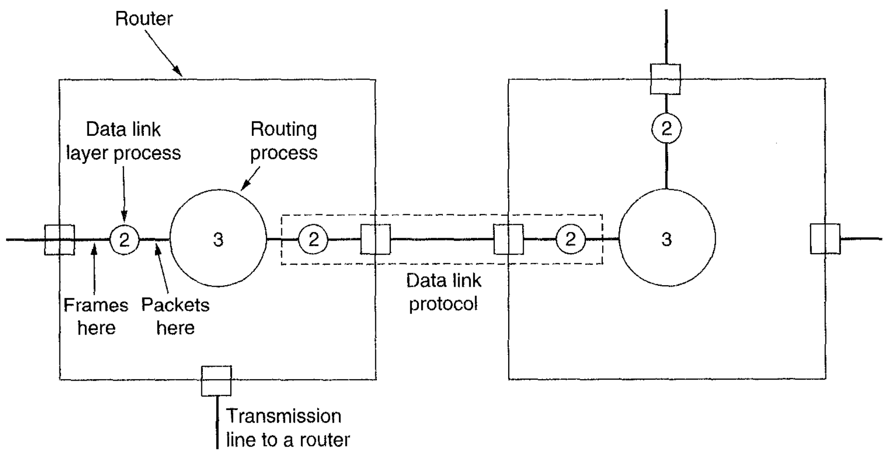

Fig. 3-2. Placement of the data link protocol.

The routing code frequently wants the job done right, that is, reliable, sequenced connections on each of the point-to-point lines. It does not want to be bothered too often with packets that got lost on the way. It is up to the data link protocol, shown in the dotted rectangle, to make unreliable communication lines look perfect, or at least, fairly good. This property is especially important for wireless links, which are inherently very unreliable. As an aside, although we have shown multiple copies of the data link layer software in each router, in fact, one copy handles all the lines, with different tables and data structures for each one.

Although this chapter is explicitly about the data link layer and the data link protocols, many of the principles we will study here, such as error control and flow control, are also found in transport and other protocols as well.

3.1.2. Framing

In order to provide service to the network layer, the data link layer must use the service provided to it by the physical layer. What the physical layer does is accept a raw bit stream and attempt to deliver it to the destination. This bit stream is not guaranteed to be error free. The number of bits received may be less than, equal to, or more than the number of bits transmitted, and they may have different values. It is up to the data link layer to detect, and if necessary, correct errors.

The usual approach is for the data link layer to break the bit stream up into discrete frames and compute the checksum for each frame. (Checksum algorithms will be discussed later in this chapter.) When a frame arrives at the destination, the checksum is recomputed. If the newly computed checksum is different from the one contained in the frame, the data link layer knows that an error has occurred and takes steps to deal with it (e.g., discarding the bad frame and sending back an error report).

Breaking the bit stream up into frames is more difficult than it at first appears. One way to achieve this framing is to insert time gaps between frames, much like the spaces between words in ordinary text. However, networks rarely make any guarantees about timing, so it is possible these gaps might be squeezed out, or other gaps might be inserted during transmission.

Since it is too risky to count on timing to mark the start and end of each frame, other methods have been devised. In this section we will look at four methods:

1. Character count.
2. Starting and ending characters, with character stuffing.
3. Starting and ending flags, with bit stuffing.
4. Physical layer coding violations.

The first framing method uses a field in the header to specify the number of characters in the frame. When the data link layer at the destination sees the character count, it knows how many characters follow, and hence where the end of the frame is. This technique is shown in Fig. 3-3(a) for four frames of sizes 5, 5, 8, and 9 characters respectively.

The trouble with this algorithm is that the count can be garbled by a transmission error. For example, if the character count of 5 in the second frame of Fig. 3-3(b) becomes a 7, the destination will get out of synchronization and will be unable to locate the start of the next frame. Even if the checksum is incorrect so the destination knows that the frame is bad, it still has no way of telling where the

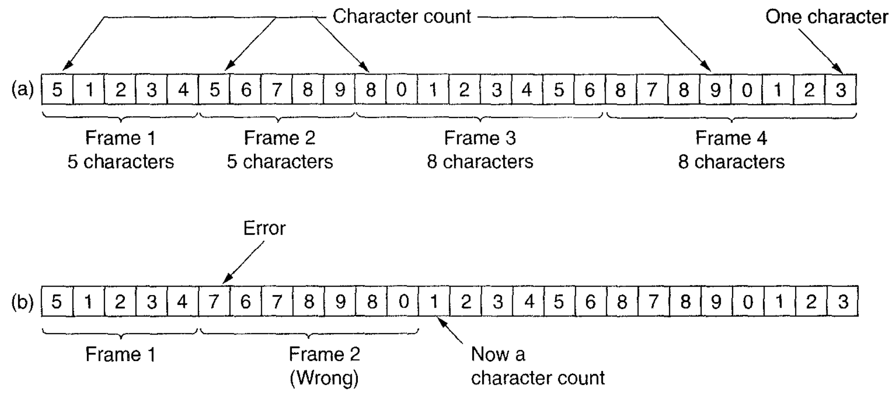

Fig. 3-3. A character stream. (a) Without errors. (b) With one error.

next frame starts. Sending a frame back to the source asking for a retransmission does not help either, since the destination does not know how many characters to skip over to get to the start of the retransmission. For this reason, the character count method is rarely used anymore.

The second framing method gets around the problem of resynchronization after an error by having each frame start with the ASCII character sequence DLE STX and end with the sequence DLE ETX. (DLE is Data Link Escape, STX is Start of TeXt, and ETX is End of TeXt.) In this way, if the destination ever loses track of the frame boundaries, all it has to do is look for DLE STX or DLE ETX characters to figure out where it is.

A serious problem occurs with this method when binary data, such as object programs or floating-point numbers, are being transmitted. It may easily happen that the characters for DLE STX or DLE ETX occur in the data, which will interfere with the framing. One way to solve this problem is to have the sender's data link layer insert an ASCII DLE character just before each "accidental" DLE character in the data. The data link layer on the receiving end removes the DLE before the data are given to the network layer. This technique is called **character stuffing**. Thus a framing DLE STX or DLE ETX can be distinguished from one in the data by the absence or presence of a single DLE. DLEs in the data are always doubled. Figure 3-4 gives an example data stream before stuffing, after stuffing, and after destuffing.

A major disadvantage of using this framing method is that it is closely tied to 8-bit characters in general and the ASCII character code in particular. As networks developed, the disadvantages of embedding the character code in the framing mechanism became more and more obvious so a new technique had to be developed to allow arbitrary sized characters.

The new technique allows data frames to contain an arbitrary number of bits

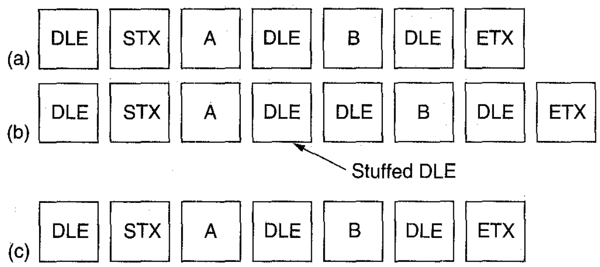

Fig. 3-4. (a) Data sent by the network layer. (b) Data after being character stuffed by the data link layer. (c) Data passed to the network layer on the receiving side.

and allows character codes with an arbitrary number of bits per character. It works like this. Each frame begins and ends with a special bit pattern, 01111110, called a **flag** byte. Whenever the sender's data link layer encounters five consecutive ones in the data, it automatically stuffs a 0 bit into the outgoing bit stream. This **bit stuffing** is analogous to character stuffing, in which a DLE is stuffed into the outgoing character stream before DLE in the data.

When the receiver sees five consecutive incoming 1 bits, followed by a 0 bit, it automatically destuffs (i.e., deletes) the 0 bit. Just as character stuffing is completely transparent to the network layer in both computers, so is bit stuffing. If the user data contain the flag pattern, 01111110, this flag is transmitted as 011111010 but stored in the receiver's memory as 01111110. Figure 3-5 gives an example of bit stuffing.

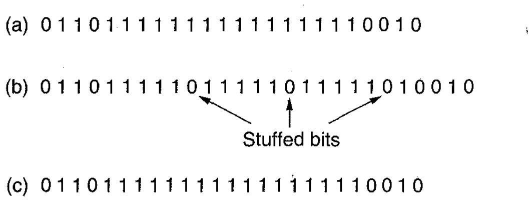

Fig. 3-5. Bit stuffing. (a) The original data. (b) The data as they appear on the line. (c) The data as they are stored in the receiver's memory after destuffing.

With bit stuffing, the boundary between two frames can be unambiguously recognized by the flag pattern. Thus if the receiver loses track of where it is, all it has to do is scan the input for flag sequences, since they can only occur at frame boundaries and never within the data.

The last method of framing is only applicable to networks in which the encoding on the physical medium contains some redundancy. For example, some LANs encode 1 bit of data by using 2 physical bits. Normally, a 1 bit is a high-low pair and a 0 bit is a low-high pair. The combinations high-high and low-low are not used for data. The scheme means that every data bit has a transition in the middle, making it easy for the receiver to locate the bit boundaries. This use of invalid physical codes is part of the 802 LAN standard, which we will study in Chap. 4.

As a final note on framing, many data link protocols use a combination of a character count with one of the other methods for extra safety. When a frame arrives, the count field is used to locate the end of the frame. Only if the appropriate delimiter is present at that position and the checksum is correct, is the frame accepted as valid. Otherwise, the input stream is scanned for the next delimiter.

3.1.3. Error Control

Having solved the problem of marking the start and end of each frame, we come to the next problem: how to make sure all frames are eventually delivered to the network layer at the destination, and in the proper order. Suppose that the sender just kept outputting frames without regard to whether they were arriving properly. This might be fine for unacknowledged connectionless service but would most certainly not be fine for reliable, connection-oriented service.

The usual way to ensure reliable delivery is to provide the sender with some feedback about what is happening at the other end of the line. Typically the protocol calls for the receiver to send back special control frames bearing positive or negative acknowledgements about the incoming frames. If the sender receives a positive acknowledgement about a frame, it knows the frame has arrived safely. On the other hand, a negative acknowledgement means that something has gone wrong, and the frame must be transmitted again.

An additional complication comes from the possibility that hardware troubles may cause a frame to vanish completely (e.g., in a noise burst). In this case, the receiver will not react at all, since it has no reason to react. It should be clear that a protocol in which the sender transmitted a frame and then waited for an acknowledgement, positive or negative, would hang forever if a frame were ever completely lost due to malfunctioning hardware.

This possibility is dealt with by introducing timers into the data link layer. When the sender transmits a frame, it generally also starts a timer. The timer is set to go off after an interval long enough for the frame to reach the destination, be processed there, and have the acknowledgement propagate back to the sender. Normally, the frame will be correctly received and the acknowledgement will get back before the timer runs out, in which case it will be canceled.

However, if either the frame or the acknowledgement is lost, the timer will go off, alerting the sender to a potential problem. The obvious solution is to just transmit the frame again. However, when frames may be transmitted multiple times there is a danger that the receiver will accept the same frame two or more times, and pass it to the network layer more than once. To prevent this from happening, it is generally necessary to assign sequence numbers to outgoing frames, so that the receiver can distinguish retransmissions from originals.

The whole issue of managing the timers and sequence numbers so as to ensure that each frame is ultimately passed to the network layer at the destination exactly once, no more and no less, is an important part of the data link layer's duties. Later in this chapter, we will study in detail how this management is done by looking at a series of increasingly sophisticated examples.

3.1.4. Flow Control

Another important design issue that occurs in the data link layer (and higher layers as well) is what to do with a sender that systematically wants to transmit frames faster than the receiver can accept them. This situation can easily occur when the sender is running on a fast (or lightly loaded) computer and the receiver is running on a slow (or heavily loaded) machine. The sender keeps pumping the frames out at a high rate until the receiver is completely swamped. Even if the transmission is error free, at a certain point the receiver will simply not be able to handle the frames as they arrive and will start to lose some. Clearly, something has to be done to prevent this situation.

The usual solution is to introduce **flow control** to throttle the sender into sending no faster than the receiver can handle the traffic. This throttling generally requires some kind of a feedback mechanism, so the sender can be made aware of whether or not the receiver is able to keep up.

Various flow control schemes are known, but most of them use the same basic principle. The protocol contains well-defined rules about when a sender may transmit the next frame. These rules often prohibit frames from being sent until the receiver has granted permission, either implicitly or explicitly. For example, when a connection is set up, the receiver might say: "You may send me *n* frames now, but after they have been sent, do not send any more until I have told you to continue." In this chapter, we will study various flow control mechanisms based on this principle. In subsequent chapters, we will study other mechanisms.

3.2. ERROR DETECTION AND CORRECTION

As we saw in Chap. 2, the telephone system has three parts: the switches, the interoffice trunks, and the local loops. The first two are now almost entirely digital in the United States and some other countries. The local loops are still analog twisted copper pairs everywhere and will continue to be so for decades due to the enormous expense of replacing them. While errors are rare on the digital part, they are still common on the local loops. Furthermore, wireless communication is becoming more common, and the error rates here are orders of magnitude worse than on the interoffice fiber trunks. The conclusion is: transmission errors are going to be a fact of life for many years to come.

As a result of the physical processes that generate them, errors on some media (e.g., radio) tend to come in bursts rather than singly. Having the errors come in bursts has both advantages and disadvantages over isolated single-bit errors. On the advantage side, computer data are always sent in blocks of bits. Suppose that the block size is 1000 bits, and the error rate is 0.001 per bit. If errors were independent, most blocks would contain an error. If the errors came in bursts of 100 however, only one or two blocks in 100 would be affected, on the average. The disadvantage of burst errors is that they are much harder to detect and correct than are isolated errors.

3.2.1. Error-Correcting Codes

Network designers have developed two basic strategies for dealing with errors. One way is to include enough redundant information along with each block of data sent to enable the receiver to deduce what the transmitted character must have been. The other way is to include only enough redundancy to allow the receiver to deduce that an error occurred, but not which error, and have it request a retransmission. The former strategy uses **error-correcting codes** and the latter uses **error-detecting codes**.

To understand how errors can be handled, it is necessary to look closely at what an error really is. Normally, a frame consists of m data (i.e., message) bits and r redundant, or check bits. Let the total length be n (i.e., $n = m + r$). An n-bit unit containing data and checkbits is often referred to as an n-bit **codeword**.

Given any two codewords, say, 10001001 and 10110001, it is possible to determine how many corresponding bits differ. In this case, 3 bits differ. To determine how many bits differ, just EXCLUSIVE OR the two codewords, and count the number of 1 bits in the result. The number of bit positions in which two codewords differ is called the **Hamming distance** (Hamming, 1950). Its significance is that if two codewords are a Hamming distance d apart, it will require d single-bit errors to convert one into the other.

In most data transmission applications, all 2^m possible data messages are legal, but due to the way the check bits are computed, not all of the 2^n possible codewords are used. Given the algorithm for computing the check bits, it is possible to construct a complete list of the legal codewords, and from this list find the two codewords whose Hamming distance is minimum. This distance is the Hamming distance of the complete code.

The error-detecting and error-correcting properties of a code depend on its Hamming distance. To detect d errors, you need a distance $d + 1$ code because with such a code there is no way that d single-bit errors can change a valid codeword into another valid codeword. When the receiver sees an invalid codeword, it can tell that a transmission error has occurred. Similarly, to correct d errors, you need a distance $2d + 1$ code because that way the legal codewords are so far apart that even with d changes, the original codeword is still closer than any other codeword, so it can be uniquely determined.

As a simple example of an error-detecting code, consider a code in which a single **parity bit** is appended to the data. The parity bit is chosen so that the number of 1 bits in the codeword is even (or odd). For example, when 10110101 is sent in even parity by adding a bit at the end, it becomes 101101011, whereas 10110001 becomes 101100010 with even parity. A code with a single parity bit has a distance 2, since any single-bit error produces a codeword with the wrong parity. It can be used to detect single errors.

As a simple example of an error-correcting code, consider a code with only four valid codewords:

0000000000, 0000011111, 1111100000, and 1111111111

This code has a distance 5, which means that it can correct double errors. If the codeword 0000000111 arrives, the receiver knows that the original must have been 0000011111. If, however, a triple error changes 0000000000 into 0000000111, the error will not be corrected properly.

Imagine that we want to design a code with m message bits and r check bits that will allow all single errors to be corrected. Each of the 2^m legal messages has n illegal codewords at a distance 1 from it. These are formed by systematically inverting each of the n bits in the n-bit codeword formed from it. Thus each of the 2^m legal messages requires $n + 1$ bit patterns dedicated to it. Since the total number of bit patterns is 2^n, we must have $(n + 1)2^m \leq 2^n$. Using $n = m + r$, this requirement becomes $(m + r + 1) \leq 2^r$. Given m, this puts a lower limit on the number of check bits needed to correct single errors.

This theoretical lower limit can, in fact, be achieved using a method due to Hamming (1950). The bits of the codeword are numbered consecutively, starting with bit 1 at the left end. The bits that are powers of 2 (1, 2, 4, 8, 16, etc.) are check bits. The rest (3, 5, 6, 7, 9, etc.) are filled up with the m data bits. Each check bit forces the parity of some collection of bits, including itself, to be even (or odd). A bit may be included in several parity computations. To see which check bits the data bit in position k contributes to, rewrite k as a sum of powers of 2. For example, $11 = 1 + 2 + 8$ and $29 = 1 + 4 + 8 + 16$. A bit is checked by just those check bits occurring in its expansion (e.g., bit 11 is checked by bits 1, 2, and 8).

When a codeword arrives, the receiver initializes a counter to zero. It then examines each check bit, k ($k = 1, 2, 4, 8, \ldots$) to see if it has the correct parity. If not, it adds k to the counter. If the counter is zero after all the check bits have been examined (i.e., if they were all correct), the codeword is accepted as valid. If the counter is nonzero, it contains the number of the incorrect bit. For example, if check bits 1, 2, and 8 are in error, the inverted bit is 11, because it is the only

Char.	ASCII	Check bits
H	1001000	00110010000
a	1100001	10111001001
m	1101101	11101010101
m	1101101	11101010101
i	1101001	01101011001
n	1101110	01101010110
g	1100111	11111001111
	0100000	10011000000
c	1100011	11111000011
o	1101111	00101011111
d	1100100	11111001100
e	1100101	00111000101

Order of bit transmission

Fig. 3-6. Use of a Hamming code to correct burst errors.

one checked by bits 1, 2, and 8. Figure 3-6 shows some 7-bit ASCII characters encoded as 11-bit codewords using a Hamming code. Remember that the data are found in bit positions 3, 5, 6, 7, 9, 10, and 11.

Hamming codes can only correct single errors. However, there is a trick that can be used to permit Hamming codes to correct burst errors. A sequence of k consecutive codewords are arranged as a matrix, one codeword per row. Normally, the data would be transmitted one codeword at a time, from left to right. To correct burst errors, the data should be transmitted one column at a time, starting with the leftmost column. When all k bits have been sent, the second column is sent, and so on. When the frame arrives at the receiver, the matrix is reconstructed, one column at a time. If a burst error of length k occurs, at most 1 bit in each of the k codewords will have been affected, but the Hamming code can correct one error per codeword, so the entire block can be restored. This method uses kr check bits to make blocks of km data bits immune to a single burst error of length k or less.

3.2.2. Error-Detecting Codes

Error-correcting codes are sometimes used for data transmission, for example, when the channel is simplex, so retransmissions cannot be requested, but most often error detection followed by retransmission is preferred because it is more efficient. As a simple example, consider a channel on which errors are isolated and the error rate is 10^{-6} per bit. Let the block size be 1000 bits. To provide error correction for 1000-bit blocks, 10 check bits are needed; a megabit of data would require 10,000 check bits. To merely detect a block with a single 1-bit error, one parity bit per block will suffice. Once every 1000 blocks an extra block (1001 bits) will have to be transmitted. The total overhead for the error detection +

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retransmission method is only 2001 bits per megabit of data, versus 10,000 bits for a Hamming code.

If a single parity bit is added to a block and the block is badly garbled by a long burst error, the probability that the error will be detected is only 0.5, which is hardly acceptable. The odds can be improved considerably by regarding each block to be sent as a rectangular matrix n bits wide and k bits high. A parity bit is computed separately for each column and affixed to the matrix as the last row. The matrix is then transmitted one row at a time. When the block arrives, the receiver checks all the parity bits. If any one of them is wrong, it requests a retransmission of the block.

This method can detect a single burst of length n, since only 1 bit per column will be changed. A burst of length $n + 1$ will pass undetected, however, if the first bit is inverted, the last bit is inverted, and all the other bits are correct. (A burst error does not imply that all the bits are wrong; it just implies that at least the first and last are wrong.) If the block is badly garbled by a long burst or by multiple shorter bursts, the probability that any of the n columns will have the correct parity, by accident, is 0.5, so the probability of a bad block being accepted when it should not be is 2^{-n}.

Although the above scheme may sometimes be adequate, in practice, another method is in widespread use: the **polynomial code** (also known as a **cyclic redundancy code** or CRC code). Polynomial codes are based upon treating bit strings as representations of polynomials with coefficients of 0 and 1 only. A k-bit frame is regarded as the coefficient list for a polynomial with k terms, ranging from x^{k-1} to x^0. Such a polynomial is said to be of degree $k - 1$. The high-order (leftmost) bit is the coefficient of x^{k-1}; the next bit is the coefficient of x^{k-2}, and so on. For example, 110001 has 6 bits and thus represents a six-term polynomial with coefficients 1, 1, 0, 0, 0, and 1: $x^5 + x^4 + x^0$.

Polynomial arithmetic is done modulo 2, according to the rules of algebraic field theory. There are no carries for addition or borrows for subtraction. Both addition and subtraction are identical to EXCLUSIVE OR. For example:

$$
\begin{array}{r} 10011011 \\ +\,11001010 \\ \hline 01010001 \end{array}
\qquad
\begin{array}{r} 00110011 \\ +\,11001101 \\ \hline 11111110 \end{array}
\qquad
\begin{array}{r} 11110000 \\ -\,10100110 \\ \hline 01010110 \end{array}
\qquad
\begin{array}{r} 01010101 \\ -\,10101111 \\ \hline 11111010 \end{array}
$$

Long division is carried out the same way as it is in binary except that the subtraction is done modulo 2, as above. A divisor is said "to go into" a dividend if the dividend has as many bits as the divisor.

When the polynomial code method is employed, the sender and receiver must agree upon a **generator polynomial**, $G(x)$, in advance. Both the high- and low-order bits of the generator must be 1. To compute the **checksum** for some frame with m bits, corresponding to the polynomial $M(x)$, the frame must be longer than the generator polynomial. The idea is to append a checksum to the end of the frame in such a way that the polynomial represented by the checksummed frame is divisible by $G(x)$. When the receiver gets the checksummed frame, it tries dividing it by $G(x)$. If there is a remainder, there has been a transmission error.

The algorithm for computing the checksum is as follows:

1. Let r be the degree of $G(x)$. Append r zero bits to the low-order end of the frame, so it now contains $m + r$ bits and corresponds to the polynomial $x^r M(x)$.

2. Divide the bit string corresponding to $G(x)$ into the bit string corresponding to $x^r M(x)$ using modulo 2 division.

3. Subtract the remainder (which is always r or fewer bits) from the bit string corresponding to $x^r M(x)$ using modulo 2 subtraction. The result is the checksummed frame to be transmitted. Call its polynomial $T(x)$.

Figure 3-7 illustrates the calculation for a frame 1101011011 and $G(x) = x^4 + x + 1$.

It should be clear that $T(x)$ is divisible (modulo 2) by $G(x)$. In any division problem, if you diminish the dividend by the remainder, what is left over is divisible by the divisor. For example, in base 10, if you divide 210,278 by 10,941, the remainder is 2399. By subtracting off 2399 from 210,278, what is left over (207,879) is divisible by 10,941.

Now let us analyze the power of this method. What kinds of errors will be detected? Imagine that a transmission error occurs, so that instead of the bit string for $T(x)$ arriving, $T(x) + E(x)$ arrives. Each 1 bit in $E(x)$ corresponds to a bit that has been inverted. If there are k 1 bits in $E(x)$, k single-bit errors have occurred. A single burst error is characterized by an initial 1, a mixture of 0s and 1s, and a final 1, with all other bits being 0.

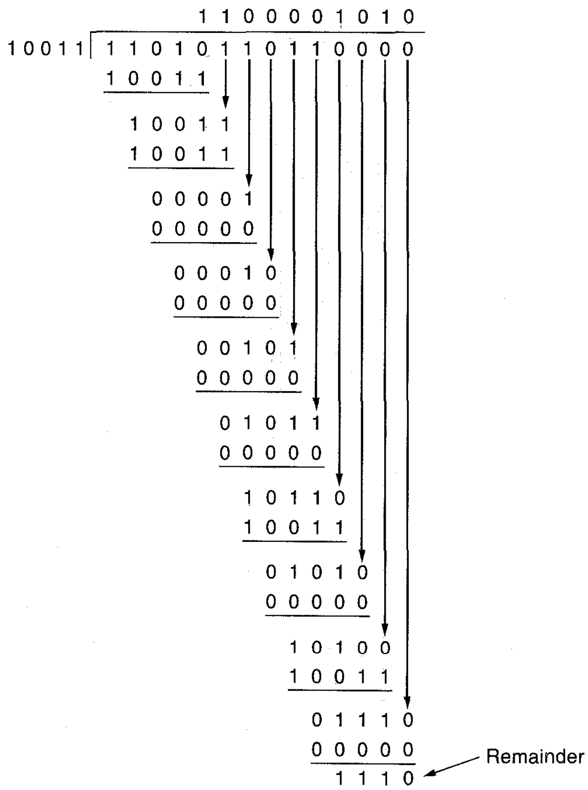
Upon receiving the checksummed frame, the receiver divides it by $G(x)$; that is, it computes $[T(x) + E(x)]/G(x)$. $T(x)/G(x)$ is 0, so the result of the computation is simply $E(x)/G(x)$. Those errors that happen to correspond to polynomials containing $G(x)$ as a factor will slip by; all other errors will be caught.

If there has been a single-bit error, $E(x) = x^i$, where i determines which bit is in error. If $G(x)$ contains two or more terms, it will never divide $E(x)$, so all single-bit errors will be detected.

If there have been two isolated single-bit errors, $E(x) = x^i + x^j$, where $i > j$. Alternatively, this can be written as $E(x) = x^j(x^{i-j} + 1)$. If we assume that $G(x)$ is not divisible by x, a sufficient condition for all double errors to be detected is that $G(x)$ does not divide $x^k + 1$ for any k up to the maximum value of $i - j$ (i.e., up to the maximum frame length). Simple, low-degree polynomials that give protection to long frames are known. For example, $x^{15} + x^{14} + 1$ will not divide $x^k + 1$ for any value of k below 32,768.

If there are an odd number of bits in error, $E(X)$ contains an odd number of terms (e.g., $x^5 + x^2 + 1$, but not $x^2 + 1$). Interestingly enough, there is no

Frame : 1 1 0 1 0 1 1 0 1 1
Generator: 1 0 0 1 1
Message after appending 4 zero bits: 1 1 0 1 0 1 1 0 1 1 0 0 0 0



Transmitted frame: 1 1 0 1 0 1 1 0 1 1 1 1 1 0

Fig. 3-7. Calculation of the polynomial code checksum.

polynomial with an odd number of terms that has $x + 1$ as a factor in the modulo 2 system. By making $x + 1$ a factor of $G(x)$, we can catch all errors consisting of an odd number of inverted bits.

To see that no polynomial with an odd number of terms is divisible by $x + 1$, assume that $E(x)$ has an odd number of terms and is divisible by $x + 1$. Factor $E(x)$ into $(x + 1)\ Q(x)$. Now evaluate $E(1) = (1 + 1)Q(1)$. Since $1 + 1 = 0$ (modulo 2), $E(1)$ must be zero. If $E(x)$ has an odd number of terms, substituting 1 for x everywhere will always yield 1 as result. Thus no polynomial with an odd number of terms is divisible by $x + 1$.

Finally, and most important, a polynomial code with r check bits will detect all burst errors of length $\leq r$. A burst error of length k can be represented by $x^i(x^{k-1} + \ldots + 1)$, where i determines how far from the right-hand end of the received frame the burst is located. If $G(x)$ contains an x^0 term, it will not have x^i as a factor, so if the degree of the parenthesized expression is less than the degree of $G(x)$, the remainder can never be zero.

If the burst length is $r + 1$, the remainder of the division by $G(x)$ will be zero if and only if the burst is identical to $G(x)$. By definition of a burst, the first and last bits must be 1, so whether it matches depends on the $r - 1$ intermediate bits. If all combinations are regarded as equally likely, the probability of such an incorrect frame being accepted as valid is $\frac{1}{2}^{r-1}$.

It can also be shown that when an error burst longer than $r + 1$ bits occurs, or several shorter bursts occur, the probability of a bad frame getting through unnoticed is $\frac{1}{2}^r$ assuming that all bit patterns are equally likely.

Three polynomials have become international standards:

CRC–12 $= x^{12} + x^{11} + x^3 + x^2 + x^1 + 1$
CRC–16 $= x^{16} + x^{15} + x^2 + 1$ 0x3EEF
CRC–CCITT $= x^{16} + x^{12} + x^5 + 1$

All three contain $x + 1$ as a prime factor. CRC-12 is used when the character length is 6 bits. The other two are used for 8-bit characters. A 16-bit checksum, such as CRC-16 or CRC-CCITT, catches all single and double errors, all errors with an odd number of bits, all burst errors of length 16 or less, 99.997 percent of 17-bit error bursts, and 99.998 percent of 18-bit and longer bursts.

Although the calculation required to compute the checksum may seem complicated, Peterson and Brown (1961) have shown that a simple shift register circuit can be constructed to compute and verify the checksums in hardware. In practice, this hardware is nearly always used.

For decades, it has been assumed that frames to be checksummed contain random bits. All analyses of checksum algorithms have been made under this assumption. More recently inspection of real data has shown this assumption to be quite wrong. As a consequence, under some circumstances, undetected errors are much more common than had been previously thought (Partridge et al., 1995).

3.3. ELEMENTARY DATA LINK PROTOCOLS

To introduce the subject of protocols, we will begin by looking at three protocols of increasing complexity. For interested readers, a simulator for these and subsequent protocols is available via the WWW (see the preface). Before we look at the protocols, it is useful to make explicit some of the assumptions underlying the model of communication. To start with, we are assuming that in the physical layer, data link layer, and network layer are independent processes that communicate by passing messages back and forth. In some cases, the physical and data link layer processes will be running on a processor inside a special network I/O chip and the network layer on the main CPU, but other implementations are also possible (e.g., three processes inside a single I/O chip; the physical and data link layers as procedures called by the network layer process, and so on). In any event, treating the three layers as separate processes makes the discussion conceptually cleaner and also serves to emphasize the independence of the layers.

Another key assumption is that machine *A* wants to send a long stream of data to machine *B* using a reliable, connection-oriented service. Later, we will consider the case where *B* also wants to send data to *A* simultaneously. *A* is assumed to have an infinite supply of data ready to send and never has to wait for data to be produced. When *A*'s data link layer asks for data, the network layer is always able to comply immediately. (This restriction, too, will be dropped later.)

As far as the data link layer is concerned, the packet passed across the interface to it from the network layer is pure data, every bit of which is to be delivered to the destination's network layer. The fact that the destination's network layer may interpret part of the packet as a header is of no concern to the data link layer.

When the data link layer accepts a packet, it encapsulates the packet in a frame by adding a data link header and trailer to it (see Fig. 1-11). Thus a frame consists of an embedded packet and some control (header) information. The frame is then transmitted to the other data link layer. We will assume that there exist suitable library procedures *to_physical_layer* to send a frame and *from_physical_layer* to receive a frame. The transmitting hardware computes and appends the checksum, so that the data link layer software need not worry about it. The polynomial algorithm discussed earlier in this chapter might be used, for example.

Initially, the receiver has nothing to do. It just sits around waiting for something to happen. In the example protocols of this chapter we indicate that the data link layer is waiting for something to happen by the procedure call *wait_for_event*(&*event*). This procedure only returns when something has happened (e.g., a frame has arrived). Upon return, the variable *event* tells what happened. The set of possible events differs for the various protocols to be described and will be defined separately for each protocol. Note that in a more realistic situation, the data link layer will not sit in a tight loop waiting for an event, as we have suggested, but will receive an interrupt, which will cause it to stop whatever it was doing and go handle the incoming frame. Nevertheless, for simplicity we will ignore all the details of parallel activity within the data link layer and assume that it is dedicated full time to handling just our one channel.

When a frame arrives at the receiver, the hardware computes the checksum. If the checksum is incorrect (i.e., there was a transmission error), the data link layer is so informed (*event* = *cksum_err*). If the inbound frame arrived undamaged, the data link layer is also informed (*event* = *frame_arrival*), so it can acquire the frame for inspection using *from_physical_layer*. As soon as the receiving data link layer has acquired an undamaged frame, it checks the control information in the header, and if everything is all right, the packet portion is passed to the network layer. Under no circumstances is a frame header ever given to a network layer.

There is a good reason why the network layer must never be given any part of the frame header: to keep the network and data link protocols completely separate. As long as the network layer knows nothing at all about the data link protocol or the frame format, these things can be changed without requiring changes to the network layer's software. Providing a rigid interface between network layer and data link layer greatly simplifies the software design, because communication protocols in different layers can evolve independently.

Figure 3-8 shows some declarations (in C) common to many of the protocols to be discussed later. Five data structures are defined there: *boolean*, *seq_nr*, *packet*, *frame_kind*, and *frame*. A *boolean* is an enumerated type and can take on the values *true* and *false*. A *seq_nr* is a small integer used to number the frames, so we can tell them apart. These sequence numbers run from 0 up to and including *MAX_SEQ*, which is defined in each protocol needing it. A *packet* is the unit of information exchanged between the network layer and the data link layer on the same machine, or between network layer peers. In our model it always contains *MAX_PKT* bytes, but more realistically it would be of variable length.

A *frame* is composed of four fields: *kind*, *seq*, *ack*, and *info*, the first three of which contain control information, and the last of which may contain actual data to be transferred. These control fields are collectively called the **frame header**. The *kind* field tells whether or not there are any data in the frame, because some of the protocols distinguish frames containing exclusively control information from those containing data as well. The *seq* and *ack* fields are used for sequence numbers and acknowledgements, respectively; their use will be described in more detail later. The *info* field of a data frame contains a single packet; the *info* field of a control frame is not used. A more realistic implementation would use a variable-length *info* field, omitting it altogether for control frames.

It is important to realize the relationship between a packet and a frame. The network layer builds a packet by taking a message from the transport layer and adding the network layer header to it. This packet is passed to the data link layer for inclusion in the *info* field of an outgoing frame. When the frame arrives at the destination, the data link layer extracts the packet from the frame and passes the packet to the network layer. In this manner, the network layer can act as though machines can exchange packets directly.

A number of procedures are also listed in Fig. 3-8. These are library routines whose details are implementation-dependent and whose inner workings will not concern us further here. The procedure *wait_for_event* sits in a tight loop waiting for something to happen, as mentioned earlier. The procedures *to_network_layer* and *from_network_layer* are used by the data link layer to pass packets to the network layer and accept packets from the network layer, respectively. Note that *from_physical_layer* and *to_physical_layer* are used for passing frames between the data link and physical layers, whereas the procedures *to_network_layer* and *from_network_layer* are used for passing packets between the data link layer and network layer. In other words, *to_network_layer* and *from_network_layer* deal with the interface between layers 2 and 3, whereas *from_physical_layer* and *to_physical_layer* deal with the interface between layers 1 and 2.

In most of the protocols we assume an unreliable channel that loses entire frames upon occasion. To be able to recover from such calamities, the sending data link layer must start an internal timer or clock whenever it sends a frame. If no reply has been received within a certain predetermined time interval, the clock times out and the data link layer receives an interrupt signal.

In our protocols this is handled by allowing the procedure *wait_for_event* to return *event* = *timeout*. The procedures *start_timer* and *stop_timer* are used to turn the timer on and off, respectively. Timeouts are possible only when the timer is running. It is explicitly permitted to call *start_timer* while the timer is running; such a call simply resets the clock to cause the next timeout after a full timer interval has elapsed (unless it is reset or turned off in the meanwhile).

The procedures *start_ack_timer* and *stop_ack_timer* are used to control an auxiliary timer used to generate acknowledgements under certain conditions.

The procedures *enable_network_layer* and *disable_network_layer* are used in the more sophisticated protocols, where we no longer assume that the network layer always has packets to send. When the data link layer enables the network layer, the network layer is then permitted to interrupt when it has a packet to be sent. We indicate this with *event* = *network_layer_ready*. When a network layer is disabled, it may not cause such events. By being careful about when it enables and disables its network layer, the data link layer can prevent the network layer from swamping it with packets for which it has no buffer space.

Frame sequence numbers are always in the range 0 to *MAX_SEQ* (inclusive), where *MAX_SEQ* is different for the different protocols. It is frequently necessary to advance a sequence number by 1 circularly (i.e., *MAX_SEQ* is followed by 0). The macro *inc* performs this incrementing. It has been defined as a macro because it is used in-line within the critical path. As we will see later in this book, the factor limiting network performance is often protocol processing, so defining simple operations like this as macros does not affect the readability of the code, but does improve performance. Also, since *MAX_SEQ* will have different values in different protocols, by making it a macro, it becomes possible to include all the protocols in the same binary without conflict. This ability is useful for the simulator.

The declarations of Fig. 3-8 are part of each of the protocols to follow. To save space and to provide a convenient reference, they have been extracted and

```
#define MAX_PKT 1024                    /* determines packet size in bytes */

typedef enum {false, true} boolean;     /* boolean type */
typedef unsigned int seq_nr;            /* sequence or ack numbers */
typedef struct {unsigned char data[MAX_PKT];} packet;/* packet definition */
typedef enum {data, ack, nak} frame_kind;   /* frame_kind definition */

typedef struct {                        /* frames are transported in this layer */
  frame_kind kind;                      /* what kind of a frame is it? */
  seq_nr seq;                           /* sequence number */
  seq_nr ack;                           /* acknowledgement number */
  packet info;                          /* the network layer packet */
} frame;

/* Wait for an event to happen; return its type in event. */
void wait_for_event(event_type *event);

/* Fetch a packet from the network layer for transmission on the channel. */
void from_network_layer(packet *p);

/* Deliver information from an inbound frame to the network layer. */
void to_network_layer(packet *p);

/* Go get an inbound frame from the physical layer and copy it to r. */
void from_physical_layer(frame *r);

/* Pass the frame to the physical layer for transmission. */
void to_physical_layer(frame *s);

/* Start the clock running and enable the timeout event. */
void start_timer(seq_nr k);

/* Stop the clock and disable the timeout event. */
void stop_timer(seq_nr k);

/* Start an auxiliary timer and enable the ack_timeout event. */
void start_ack_timer(void);

/* Stop the auxiliary timer and disable the ack_timeout event. */
void stop_ack_timer(void);

/* Allow the network layer to cause a network_layer_ready event. */
void enable_network_layer(void);

/* Forbid the network layer from causing a network_layer_ready event. */
void disable_network_layer(void);

/* Macro inc is expanded in-line: Increment k circularly. */
#define inc(k) if (k < MAX_SEQ) k = k + 1; else k = 0
```

Fig. 3-8. Some definitions needed in the protocols to follow. These definitions are located in the file *protocol.h*.

listed together, but conceptually they should be merged with the protocols themselves. In C, this merging is done by putting the definitions in a special header file, in this case *protocol.h*, and using the #include facility of the C preprocessor to include them in the protocol files.

3.3.1. An Unrestricted Simplex Protocol

As an initial example we will consider a protocol that is as simple as can be. Data are transmitted in one direction only. Both the transmitting and receiving network layers are always ready. Processing time can be ignored. Infinite buffer space is available. And best of all, the communication channel between the data link layers never damages or loses frames. This thoroughly unrealistic protocol, which we will nickname "utopia," is shown in Fig. 3-9.

The protocol consists of two distinct procedures, a sender and a receiver. The sender runs in the data link layer of the source machine, and the receiver runs in the data link layer of the destination machine. No sequence numbers or acknowledgements are used here, so *MAX_SEQ* is not needed. The only event type possible is *frame_arrival* (i.e., the arrival of an undamaged frame).

The sender is in an infinite while loop just pumping data out onto the line as fast as it can. The body of the loop consists of three actions: go fetch a packet from the (always obliging) network layer, construct an outbound frame using the variable *s*, and send the frame on its way. Only the *info* field of the frame is used by this protocol, because the other fields have to do with error and flow control, and there are no errors or flow control restrictions here.

The receiver is equally simple. Initially, it waits for something to happen, the only possibility being the arrival of an undamaged frame. Eventually, the frame arrives and the procedure *wait_for_event* returns, with *event* set to *frame_arrival* (which is ignored anyway). The call to *from_physical_layer* removes the newly arrived frame from the hardware buffer and puts it in the variable *r*. Finally, the data portion is passed on to the network layer and the data link layer settles back to wait for the next frame, effectively suspending itself until the frame arrives.

3.3.2. A Simplex Stop-and-Wait Protocol

Now we will drop the most unrealistic restriction used in protocol 1: the ability of the receiving network layer to process incoming data infinitely fast (or equivalently, the presence in the receiving data link layer of an infinite amount of buffer space in which to store all incoming frames while they are waiting their respective turns). The communication channel is still assumed to be error free however, and the data traffic is still simplex.

The main problem we have to deal with here is how to prevent the sender from flooding the receiver with data faster than the latter is able to process it. In essence, if the receiver requires a time Δt to execute *from_physical_layer* plus

```
/* Protocol 1 (utopia) provides for data transmission in one direction only, from
   sender to receiver. The communication channel is assumed to be error free,
   and the receiver is assumed to be able to process all the input infinitely fast.
   Consequently, the sender just sits in a loop pumping data out onto the line as
   fast as it can. */

typedef enum {frame_arrival} event_type;
#include "protocol.h"

void sender1(void)
{
 frame s;                           /* buffer for an outbound frame */
 packet buffer;                     /* buffer for an outbound packet */

 while (true) {
     from_network_layer(&buffer);   /* go get something to send */
     s.info = buffer;               /* copy it into s for transmission */
     to_physical_layer(&s);         /* send it on its way */
 }                                  /* Tomorrow, and tomorrow, and tomorrow,
                                       Creeps in this petty pace from day to day
                                       To the last syllable of recorded time
                                          - Macbeth, V, v */
}

void receiver1(void)
{
 frame r;
 event_type event;                  /* filled in by wait, but not used here */

 while (true) {
     wait_for_event(&event);        /* only possibility is frame_arrival */
     from_physical_layer(&r);       /* go get the inbound frame */
     to_network_layer(&r.info);     /* pass the data to the network layer */
 }
}
```

Fig. 3-9. An unrestricted simplex protocol.

to_network_layer, the sender must transmit at an average rate less than one frame per time Δt. Moreover, if we assume that there is no automatic buffering and queueing done within the receiver's hardware, the sender must never transmit a new frame until the old one has been fetched by *from_physical_layer*, lest the new one overwrite the old one.

In certain restricted circumstances (e.g., synchronous transmission and a receiving data link layer fully dedicated to processing the one input line), it might be possible for the sender to simply insert a delay into protocol 1 to slow it down sufficiently to keep from swamping the receiver. However, more usually, each data link layer will have several lines to attend to, and the time interval between a frame arriving and its being processed may vary considerably. If the network designers can calculate the worst-case behavior of the receiver, they can program the sender to transmit so slowly that even if every frame suffers the maximum delay, there will be no overruns. The trouble with this approach is that it is too conservative. It leads to a bandwidth utilization that is far below the optimum, unless the best and worst cases are almost the same (i.e., the variation in the data link layer's reaction time is small).

A more general solution to this dilemma is to have the receiver provide feedback to the sender. After having passed a packet to its network layer, the receiver sends a little dummy frame back to the sender which, in effect, gives the sender permission to transmit the next frame. After having sent a frame, the sender is required by the protocol to bide its time until the little dummy (i.e., acknowledgement) frame arrives.

Protocols in which the sender sends one frame and then waits for an acknowledgement before proceeding are called **stop-and-wait**. Figure 3-10 gives an example of a simplex stop-and-wait protocol.

As in protocol 1, the sender starts out by fetching a packet from the network layer, using it to construct a frame and sending it on its way. Only now, unlike in protocol 1, the sender must wait until an acknowledgement frame arrives before looping back and fetching the next packet from the network layer. The sending data link layer need not even inspect the incoming frame: there is only one possibility.

The only difference between *receiver1* and *receiver2* is that after delivering a packet to the network layer, *receiver2* sends an acknowledgement frame back to the sender before entering the wait loop again. Because only the arrival of the frame back at the sender is important, not its contents, the receiver need not put any particular information in it.

Although data traffic in this example is simplex, going only from the sender to the receiver, frames do travel in both directions. Consequently, the communication channel between the two data link layers needs to be capable of bidirectional information transfer. However, this protocol entails a strict alternation of flow: first the sender sends a frame, then the receiver sends a frame, then the sender sends another frame, then the receiver sends another one, and so on. A half-duplex physical channel would suffice here.

3.3.3. A Simplex Protocol for a Noisy Channel

Now let us consider the normal situation of a communication channel that makes errors. Frames may be either damaged or lost completely. However, we assume that if a frame is damaged in transit, the receiver hardware will detect this

```
/* Protocol 2 (stop-and-wait) also provides for a one-directional flow of data from
   sender to receiver. The communication channel is once again assumed to be error
   free, as in protocol 1. However, this time, the receiver has only a finite buffer
   capacity and a finite processing speed, so the protocol must explicitly prevent
   the sender from flooding the receiver with data faster than it can be handled. */

typedef enum {frame_arrival} event_type;
#include "protocol.h"

void sender2(void)
{
 frame s;                          /* buffer for an outbound frame */
 packet buffer;                    /* buffer for an outbound packet */
 event_type event;                 /* frame_arrival is the only possibility */

 while (true) {
     from_network_layer(&buffer);  /* go get something to send */
     s.info = buffer;              /* copy it into s for transmission */
     to_physical_layer(&s);        /* bye bye little frame */
     wait_for_event(&event);       /* do not proceed until given the go ahead */
 }
}

void receiver2(void)
{
 frame r, s;                       /* buffers for frames */
 event_type event;                 /* frame_arrival is the only possibility */
 while (true) {
     wait_for_event(&event);       /* only possibility is frame_arrival */
     from_physical_layer(&r);      /* go get the inbound frame */
     to_network_layer(&r.info);    /* pass the data to the network layer */
     to_physical_layer(&s);        /* send a dummy frame to awaken sender */
 }
}
```

Fig. 3-10. A simplex stop-and-wait protocol.

when it computes the checksum. If the frame is damaged in such a way that the checksum is nevertheless correct, an exceedingly unlikely occurrence, this protocol (and all other protocols) can fail (i.e., deliver an incorrect packet to the network layer).

At first glance it might seem that a variation of protocol 2 would work: adding a timer. The sender could send a frame, but the receiver would only send an acknowledgement frame if the data were correctly received. If a damaged frame arrived at the receiver, it would be discarded. After a while the sender would time out and send the frame again. This process would be repeated until the frame finally arrived intact.

The above scheme has a fatal flaw in it. Think about the problem and try to discover what might go wrong before reading further.

To see what might go wrong, remember that it is the task of the data link layer processes to provide error free, transparent communication between network layers processes. The network layer on machine *A* gives a series of packets to its data link layer, which must ensure that an identical series of packets are delivered to the network layer on machine *B* by its data link layer. In particular, the network layer on *B* has no way of knowing that a packet has been lost or duplicated, so the data link layer must guarantee that no combination of transmission errors, no matter how unlikely, can cause a duplicate packet to be delivered to a network layer.

Consider the following scenario:

1. The network layer on *A* gives packet 1 to its data link layer. The packet is correctly received at *B* and passed to the network layer on *B*. *B* sends an acknowledgement frame back to *A*.

2. The acknowledgement frame gets lost completely. It just never arrives at all. Life would be a great deal simpler if the channel only mangled and lost data frames and not control frames, but sad to say, the channel is not very discriminating.

3. The data link layer on *A* eventually times out. Not having received an acknowledgement, it (incorrectly) assumes that its data frame was lost or damaged and sends the frame containing packet 1 again.

4. The duplicate frame also arrives at data link layer on *B* perfectly and is unwittingly passed to the network layer there. If *A* is sending a file to *B*, part of the file will be duplicated (i.e., the copy of the file made by *B* will be incorrect and the error will not have been detected). In other words, the protocol will fail.

Clearly, what is needed is some way for the receiver to be able to distinguish a frame that it is seeing for the first time from a retransmission. The obvious way to achieve this is to have the sender put a sequence number in the header of each frame it sends. Then the receiver can check the sequence number of each arriving frame to see if it is a new frame or a duplicate to be discarded.

Since a small frame header is desirable, the question arises: What is the minimum number of bits needed for the sequence number? The only ambiguity in this protocol is between a frame, m, and its direct successor, $m + 1$. If frame m is lost or damaged, the receiver will not acknowledge it, so the sender will keep trying to send it. Once it has been correctly received, the receiver will send an acknowledgement back to the sender. It is here that the potential trouble crops up. Depending upon whether the acknowledgement frame gets back to the sender correctly or not, the sender may try to send m or $m + 1$.

The event that triggers the sender to start sending $m + 2$ is the arrival of an acknowledgement for $m + 1$. But this implies that m has been correctly received, and furthermore that its acknowledgement has also been correctly received by the sender (otherwise, the sender would not have begun with $m + 1$, let alone $m + 2$). As a consequence, the only ambiguity is between a frame and its immediate predecessor or successor, not between the predecessor and successor themselves.

A 1-bit sequence number (0 or 1) is therefore sufficient. At each instant of time, the receiver expects a particular sequence number next. Any arriving frame containing the wrong sequence number is rejected as a duplicate. When a frame containing the correct sequence number arrives, it is accepted, passed to the network layer, and the expected sequence number is incremented modulo 2 (i.e., 0 becomes 1 and 1 becomes 0).

An example of this kind of protocol is shown in Fig. 3-11. Protocols in which the sender waits for a positive acknowledgement before advancing to the next data item are often called **PAR** (**Positive Acknowledgement with Retransmission**) or **ARQ** (**Automatic Repeat reQuest**). Like protocol 2, this one also transmits data only in one direction. Although it can handle lost frames (by timing out), it requires the timeout interval to be long enough to prevent premature timeouts. If the sender times out too early, while the acknowledgement is still on the way, it will send a duplicate.

When the previous acknowledgement finally does arrive, the sender will mistakenly think that the just-sent frame is the one being acknowledged and will not realize that there is potentially another acknowledgement frame somewhere "in the pipe." If the next frame sent is lost completely but the extra acknowledgement arrives correctly, the sender will not attempt to retransmit the lost frame, and the protocol will fail. In later protocols the acknowledgement frames will contain information to prevent just this sort of trouble. For the time being, the acknowledgement frames will just be dummies, and we will assume a strict alternation of sender and receiver.

Protocol 3 differs from its predecessors in that both sender and receiver have a variable whose value is remembered while the data link layer is in wait state. The sender remembers the sequence number of the next frame to send in *next_frame_to_send*; the receiver remembers the sequence number of the next frame expected in *frame_expected*. Each protocol has a short initialization phase before entering the infinite loop.

After transmitting a frame, the sender starts the timer running. If it was already running, it will be reset to allow another full timer interval. The time interval must be chosen to allow enough time for the frame to get to the receiver, for the receiver to process it in the worst case, and for the acknowledgement frame to propagate back to the sender. Only when that time interval has elapsed

```
/* Protocol 3 (par) allows unidirectional data flow over an unreliable channel. */

#define MAX_SEQ 1                        /* must be 1 for protocol 3 */
typedef enum {frame_arrival, cksum_err, timeout} event_type;
#include "protocol.h"

void sender3(void)
{
 seq_nr next_frame_to_send;              /* seq number of next outgoing frame */
 frame s;                                /* scratch variable */
 packet buffer;                          /* buffer for an outbound packet */
 event_type event;

 next_frame_to_send = 0;                 /* initialize outbound sequence numbers */
 from_network_layer(&buffer);            /* fetch first packet */
 while (true) {
     s.info = buffer;                    /* construct a frame for transmission */
     s.seq = next_frame_to_send;         /* insert sequence number in frame */
     to_physical_layer(&s);              /* send it on its way */
     start_timer(s.seq);                 /* if answer takes too long, time out */
     wait_for_event(&event);             /* frame_arrival, cksum_err, timeout */
     if (event == frame_arrival) {
          from_physical_layer(&s);       /* get the acknowledgement */
          if (s.ack == next_frame_to_send) {
               from_network_layer(&buffer); /* get the next one to send */
               inc(next_frame_to_send);  /* invert next_frame_to_send */
          }
     }
 }
}

void receiver3(void)
{
 seq_nr frame_expected;
 frame r, s;
 event_type event;

 frame_expected = 0;
 while (true) {
     wait_for_event(&event);             /* possibilities: frame_arrival, cksum_err */
     if (event == frame_arrival) {       /* a valid frame has arrived. */
          from_physical_layer(&r);       /* go get the newly arrived frame */
          if (r.seq == frame_expected) { /* this is what we have been waiting for. */
               to_network_layer(&r.info); /* pass the data to the network layer */
               inc(frame_expected);      /* next time expect the other sequence nr */
          }
          s.ack = 1 – frame_expected;    /* tell which frame is being acked */
          to_physical_layer(&s);         /* none of the fields are used */
     }
 }
}
```

Fig. 3-11. A positive acknowledgement with retransmission protocol.

is it safe to assume that either the transmitted frame or its acknowledgement has been lost, and to send a duplicate.

After transmitting a frame and starting the timer, the sender waits for something exciting to happen. There are three possibilities: an acknowledgement frame arrives undamaged, a damaged acknowledgement frame staggers in, or the timer goes off. If a valid acknowledgement comes in, the sender fetches the next packet from its network layer and puts it in the buffer, overwriting the previous packet. It also advances the sequence number. If a damaged frame arrives or no frame at all arrives, neither the buffer nor the sequence number are changed, so that a duplicate can be sent.

When a valid frame arrives at the receiver, its sequence number is checked to see if it is a duplicate. If not, it is accepted, passed to the network layer, and an acknowledgement generated. Duplicates and damaged frames are not passed to the network layer.

3.4. SLIDING WINDOW PROTOCOLS

In the previous protocols, data frames were transmitted in one direction only. In most practical situations, there is a need for transmitting data in both directions. One way of achieving full-duplex data transmission is to have two separate communication channels and use each one for simplex data traffic (in different directions). If this is done, we have two separate physical circuits, each with a "forward" channel (for data) and a "reverse" channel (for acknowledgements). In both cases the bandwidth of the reverse channel is almost entirely wasted. In effect, the user is paying for two circuits but using only the capacity of one.

A better idea is to use the same circuit for data in both directions. After all, in protocols 2 and 3 it was already being used to transmit frames both ways, and the reverse channel has the same capacity as the forward channel. In this model the data frames from A to B are intermixed with the acknowledgement frames from A to B. By looking at the *kind* field in the header of an incoming frame, the receiver can tell whether the frame is data or acknowledgement.

Although interleaving data and control frames on the same circuit is an improvement over having two separate physical circuits, yet another improvement is possible. When a data frame arrives, instead of immediately sending a separate control frame, the receiver restrains itself and waits until the network layer passes it the next packet. The acknowledgement is attached to the outgoing data frame (using the *ack* field in the frame header). In effect, the acknowledgement gets a free ride on the next outgoing data frame. The technique of temporarily delaying outgoing acknowledgements so that they can be hooked onto the next outgoing data frame is known as **piggybacking**.

The principal advantage of using piggybacking over having distinct acknowledgement frames is a better use of the available channel bandwidth. The *ack* field in the frame header costs only a few bits, whereas a separate frame would need a header, the acknowledgement, and a checksum. In addition, fewer frames sent means fewer "frame arrived" interrupts, and perhaps fewer buffers in the receiver, depending on how the receiver's software is organized. In the next protocol to be examined, the piggyback field costs only 1 bit in the frame header. It rarely costs more than a few bits.

However, piggybacking introduces a complication not present with separate acknowledgements. How long should the data link layer wait for a packet onto which to piggyback the acknowledgement? If the data link layer waits longer than the sender's timeout period, the frame will be retransmitted, defeating the whole purpose of having acknowledgements. If the data link layer were an oracle and could foretell the future, it would know when the next network layer packet was going to come in, and could decide either to wait for it or send a separate acknowledgement immediately, depending on how long the projected wait was going to be. Of course, the data link layer cannot foretell the future, so it must resort to some ad hoc scheme, such as waiting a fixed number of milliseconds. If a new packet arrives quickly, the acknowledgement is piggybacked onto it; otherwise, if no new packet has arrived by the end of this time period, the data link layer just sends a separate acknowledgement frame.

In addition to its being only simplex, protocol 3 can fail under some peculiar conditions involving early timeout. It would be nicer to have a protocol that remained synchronized in the face of any combination of garbled frames, lost frames, and premature timeouts. The next three protocols are more robust and continue to function even under pathological conditions. All three belong to a class of protocols called **sliding window** protocols. The three differ among themselves in terms of efficiency, complexity, and buffer requirements, as discussed later.

In all sliding window protocols, each outbound frame contains a sequence number, ranging from 0 up to some maximum. The maximum is usually $2^n - 1$ so the sequence number fits nicely in an n-bit field. The stop-and-wait sliding window protocol uses $n = 1$, restricting the sequence numbers to 0 and 1, but more sophisticated versions can use arbitrary n.

The essence of all sliding window protocols is that at any instant of time, the sender maintains a set of sequence numbers corresponding to frames it is permitted to send. These frames are said to fall within the **sending window**. Similarly, the receiver also maintains a **receiving window** corresponding to the set of frames it is permitted to accept. The sender's window and the receiver's window need not have the same lower and upper limits, or even have the same size. In some protocols they are fixed in size, but in others they can grow or shrink as frames are sent and received.

Although these protocols give the data link layer more freedom about the order in which it may send and receive frames, we have most emphatically not dropped the requirement that the protocol must deliver packets to the destination

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network layer in the same order that they were passed to the data link layer on the sending machine. Nor have we changed the requirement that the physical communication channel is "wire-like," that is, it must deliver all frames in the order sent.

The sequence numbers within the sender's window represent frames sent but as yet not acknowledged. Whenever a new packet arrives from the network layer, it is given the next highest sequence number, and the upper edge of the window is advanced by one. When an acknowledgement comes in, the lower edge is advanced by one. In this way the window continuously maintains a list of unacknowledged frames.

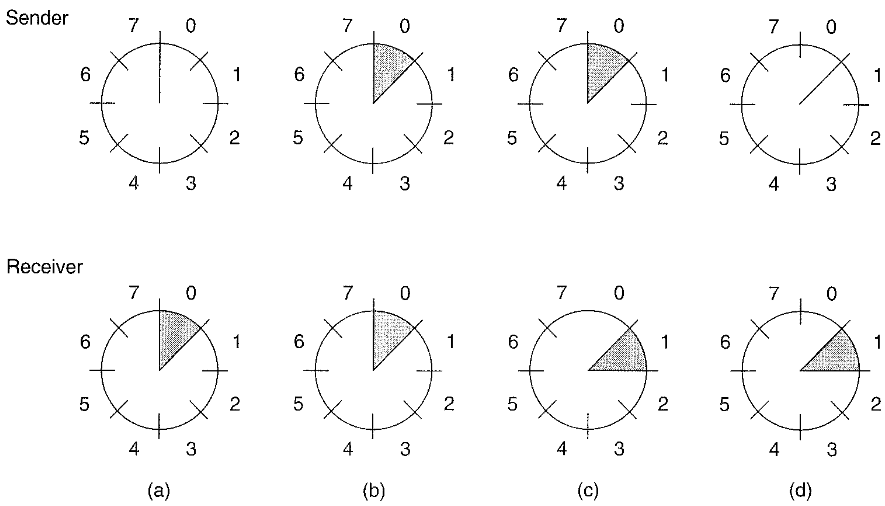

Fig. 3-12. A sliding window of size 1, with a 3-bit sequence number. (a) Initially. (b) After the first frame has been sent. (c) After the first frame has been received. (d) After the first acknowledgement has been received.

Since frames currently within the sender's window may ultimately be lost or damaged in transit, the sender must keep all these frames in its memory for possible retransmission. Thus if the maximum window size is n, the sender needs n buffers to hold the unacknowledged frames. If the window ever grows to its maximum size, the sending data link layer must forcibly shut off the network layer until another buffer becomes free.

The receiving data link layer's window corresponds to the frames it may accept. Any frame falling outside the window is discarded without comment. When a frame whose sequence number is equal to the lower edge of the window is received, it is passed to the network layer, an acknowledgement is generated, and the window is rotated by one. Unlike the sender's window, the receiver's

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```
/* Protocol 4 (sliding window) is bidirectional and is more robust than protocol 3. */
#define MAX_SEQ 1                          /* must be 1 for protocol 4 */
typedef enum {frame_arrival, cksum_err, timeout} event_type;
#include "protocol.h"
void protocol4 (void)
{
  seq_nr next_frame_to_send;               /* 0 or 1 only */
  seq_nr frame_expected;                   /* 0 or 1 only */
  frame r, s;                              /* scratch variables */
  packet buffer;                           /* current packet being sent */
  event_type event;
  next_frame_to_send = 0;                  /* next frame on the outbound stream */
  frame_expected = 0;                      /* number of frame arriving frame expected */
  from_network_layer(&buffer);             /* fetch a packet from the network layer */
  s.info = buffer;                         /* prepare to send the initial frame */
  s.seq = next_frame_to_send;              /* insert sequence number into frame */
  s.ack = 1 – frame_expected;              /* piggybacked ack */
  to_physical_layer(&s);                   /* transmit the frame */
  start_timer(s.seq);                      /* start the timer running */
  while (true) {
     wait_for_event(&event);               /* frame_arrival, cksum_err, or timeout */
     if (event == frame_arrival) { /* a frame has arrived undamaged. */
          from_physical_layer(&r);         /* go get it */
          if (r.seq == frame_expected) {
               /* Handle inbound frame stream. */
               to_network_layer(&r.info); /* pass packet to network layer */
               inc(frame_expected);        /* invert sequence number expected next */
          }
          if (r.ack == next_frame_to_send) { /* handle outbound frame stream. */
               from_network_layer(&buffer);    /* fetch new pkt from network layer */
               inc(next_frame_to_send);    /* invert sender's sequence number */
          }
     }
     s.info = buffer;                      /* construct outbound frame */
     s.seq = next_frame_to_send;           /* insert sequence number into it */
     s.ack = 1 – frame_expected;           /* seq number of last received frame */
     to_physical_layer(&s);                /* transmit a frame */
     start_timer(s.seq);                   /* start the timer running */
  }
}
```

Fig. 3-13. A 1-bit sliding window protocol.

window always remains at its initial size. Note that a window size of 1 means that the data link layer only accepts frames in order, but for larger windows this is not so. The network layer, in contrast, is always fed data in the proper order, regardless of the data link layer's window size.

Figure 3-12 shows an example with a maximum window size of 1. Initially, no frames are outstanding, so the lower and upper edges of the sender's window are equal, but as time goes on, the situation progresses as shown.

3.4.1. A One Bit Sliding Window Protocol

Before tackling the general case, let us first examine a sliding window protocol with a maximum window size of 1. Such a protocol uses stop-and-wait, since the sender transmits a frame and waits for its acknowledgement before sending the next one.

Figure 3-13 depicts such a protocol. Like the others, it starts out by defining some variables. *Next_frame_to_send* tells which frame the sender is trying to send. Similarly, *frame_expected* tells which frame the receiver is expecting. In both cases, 0 and 1 are the only possibilities.

Normally, one of the two data link layers goes first. In other words, only one of the data link layer programs should contain the *to_physical_layer* and *start_timer* procedure calls outside the main loop. In the event both data link layers start off simultaneously, a peculiar situation arises, which is discussed later. The starting machine fetches the first packet from its network layer, builds a frame from it, and sends it. When this (or any) frame arrives, the receiving data link layer checks to see if it is a duplicate, just as in protocol 3. If the frame is the one expected, it is passed to the network layer and the receiver's window is slid up.

The acknowledgement field contains the number of the last frame received without error. If this number agrees with the sequence number of the frame the sender is trying to send, the sender knows it is done with the frame stored in *buffer* and can fetch the next packet from its network layer. If the sequence number disagrees, it must continue trying to send the same frame. Whenever a frame is received, a frame is also sent back.

Now let us examine protocol 4 to see how resilient it is to pathological scenarios. Assume that A is trying to send its frame 0 to B and that B is trying to send its frame 0 to A. Suppose that A sends a frame to B, but A's timeout interval is a little too short. Consequently, A may time out repeatedly, sending a series of identical frames, all with $seq = 0$ and $ack = 1$.

When the first valid frame arrives at B, it will be accepted, and *frame_expected* will be set to 1. All the subsequent frames will be rejected, because B is now expecting frames with sequence number 1, not 0. Furthermore, since all the duplicates have $ack = 1$ and B is still waiting for an acknowledgement of 0, B will not fetch a new packet from its network layer.

After every rejected duplicate comes in, *B* sends *A* a frame containing *seq* = 0 and *ack* = 0. Eventually, one of these arrives correctly at *A*, causing *A* to begin sending the next packet. No combination of lost frames or premature timeouts can cause the protocol to deliver duplicate packets to either network layer, or to skip a packet, or to get into a deadlock.

However, a peculiar situation arises if both sides simultaneously send an initial packet. This synchronization difficulty is illustrated by Fig. 3-14. In part (a), the normal operation of the protocol is shown. In (b) the peculiarity is illustrated. If *B* waits for *A*'s first frame before sending one of its own, the sequence is as shown in (a), and every frame is accepted. However, if *A* and *B* simultaneously initiate communication, their first frames cross, and the data link layers then get into situation (b). In (a) each frame arrival brings a new packet for the network layer; there are no duplicates. In (b) half of the frames contain duplicates, even though there are no transmission errors. Similar situations can occur as a result of premature timeouts, even when one side clearly starts first. In fact, if multiple premature timeouts occur, frames may be sent three or more times.

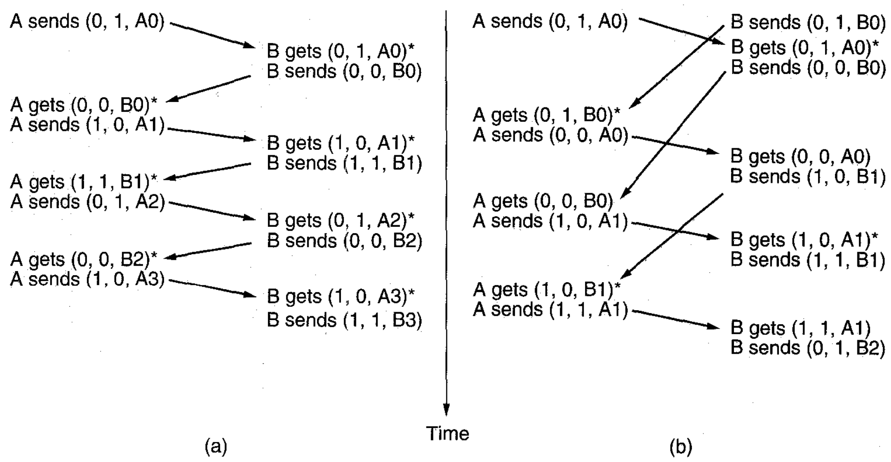
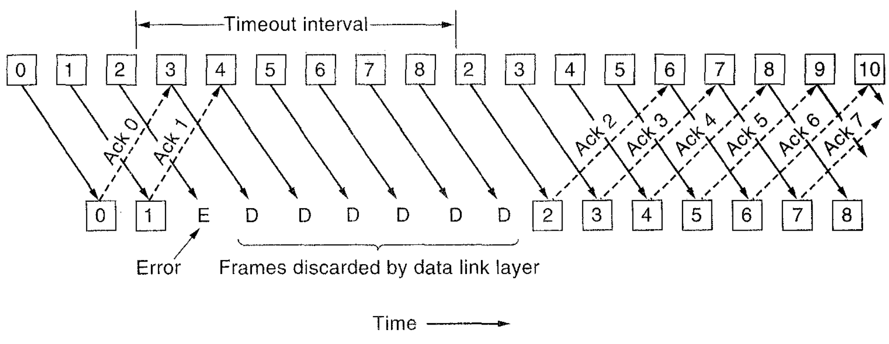


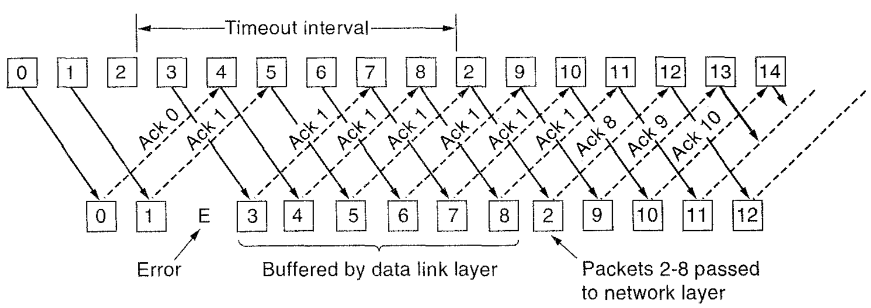
Fig. 3-14. Two scenarios for protocol 4. The notation is (seq, ack, packet number). An asterisk indicates where a network layer accepts a packet.

3.4.2. A Protocol Using Go Back n

Until now we have made the tacit assumption that the transmission time required for a frame to arrive at the receiver plus the transmission time for the acknowledgement to come back is negligible. Sometimes this assumption is clearly false. In these situations the long round-trip time can have important implications for the efficiency of the bandwidth utilization. As an example, consider a 50-kbps satellite channel with a 500-msec round-trip propagation delay. Let us imagine trying to use protocol 4 to send 1000-bit frames via the satellite. At $t = 0$ the sender starts sending the first frame. At $t = 20$ msec the frame has been completely sent. Not until $t = 270$ msec has the frame fully arrived at the receiver, and not until $t = 520$ msec has the acknowledgement arrived back at the sender, under the best of circumstances (no waiting in the receiver and a short acknowledgement frame). This means that the sender was blocked during 500/520 or 96 percent of the time (i.e., only 4 percent of the available bandwidth was used). Clearly, the combination of a long transit time, high bandwidth, and short frame length is disastrous in terms of efficiency.



(a)



(b)

Fig. 3-15. (a) Effect of an error when the receiver window size is 1. (b) Effect of an error when the receiver window size is large.

The problem described above can be viewed as a consequence of the rule requiring a sender to wait for an acknowledgement before sending another frame. If we relax that restriction, much better efficiency can be achieved. Basically the solution lies in allowing the sender to transmit up to w frames before blocking, instead of just 1. With an appropriate choice of w the sender will be able to continuously transmit frames for a time equal to the round-trip transit time without filling up the window. In the example above, w should be at least 26. The sender begins sending frame 0 as before. By the time it has finished sending 26 frames, at $t = 520$, the acknowledgement for frame 0 will have just arrived. Thereafter, acknowledgements will arrive every 20 msec, so the sender always gets permission to continue just when it needs it. At all times, 25 or 26 unacknowledged frames are outstanding. Put in other terms, the sender's maximum window size is 26.

This technique is known as **pipelining**. If the channel capacity is b bits/sec, the frame size l bits, and the round-trip propagation time R sec, the time required to transmit a single frame is l/b sec. After the last bit of a data frame has been sent, there is a delay of $R/2$ before that bit arrives at the receiver, and another delay of at least $R/2$ for the acknowledgement to come back, for a total delay of R. In stop-and-wait the line is busy for l/b and idle for R, giving a line utilization of $l/(l + bR)$. If $l < bR$ the efficiency will be less than 50 percent. Since there is always a nonzero delay for the acknowledgement to propagate back, in principle pipelining can be used to keep the line busy during this interval, but if the interval is small, the additional complexity is not worth the trouble.

Pipelining frames over an unreliable communication channel raises some serious issues. First, what happens if a frame in the middle of a long stream is damaged or lost? Large numbers of succeeding frames will arrive at the receiver before the sender even finds out that anything is wrong. When a damaged frame arrives at the receiver, it obviously should be discarded, but what should the receiver do with all the correct frames following it? Remember that the receiving data link layer is obligated to hand packets to the network layer in sequence.

There are two basic approaches to dealing with errors in the presence of pipelining. One way, called **go back n**, is for the receiver simply to discard all subsequent frames, sending no acknowledgements for the discarded frames. This strategy corresponds to a receive window of size 1. In other words, the data link layer refuses to accept any frame except the next one it must give to the network layer. If the sender's window fills up before the timer runs out, the pipeline will begin to empty. Eventually, the sender will time out and retransmit all unacknowledged frames in order, starting with the damaged or lost one. This approach, shown in Fig. 3-15(a) can waste a lot of bandwidth if the error rate is high.

The other general strategy for handling errors when frames are pipelined, called **selective repeat**, is to have the receiving data link layer store all the correct frames following the bad one. When the sender finally notices that something is wrong, it just retransmits the one bad frame, not all its successors, as shown in Fig. 3-15(b). If the second try succeeds, the receiving data link layer will now have many correct frames in sequence, so they can all be handed off to the network layer quickly and the highest number acknowledged.

This strategy corresponds to a receiver window larger than 1. Any frame within the window may be accepted and buffered until all the preceding ones have

```
/* Protocol 5 (pipelining) allows multiple outstanding frames. The sender may transmit up
   to MAX_SEQ frames without waiting for an ack. In addition, unlike the previous protocols,
   the network layer is not assumed to have a new packet all the time. Instead, the
   network layer causes a network_layer_ready event when there is a packet to send. */

#define MAX_SEQ 7                     /* should be 2^n - 1 */
typedef enum {frame_arrival, cksum_err, timeout, network_layer_ready} event_type;
#include "protocol.h"

static boolean between(seq_nr a, seq_nr b, seq_nr c)
{
/* Return true if (a <=b < c circularly; false otherwise. */
  if (((a <= b) && (b < c)) || ((c < a) && (a <= b)) || ((b < c) && (c < a)))
      return(true);
    else
      return(false);
}

static void send_data(seq_nr frame_nr, seq_nr frame_expected, packet buffer[])
{
/* Construct and send a data frame. */
  frame s;                            /* scratch variable */

  s.info = buffer[frame_nr];          /* insert packet into frame */
  s.seq = frame_nr;                   /* insert sequence number into frame */
  s.ack = (frame_expected + MAX_SEQ) % (MAX_SEQ + 1);/* piggyback ack */
  to_physical_layer(&s);              /* transmit the frame */
  start_timer(frame_nr);              /* start the timer running */
}

void protocol5(void)
{
  seq_nr next_frame_to_send;          /* MAX_SEQ > 1; used for outbound stream */
  seq_nr ack_expected;                /* oldest frame as yet unacknowledged */
  seq_nr frame_expected;              /* next frame expected on inbound stream */
  frame r;                            /* scratch variable */
  packet buffer[MAX_SEQ + 1];         /* buffers for the outbound stream */
  seq_nr nbuffered;                   /* # output buffers currently in use */
  seq_nr i;                           /* used to index into the buffer array */
  event_type event;

  enable_network_layer();             /* allow network_layer_ready events */
  ack_expected = 0;                   /* next ack expected inbound */
  next_frame_to_send = 0;             /* next frame going out */
  frame_expected = 0;                 /* number of frame expected inbound */
  nbuffered = 0;                      /* initially no packets are buffered */
```

```
while (true) {
  wait_for_event(&event);             /* four possibilities: see event_type above */

  switch(event) {
    case network_layer_ready:        /* the network layer has a packet to send */
         /* Accept, save, and transmit a new frame. */
         from_network_layer(&buffer[next_frame_to_send]); /* fetch new packet */
         nbuffered = nbuffered + 1;  /* expand the sender's window */
         send_data(next_frame_to_send, frame_expected, buffer);/* transmit the frame */
         inc(next_frame_to_send);    /* advance sender's upper window edge */
         break;

    case frame_arrival:              /* a data or control frame has arrived */
         from_physical_layer(&r);    /* get incoming frame from physical layer */

         if (r.seq == frame_expected) {
              /* Frames are accepted only in order. */
              to_network_layer(&r.info);  /* pass packet to network layer */
              inc(frame_expected);  /* advance lower edge of receiver's window */
         }

         /* Ack n implies n – 1, n – 2, etc.  Check for this. */
         while (between(ack_expected, r.ack, next_frame_to_send)) {
              /* Handle piggybacked ack. */
              nbuffered = nbuffered – 1; /* one frame fewer buffered */
              stop_timer(ack_expected); /* frame arrived intact; stop timer */
              inc(ack_expected);      /* contract sender's window */
         }
         break;

    case cksum_err: break;           /* just ignore bad frames */

    case timeout:                    /* trouble; retransmit all outstanding frames */
         next_frame_to_send = ack_expected;    /* start retransmitting here */
         for (i = 1; i <= nbuffered; i++) {
              send_data(next_frame_to_send, frame_expected, buffer);/* resend 1 frame */
              inc(next_frame_to_send);  /* prepare to send the next one */
         }

  }

  if (nbuffered < MAX_SEQ)
         enable_network_layer();
  else
         disable_network_layer();
 }
}
```

Fig. 3-16. A sliding window protocol using go back n.

been passed to the network layer. This approach can require large amounts of data link layer memory if the window is large.

These two alternative approaches are trade-offs between bandwidth and data link layer buffer space. Depending on which resource is more valuable, one or the other can be used. Figure 3-16 shows a pipelining protocol in which the receiving data link layer only accepts frames in order; frames following an error are discarded. In this protocol, for the first time, we have now dropped the assumption that the network layer always has an infinite supply of packets to send. When the network layer has a packet it wants to send, it can cause a *network_layer_ready* event to happen. However, in order to enforce the flow control rule of no more than *MAX_SEQ* unacknowledged frames outstanding at any time, the data link layer must be able to prohibit the network layer from bothering it with more work. The library procedures *enable_network_layer* and *disable_network_layer* perform this function.

Note that a maximum of *MAX_SEQ* frames and not *MAX_SEQ* + 1 frames may be outstanding at any instant, even though there are *MAX_SEQ* + 1 distinct sequence numbers: 0, 1, 2, . . . , *MAX_SEQ*. To see why this restriction is needed, consider the following scenario with *MAX_SEQ* = 7.

1. The sender sends frames 0 through 7.
2. A piggybacked acknowledgement for frame 7 eventually comes back to the sender.
3. The sender sends another eight frames, again with sequence numbers 0 through 7.
4. Now another piggybacked acknowledgement for frame 7 comes in.

The question is: Did all eight frames belonging to the second batch arrive successfully, or did all eight get lost (counting discards following an error as lost)? In both cases the receiver would be sending frame 7 as the acknowledgement. The sender has no way of telling. For this reason the maximum number of outstanding frames must be restricted to *MAX_SEQ*.

Although protocol 5 does not buffer the frames arriving after an error, it does not escape the problem of buffering altogether. Since a sender may have to retransmit all the unacknowledged frames at a future time, it must hang on to all transmitted frames until it knows for sure that they have been accepted by the receiver. When an acknowledgement comes in for frame n, frames $n - 1$, $n - 2$, and so on, are also automatically acknowledged. This property is especially important when some of the previous acknowledgement-bearing frames were lost or garbled. Whenever any acknowledgement comes in, the data link layer checks to see if any buffers can now be released. If buffers can be released (i.e., there is some room available in the window), a previously blocked network layer can now be allowed to cause more *network_layer_ready* events.

Because this protocol has multiple outstanding frames, it logically needs multiple timers, one per outstanding frame. Each frame times out independently of all the other ones. All of these timers can easily be simulated in software, using a single hardware clock that causes interrupts periodically. The pending timeouts form a linked list, with each node of the list telling how many clock ticks until the timer goes off, the frame being timed, and a pointer to the next node.

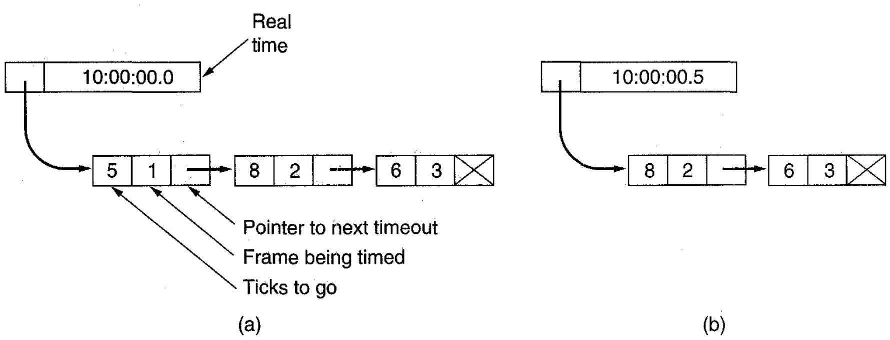


Fig. 3-17. Simulation of multiple timers in software.

As an illustration of how the timers could be implemented, consider the example of Fig. 3-17. Assume that the clock ticks once every 100 msec. Initially the real time is 10:00:00.0 and there are three timeouts pending, at 10:00:00.5, 10:00:01.3, and 10:00:01.9. Every time the hardware clock ticks, the real time is updated and the tick counter at the head of the list is decremented. When the tick counter becomes zero, a timeout is caused and the node removed from the list, as shown in Fig. 3-17(b). Although this organization requires the list to be scanned when *start_timer* or *stop_timer* is called, it does not require much work per tick. In protocol 5, both of these routines have been given a parameter, indicating which frame is to be timed.

3.4.3. A Protocol Using Selective Repeat

Protocol 5 works well if errors are rare, but if the line is poor it wastes a lot of bandwidth on retransmitted frames. An alternative strategy for handling errors is to allow the receiver to accept and buffer the frames following a damaged or lost one. Such a protocol does not discard frames merely because an earlier frame was damaged or lost.

In this protocol, both sender and receiver maintain a window of acceptable sequence numbers. The sender's window size starts out at 0 and grows to some predefined maximum, *MAX_SEQ*. The receiver's window, in contrast, is always fixed in size and equal to *MAX_SEQ*. The receiver has a buffer reserved for each sequence number within its window. Associated with each buffer is a bit (*arrived*) telling whether the buffer is full or empty. Whenever a frame arrives, its sequence number is checked by the function *between* to see if it falls within the window. If so, and if it has not already been received, it is accepted and stored. This action is taken without regard to whether or not it contains the next packet expected by the network layer. Of course, it must be kept within the data link layer and not passed to the network layer until all the lower numbered frames have already been delivered to the network layer in the correct order. A protocol using this algorithm is given in Fig. 3-18.

Nonsequential receive introduces certain problems not present in protocols in which frames are only accepted in order. We can illustrate the trouble most easily with an example. Suppose that we have a 3-bit sequence number, so that the sender is permitted to transmit up to seven frames before being required to wait for an acknowledgement. Initially the sender and receiver's windows are as shown in Fig. 3-19(a). The sender now transmits frames 0 through 6. The receiver's window allows it to accept any frame with sequence number between 0 and 6 inclusive. All seven frames arrive correctly, so the receiver acknowledges them and advance its window to allow receipt of 7, 0, 1, 2, 3, 4, or 5, as shown in Fig. 3-19(b). All seven buffers are marked empty.

It is at this point that disaster strikes in the form of a lightning bolt hitting the telephone pole and wiping out all the acknowledgements. The sender eventually times out and retransmits frame 0. When this frame arrives at the receiver, a check is made to see if it is within the receiver's window. Unfortunately, in Fig. 3-19(b) frame 0 is within the new window, so it will be accepted. The receiver sends a piggybacked acknowledgement for frame 6, since 0 through 6 have been received.

The sender is happy to learn that all its transmitted frames did actually arrive correctly, so it advances its window and immediately sends frames 7, 0, 1, 2, 3, 4, and 5. Frame 7 will be accepted by the receiver and its packet will be passed directly to the network layer. Immediately thereafter, the receiving data link layer checks to see if it has a valid frame 0 already, discovers that it does, and passes the embedded packet to the network layer. Consequently, the network layer gets an incorrect packet, and the protocol fails.

The essence of the problem is that after the receiver advanced its window, the new range of valid sequence numbers overlapped the old one. The following batch of frames might be either duplicates (if all the acknowledgements were lost) or new ones (if all the acknowledgements were received). The poor receiver has no way of distinguishing these two cases.

The way out of this dilemma lies in making sure that after the receiver has advanced its window, there is no overlap with the original window. To ensure that there is no overlap, the maximum window size should be at most half the range of the sequence numbers, as is done in Fig. 3-19(c) and Fig. 3-19(d). For example, if 4 bits are used for sequence numbers, these will range from 0 to 15.

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Only eight unacknowledged frames should be outstanding at any instant. That way, if the receiver has just accepted frames 0 through 7 and advanced its window to permit acceptance of frames 8 through 15, it can unambiguously tell if subsequent frames are retransmissions (0 through 7) or new ones (8 through 15). In general, the window size for protocol 6 will be $(MAX\_SEQ + 1)/2$.

An interesting question is: How many buffers must the receiver have? Under no conditions will it ever accept frames whose sequence numbers are below the lower edge of the window or frames whose sequence numbers are above the upper edge of the window. Consequently, the number of buffers needed is equal to the window size, not the range of sequence numbers. In the above example of a 4-bit sequence number, eight buffers, numbered 0 through 7, are needed. When frame i arrives, it is put in buffer i mod 8. Notice that although i and $(i + 8)$ mod 8 are "competing" for the same buffer, they are never within the window at the same time, because that would imply a window size of at least 9.

For the same reason, the number of timers needed is equal to the number of buffers, not the size of the sequence space. Effectively, there is a timer associated with each buffer. When the timer runs out, the contents of the buffer are retransmitted.

In protocol 5, there is an implicit assumption that the channel is heavily loaded. When a frame arrives, no acknowledgement is sent immediately. Instead, the acknowledgement is piggybacked onto the next outgoing data frame. If the reverse traffic is light, the acknowledgement will be held up for a long period of time. If there is a lot of traffic in one direction and no traffic in the other direction, only *MAX_SEQ* packets are sent, and then the protocol blocks.

In protocol 6 this problem is fixed. After an in-sequence data frame arrives, an auxiliary timer is started by *start_ack_timer*. If no reverse traffic has presented itself before this timer goes off, a separate acknowledgement frame is sent. An interrupt due to the auxiliary timer is called an *ack_timeout* event. With this arrangement, one-directional traffic flow is now possible, because the lack of reverse data frames onto which acknowledgements can be piggybacked is no longer an obstacle. Only one auxiliary timer exists, and if *start_ack_timer* is called while the timer is running, it is reset to a full acknowledgement timeout interval.

It is essential that the timeout associated with the auxiliary timer be appreciably shorter than the timer used for timing out data frames. This condition is required to make sure that the acknowledgement for a correctly received frame arrives before the sender times out and retransmits the frame.

Protocol 6 uses a more efficient strategy than protocol 5 for dealing with errors. Whenever the receiver has reason to suspect that an error has occurred, it sends a negative acknowledgement (NAK) frame back to the sender. Such a frame is a request for retransmission of the frame specified in the NAK. There are two cases when the receiver should be suspicious: a damaged frame has arrived or a frame other than the expected one arrived (potential lost frame). To avoid making

```
/* Protocol 6 (nonsequential receive) accepts frames out of order, but passes packets to the
   network layer in order. Associated with each outstanding frame is a timer. When the timer
   goes off, only that frame is retransmitted, not all the outstanding frames, as in protocol 5. */

#define MAX_SEQ 7                          /* should be 2^n - 1 */
#define NR_BUFS ((MAX_SEQ + 1)/2)
typedef enum {frame_arrival, cksum_err, timeout, network_layer_ready, ack_timeout} event_type;
#include "protocol.h"
boolean no_nak = true;                     /* no nak has been sent yet */
seq_nr oldest_frame = MAX_SEQ + 1;         /* initial value is only for the simulator */

static boolean between(seq_nr a, seq_nr b, seq_nr c)
{
/* Same as between in protocol5, but shorter and more obscure. */
  return ((a <= b) && (b < c)) || ((c < a) && (a <= b)) || ((b < c) && (c < a));
}

static void send_frame(frame_kind fk, seq_nr frame_nr, seq_nr frame_expected, packet buffer[])
{
/* Construct and send a data, ack, or nak frame. */
  frame s;                                 /* scratch variable */

  s.kind = fk;                             /* kind == data, ack, or nak */
  if (fk == data) s.info = buffer[frame_nr % NR_BUFS];
  s.seq = frame_nr;                        /* only meaningful for data frames */
  s.ack = (frame_expected + MAX_SEQ) % (MAX_SEQ + 1);
  if (fk == nak) no_nak = false;           /* one nak per frame, please */
  to_physical_layer(&s);                   /* transmit the frame */
  if (fk == data) start_timer(frame_nr % NR_BUFS);
  stop_ack_timer();                        /* no need for separate ack frame */
}

void protocol6(void)
{
  seq_nr ack_expected;                     /* lower edge of sender's window */
  seq_nr next_frame_to_send;               /* upper edge of sender's window + 1 */
  seq_nr frame_expected;                   /* lower edge of receiver's window */
  seq_nr too_far;                          /* upper edge of receiver's window + 1 */
  int i;                                   /* index into buffer pool */
  frame r;                                 /* scratch variable */
  packet out_buf[NR_BUFS];                 /* buffers for the outbound stream */
  packet in_buf[NR_BUFS];                  /* buffers for the inbound stream */
  boolean arrived[NR_BUFS];                /* inbound bit map */
  seq_nr nbuffered;                        /* how many output buffers currently used */
  event_type event;

  enable_network_layer();                  /* initialize */
  ack_expected = 0;                        /* next ack expected on the inbound stream */
  next_frame_to_send = 0;                  /* number of next outgoing frame */
  frame_expected = 0;
  too_far = NR_BUFS;
  nbuffered = 0;                           /* initially no packets are buffered */
  for (i = 0; i < NR_BUFS; i++) arrived[i] = false;
```

```
    wait_for_event(&event);                  /* five possibilities: see event_type above */
    switch(event) {
      case network_layer_ready:              /* accept, save, and transmit a new frame */
           nbuffered = nbuffered + 1;        /* expand the window */
           from_network_layer(&out_buf[next_frame_to_send % NR_BUFS]); /* fetch new packet */
           send_frame(data, next_frame_to_send, frame_expected, out_buf);/* transmit the frame */
           inc(next_frame_to_send);          /* advance upper window edge */
           break;

      case frame_arrival:                    /* a data or control frame has arrived */
           from_physical_layer(&r);          /* fetch incoming frame from physical layer */
           if (r.kind == data) {
                /* An undamaged frame has arrived. */
                if ((r.seq != frame_expected) && no_nak)
                   send_frame(nak, 0, frame_expected, out_buf); else start_ack_timer();
                if (between(frame_expected, r.seq, too_far) && (arrived[r.seq%NR_BUFS] == false)) {
                     /* Frames may be accepted in any order. */
                     arrived[r.seq % NR_BUFS] = true;    /* mark buffer as full */
                     in_buf[r.seq % NR_BUFS] = r.info;   /* insert data into buffer */
                     while (arrived[frame_expected % NR_BUFS]) {
                          /* Pass frames and advance window. */
                          to_network_layer(&in_buf[frame_expected % NR_BUFS]);
                          no_nak = true;
                          arrived[frame_expected % NR_BUFS] = false;
                          inc(frame_expected);   /* advance lower edge of receiver's window */
                          inc(too_far);          /* advance upper edge of receiver's window */
                          start_ack_timer();     /* to see if a separate ack is needed */
                     }
                }
           }
           if((r.kind==nak) && between(ack_expected,(r.ack+1)%(MAX_SEQ+1),next_frame_to_send))
                send_frame(data, (r.ack+1) % (MAX_SEQ + 1), frame_expected, out_buf);

           while (between(ack_expected, r.ack, next_frame_to_send)) {
                nbuffered = nbuffered - 1;        /* handle piggybacked ack */
                stop_timer(ack_expected % NR_BUFS);   /* frame arrived intact */
                inc(ack_expected);                /* advance lower edge of sender's window */
           }
           break;

      case cksum_err:
           if (no_nak) send_frame(nak, 0, frame_expected, out_buf);/* damaged frame */
           break;

      case timeout:
           send_frame(data, oldest_frame, frame_expected, out_buf);/* we timed out */
           break;

      case ack_timeout:
           send_frame(ack,0,frame_expected, out_buf);    /* ack timer expired; send ack */
    }

    if (nbuffered < NR_BUFS) enable_network_layer(); else disable_network_layer();
  }
}
```

Fig. 3-18. A sliding window protocol using selective repeat.

multiple requests for retransmission of the same lost frame, the receiver should keep track of whether a NAK has already been sent for a given frame. The variable *no_nak* in protocol 6 is true if no NAK has been sent yet for *frame_expected*. If the NAK gets mangled or lost, no real harm is done, since the sender will eventually time out and retransmit the missing frame anyway. If the wrong frame arrives after a NAK has been sent and lost, *no_nak* will be true and the auxiliary timer will be started. When it goes off, an ACK will be sent to resynchronize the sender to the receiver's current status.

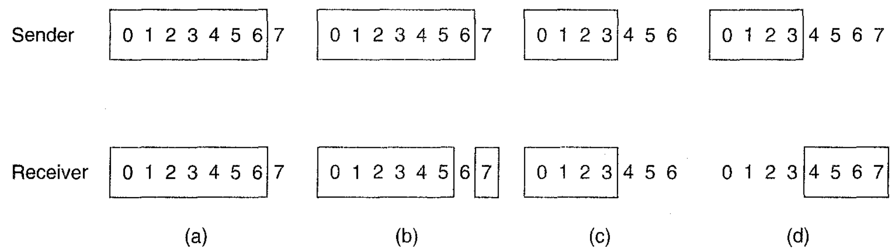

Fig. 3-19. (a) Initial situation with a window of size seven. (b) After seven frames have been sent and received but not acknowledged. (c) Initial situation with a window size of four. (d) After four frames have been sent and received but not acknowledged.

In some situations, the time required for a frame to propagate to the destination, be processed there, and have the acknowledgement come back is (nearly) constant. In these situations, the sender can adjust its timer to be just slightly larger than the normal time interval expected between sending a frame and receiving its acknowledgement. However, if this time is highly variable, the sender is faced with the choice of either setting the interval to a small value and risking unnecessary retransmissions, thus wasting bandwidth, or setting it to a large value, going idle for a long period after an error, thus also wasting bandwidth. If the reverse traffic is sporadic, the time before acknowledgement will be irregular, being shorter when there is reverse traffic and longer when there is not. Variable processing time within the receiver can also be a problem here. In general, whenever the standard deviation of the acknowledgement interval is small compared to the interval itself, the timer can be set "tight" and NAKs are not useful. Otherwise, the timer must be set "loose," and NAKs can appreciably speed up retransmission of lost or damaged frames.

Closely related to the matter of timeouts and NAKs is the question of determining which frame caused a timeout. In protocol 5 it is always *ack_expected*, because it is always the oldest. In protocol 6, there is no trivial way to determine who timed out. Suppose that frames 0 through 4 have been transmitted, meaning that the list of outstanding frames is 01234, in order from oldest to youngest. Now imagine that 0 times out, 5 (a new frame) is transmitted, 1 times out, 2 times

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out, and 6 (another new frame) is transmitted. At this point the list of outstanding frames is 3405126, from oldest to youngest. If all inbound traffic is lost for a while, the seven outstanding frames will time out in that order. To keep the example from getting even more complicated than it already is, we have not shown the timer administration. Instead, we just assume that the variable *oldest_frame* is set upon timeout to indicate which frame timed out.

3.5. PROTOCOL SPECIFICATION AND VERIFICATION

Realistic protocols, and the programs that implement them, are often quite complicated. Consequently, much research has been done trying to find formal, mathematical techniques for specifying and verifying protocols. In the following sections we will look at some models and techniques. Although we are looking at them in the context of the data link layer, they are also applicable to other layers.

3.5.1. Finite State Machine Models

A key concept used in many protocol models is the **finite state machine**. With this technique, each **protocol machine** (i.e., sender or receiver) is always in a specific state at every instant of time. Its state consists of all the values of its variables, including the program counter.

In most cases, a large number of states can be grouped together for purposes of analysis. For example, considering the receiver in protocol 3, we could abstract out from all the possible states two important ones: waiting for frame 0 or waiting for frame 1. All other states can be thought of as transient, just steps on the way to one of the main states. Typically, the states are chosen to be those instants that the protocol machine is waiting for the next event to happen [i.e., executing the procedure call *wait*(*event*) in our examples]. At this point the state of the protocol machine is completely determined by the states of its variables. The number of states is then 2^n, where n is the number of bits needed to represent all the variables combined.

The state of the complete system is the combination of all the states of the two protocol machines and the channel. The state of the channel is determined by its contents. Using protocol 3 again as an example, the channel has four possible states: a zero frame or a one frame moving from sender to receiver, an acknowledgement frame going the other way, or an empty channel. If we model the sender and receiver as each having two states, the complete system has 16 distinct states.

A word about the channel state is in order. The concept of a frame being "on the channel" is an abstraction, of course. What we really mean is that a frame has been partially transmitted, partially received, but not yet processed at the destination. A frame remains "on the channel" until the protocol machine executes *FromPhysicalLayer* and processes it.

From each state, there are zero or more possible **transitions** to other states. Transitions occur when some event happens. For a protocol machine a transition might occur when a frame is sent, when a frame arrives, when a timer goes off, when an interrupt occurs, etc. For the channel, typical events are insertion of a new frame onto the channel by a protocol machine, delivery of a frame to a protocol machine, or loss of a frame due to a noise burst. Given a complete description of the protocol machines and the channel characteristics, it is possible to draw a directed graph showing all the states as nodes and all the transitions as directed arcs.

One particular state is designated as the **initial state**. This state corresponds to the description of the system when it starts running, or some convenient starting place shortly thereafter. From the initial state, some, perhaps all, of the other states can be reached by a sequence of transitions. Using well-known techniques from graph theory (e.g., computing the transitive closure of a graph), it is possible to determine which states are reachable and which are not. This technique is called **reachability analysis** (Lin et al., 1987). This analysis can be helpful in determining if a protocol is correct or not.

Formally, a finite state machine model of a protocol can be regarded as a quadruple (S, M, I, T) where:

S is the set of states the processes and channel can be in.

M is the set of frames that can be exchanged over the channel.

I is the set of initial states of the processes.

T is the set of transitions between states.

At the beginning of time, all processes are in their initial states. Then events begin to happen, such as frames becoming available for transmission or timers going off. Each event may cause one of the processes or the channel to take an action and switch to a new state. By carefully enumerating each possible successor to each state, one can build the reachability graph and analyze the protocol.

Reachability analysis can be used to detect a variety of errors in the protocol specification. For example, if it is possible for a certain frame to occur in a certain state and the finite state machine does not say what action should be taken, the specification is in error (incompleteness). If there exists a set of states from which there is no exit and from which no progress can be made (correct frames received), we have another error (deadlock). A less serious error is protocol specification that tells how to handle an event in a state in which the event cannot occur (extraneous transition). Other errors can also be detected.

As an example of a finite state machine model, consider Fig. 3-20(a). This graph corresponds to protocol 3 as described above: each protocol machine has

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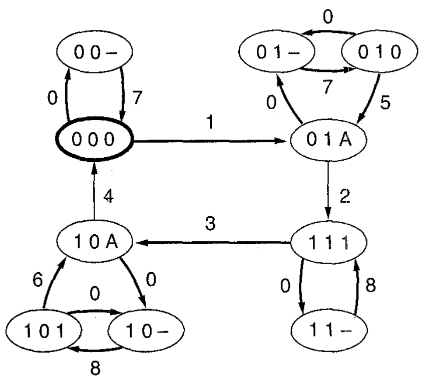
two states and the channel has four states. A total of 16 states exist, not all of them reachable from the initial one. The unreachable ones are not shown in the figure. Each state is labeled by three characters, *XYZ*, where *X* is 0 or 1, corresponding to the frame the sender is trying to send; *Y* is also 0 or 1, corresponding to the frame the receiver expects, and *Z* is 0, 1, *A*, or empty (–), corresponding to the state of the channel. In this example the initial state has been chosen as (000). In other words, the sender has just sent frame 0, the receiver expects frame 0, and frame 0 is currently on the channel.



(a)

Transition	Who runs?	Frame accepted	Frame emitted	To network layer
0	–	(frame lost)		–
1	R	0	A	Yes
2	S	A	1	–
3	R	1	A	Yes
4	S	A	0	–
5	R	0	A	No
6	R	1	A	No
7	S	(timeout)	0	–
8	S	(timeout)	1	–

(b)

Fig. 3-20. (a) State diagram for protocol 3. (b) Transitions.

Nine kinds of transitions are shown in Fig. 3-20. Transition 0 consists of the channel losing its contents. Transition 1 consists of the channel correctly delivering packet 0 to the receiver, with the receiver then changing its state to expect frame 1 and emitting an acknowledgement. Transition 1 also corresponds to the receiver delivering packet 0 to the network layer. The other transitions are listed in Fig. 3-20(b). The arrival of a frame with a checksum error has not been shown because it does not change the state (in protocol 3).

During normal operation, transitions 1, 2, 3, and 4 are repeated in order over and over. In each cycle, two packets are delivered, bringing the sender back to the initial state of trying to send a new frame with sequence number 0. If the channel loses frame 0, it makes a transition from state (000) to state (00–). Eventually, the sender times out (transition 7) and the system moves back to (000). The loss of an acknowledgement is more complicated, requiring two transitions, 7 and 5, or 8 and 6, to repair the damage.

One of the properties that a protocol with a 1-bit sequence number must have is that no matter what sequence of events happens, the receiver never delivers two odd packets without an intervening even packet, and vice versa. From the graph of Fig. 3-20 we see that this requirement can be stated more formally as "there

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must not exist any paths from the initial state on which two occurrences of transition 1 occur without an occurrence of transition 3 between them, or vice versa." From the figure it can be seen that the protocol is correct in this respect.

Another, similar requirement is that there not be any paths on which the sender changes state twice (e.g., from 0 to 1 and back to 0) while the receiver state remains constant. Were such a path to exist, then in the corresponding sequence of events two frames would be irretrievably lost, without the receiver noticing. The packet sequence delivered would have an undetected gap of two packets in it.

Yet another important property of a protocol is the absence of deadlocks. A **deadlock** is a situation in which the protocol can make no more forward progress (i.e., deliver packets to the network layer) no matter what sequence of events happen. In terms of the graph model, a deadlock is characterized by the existence of a subset of states that is reachable from the initial state and which has two properties:

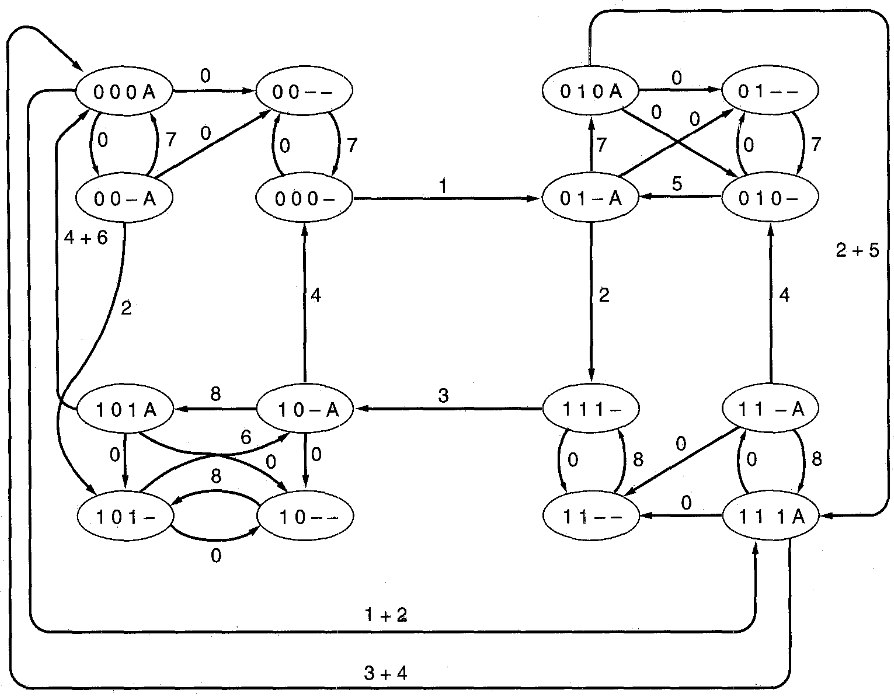
1. There is no transition out of the subset.
2. There are no transitions in the subset that cause forward progress.

Once in the deadlock situation, the protocol remains there forever. Again, it is easy to see from the graph that protocol 3 does not suffer from deadlocks.

Now let us consider a variation of protocol 3, one in which the half-duplex channel is replaced by a full-duplex channel. In Fig. 3-21 we show the states as the product of the states of the two protocol machines and the states of the two channels. Note that the forward channel now has three states: frame 0, frame 1, or empty, and the reverse channel has two states, *A* or empty. The transitions are the same as in Fig. 3-20(b), except that when a data frame and an acknowledgement are on the channel simultaneously, there is a slight peculiarity. The receiver cannot remove the data frame by itself, because that would entail having two acknowledgements on the channel at the same time, something not permitted in our model (although it is easy to devise a model that does allow it). Similarly, the sender cannot remove the acknowledgement, because that would entail emitting a second data frame before the first had been accepted. Consequently, both events must occur together, for example, the transition between state (000A) and state (111A), labeled as 1 + 2 in the figure.

In Fig. 3-21(a) there exist paths that cause the protocol to fail. In particular, there are paths in which the sender repeatedly fetches new packets, even though the previous ones have not been delivered correctly. The problem arises because it is now possible for the sender to time out and send a new frame without disturbing the acknowledgement on the reverse channel. When this acknowledgement arrives, it will be mistakenly regarded as referring to the current transmission and not the previous one.

One state sequence causing the protocol to fail is shown in Fig. 3-21(b). In



(a)

(0 0 0 –), (0 1 – A), (0 1 0 A), (1 1 1 A), (1 1 – A), (0 1 0 –), (0 1 – A), (1 1 1 –)

(b)

Fig. 3-21. (a) State graph for protocol 3 and a full-duplex channel. (b) Sequence of states causing the protocol to fail.

the fourth and sixth states of this sequence, the sender changes state, indicating that it fetches a new packet from the network layer, while the receiver does not change state, that is, does not deliver any packets to the network layer.

3.5.2. Petri Net Models

The finite state machine is not the only technique for formally specifying protocols. In this section we will describe another technique, the **Petri Net** (Danthine, 1980). A Petri net has four basic elements: places, transitions, arcs, and tokens. A **place** represents a state which (part of) the system may be in. Figure 3-22 shows a Petri net with two places, *A* and *B*, both shown as circles. The

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system is currently in state A, indicated by the **token** (heavy dot) in place A. A **transition** is indicated by a horizontal or vertical bar. Each transition has zero or more **input arcs**, coming from its input places, and zero or more **output arcs**, going to its output places.

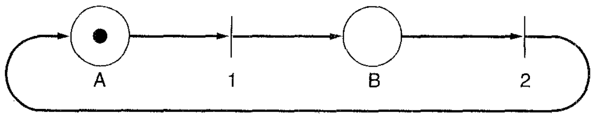

Fig. 3-22. A Petri net with two places and two transitions.

A transition is **enabled** if there is at least one input token in each of its input places. Any enabled transition may **fire** at will, removing one token from each input place and depositing a token in each output place. If the number of input arcs and output arcs differ, tokens will not be conserved. If two or more transitions are enabled, any one of them may fire. The choice of a transition to fire is indeterminate, which is why Petri nets are useful for modeling protocols. The Petri net of Fig. 3-22 is deterministic and can be used to model any two-phase process (e.g., the behavior of a baby: eat, sleep, eat, sleep, and so on). As with all modeling tools, unnecessary detail is suppressed.

Figure 3-23 gives the Petri net model of Fig. 3-21. Unlike the finite state machine model, there are no composite states here; the sender's state, channel state, and receiver's state are represented separately. Transitions 1 and 2 correspond to transmission of frame 0 by the sender, normally, and on a timeout respectively. Transitions 3 and 4 are analogous for frame 1. Transitions 5, 6, and 7 correspond to the loss of frame 0, an acknowledgement, and frame 1, respectively. Transitions 8 and 9 occur when a data frame with the wrong sequence number arrives at the receiver. Transitions 10 and 11 represent the arrival at the receiver of the next frame in sequence and its delivery to the network layer.

Petri nets can be used to detect protocol failures in a way similar to the use of finite state machines. For example, if some firing sequence included transition 10 twice without transition 11 intervening, the protocol would be incorrect. The concept of a deadlock in a Petri net is also similar to its finite state machine counterpart.

Petri nets can be represented in convenient algebraic form resembling a grammar. Each transition contributes one rule to the grammar. Each rule specifies the input and output places of the transition, for example, transition 1 in Fig. 3-23 is $BD \rightarrow AC$. The current state of the Petri net is represented as an unordered collection of places, each place represented in the collection as many times as it has tokens. Any rule all of whose left-hand side places are present, can be fired, removing those places from the current state, and adding its output places to the current state. The marking of Fig. 3-23 is ACG, so rule 10 ($CG \rightarrow DF$) can be applied but rule 3 ($AD \rightarrow BE$) cannot be applied.

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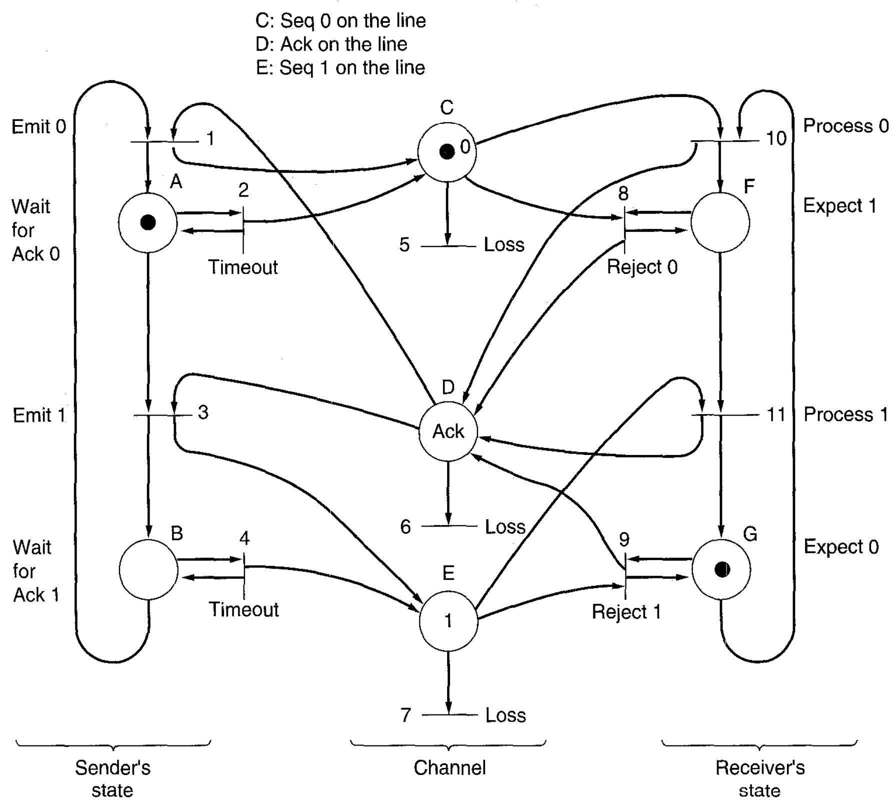

Fig. 3-23. A Petri net model for protocol 3.

3.6. EXAMPLE DATA LINK PROTOCOLS

In the following sections we will examine several widely-used data link protocols. The first one, HDLC, is common in X.25 and many other networks. After that, we will examine data link protocols used in the Internet and ATM networks, respectively. In subsequent chapters, we will also use the Internet and ATM as running examples as well.

3.6.1. HDLC—High-level Data Link Control

In this section we will examine a group of closely related protocols that are a bit old but are still heavily used in networks throughout the world. They are all derived from the data link protocol used in IBM's SNA, called **SDLC**

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(**Synchronous Data Link Control**) protocol. After developing SDLC, IBM submitted it to ANSI and ISO for acceptance as U.S. and international standards, respectively. ANSI modified it to become **ADCCP** (**Advanced Data Communication Control Procedure**), and ISO modified it to become **HDLC** (**High-level Data Link Control**). CCITT then adopted and modified HDLC for its **LAP** (**Link Access Procedure**) as part of the X.25 network interface standard but later modified it again to **LAPB**, to make it more compatible with a later version of HDLC. The nice thing about standards is that you have so many to choose from. Furthermore, if you do not like any of them, you can just wait for next year's model.

All of these protocols are based on the same principles. All are bit-oriented, and all use bit stuffing for data transparency. They differ only in minor, but nevertheless irritating, ways. The discussion of bit-oriented protocols that follows is intended as a general introduction. For the specific details of any one protocol, please consult the appropriate definition.

All the bit-oriented protocols use the frame structure shown in Fig. 3-24. The *Address* field is primarily of importance on lines with multiple terminals, where it is used to identify one of the terminals. For point-to-point lines, it is sometimes used to distinguish commands from responses.

Bits	8	8	8	⩾ 0	16	8
	0 1 1 1 1 1 1 0	Address	Control	Data	Checksum	0 1 1 1 1 1 1 0

Fig. 3-24. Frame format for bit-oriented protocols.

The *Control* field is used for sequence numbers, acknowledgements, and other purposes, as discussed below.

The *Data* field may contain arbitrary information. It may be arbitrarily long, although the efficiency of the checksum falls off with increasing frame length due to the greater probability of multiple burst errors.

The *Checksum* field is a minor variation on the well-known cyclic redundancy code, using CRC-CCITT as the generator polynomial. The variation is to allow lost flag bytes to be detected.

The frame is delimited with another flag sequence (01111110). On idle point-to-point lines, flag sequences are transmitted continuously. The minimum frame contains three fields and totals 32 bits, excluding the flags on either end.

There are three kinds of frames: **Information**, **Supervisory**, and **Unnumbered**. The contents of the *Control* field for these three kinds are shown in Fig. 3-25. The protocol uses a sliding window, with a 3-bit sequence number. Up to seven unacknowledged frames may be outstanding at any instant. The *Seq* field in Fig. 3-25(a) is the frame sequence number. The *Next* field is a piggybacked

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acknowledgement. However, all the protocols adhere to the convention that instead of piggybacking the number of the last frame received correctly, they use the number of the first frame not received (i.e., the next frame expected). The choice of using the last frame received or the next frame expected is arbitrary; it does not matter which convention is used, provided that it is used consistently.

Bits	1	3		1	3
(a)	0	Seq		P/F	Next
(b)	1	0	Type	P/F	Next
(c)	1	1	Type	P/F	Modifier

Fig. 3-25. Control field of (a) an information frame, (b) a supervisory frame, (c) an unnumbered frame.

The *P/F* bit stands for *Poll/Final.* It is used when a computer (or concentrator) is polling a group of terminals. When used as *P*, the computer is inviting the terminal to send data. All the frames sent by the terminal, except the final one, have the *P/F* bit set to *P*. The final one is set to *F*.

In some of the protocols, the *P/F* bit is used to force the other machine to send a Supervisory frame immediately rather than waiting for reverse traffic onto which to piggyback the window information. The bit also has some minor uses in connection with the Unnumbered frames.

The various kinds of Supervisory frames are distinguished by the *Type* field. Type 0 is an acknowledgement frame (officially called RECEIVE READY) used to indicate the next frame expected. This frame is used when there is no reverse traffic to use for piggybacking.

Type 1 is a negative acknowledgement frame (officially called REJECT). It is used to indicate that a transmission error has been detected. The *Next* field indicates the first frame in sequence not received correctly (i.e., the frame to be retransmitted). The sender is required to retransmit all outstanding frames starting at *Next*. This strategy is similar to our protocol 5 rather than our protocol 6.

Type 2 is RECEIVE NOT READY. It acknowledges all frames up to but not including *Next*, just as RECEIVE READY, but it tells the sender to stop sending. RECEIVE NOT READY is intended to signal certain temporary problems with the receiver, such as a shortage of buffers, and not as an alternative to the sliding window flow control. When the condition has been repaired, the receiver sends a RECEIVE READY, REJECT, or certain control frames.

Type 3 is the SELECTIVE REJECT. It calls for retransmission of only the frame specified. In this sense it is like our protocol 6 rather than 5 and is therefore most

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useful when the sender's window size is half the sequence space size, or less. Thus if a receiver wishes to buffer out of sequence frames for potential future use, it can force the retransmission of any specific frame using Selective Reject. HDLC and ADCCP allow this frame type, but SDLC and LAPB do not allow it (i.e., there is no Selective Reject), and type 3 frames are undefined.

The third class of frame is the Unnumbered frame. It is sometimes used for control purposes but can also be used to carry data when unreliable connectionless service is called for. The various bit-oriented protocols differ considerably here, in contrast with the other two kinds, where they are nearly identical. Five bits are available to indicate the frame type, but not all 32 possibilities are used.

All the protocols provide a command, DISC (DISConnect), that allows a machine to announce that it is going down (e.g., for preventive maintenance). They also have a command that allows a machine that has just come back on-line to announce its presence and force all the sequence numbers back to zero. This command is called SNRM (Set Normal Response Mode). Unfortunately, "Normal Response Mode" is anything but normal. It is an unbalanced (i.e., asymmetric) mode in which one end of the line is the master and the other the slave. SNRM dates from a time when data communication meant a dumb terminal talking to a computer, which clearly is asymmetric. To make the protocol more suitable when the two partners are equals, HDLC and LAPB have an additional command, SABM (Set Asynchronous Balanced Mode), which resets the line and declares both parties to be equals. They also have commands SABME and SNRME, which are the same as SABM and SNRM, respectively, except that they enable an extended frame format that uses 7-bit sequence numbers instead of 3-bit sequence numbers.

A third command provided by all the protocols is FRMR (FRaMe Reject), used to indicate that a frame with a correct checksum but impossible semantics arrived. Examples of impossible semantics are a type 3 Supervisory frame in LAPB, a frame shorter than 32 bits, an illegal control frame, and an acknowledgement of a frame that was outside the window, etc. FRMR frames contain a 24-bit data field telling what was wrong with the frame. The data include the control field of the bad frame, the window parameters, and a collection of bits used to signal specific errors.

Control frames may be lost or damaged, just like data frames, so they must be acknowledged too. A special control frame is provided for this purpose, called UA (Unnumbered Acknowledgement). Since only one control frame may be outstanding, there is never any ambiguity about which control frame is being acknowledged.

The remaining control frames deal with initialization, polling, and status reporting. There is also a control frame that may contain arbitrary information, UI (Unnumbered Information). These data are not passed to the network layer but are for the receiving data link layer itself.

Despite its widespread use, HDLC is far from perfect. A discussion of a variety of problems associated with it can be found in (Fiorini et al., 1995).

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3.6.2. The Data Link Layer in the Internet

The Internet consists of individual machines (hosts and routers), and the communication infrastructure that connects them. Within a single building, LANs are widely used for interconnection, but most of the wide area infrastructure is built up from point-to-point leased lines. In Chap. 4, we will look at LANs; here we will examine the data link protocols used on point-to-point lines in the Internet.

In practice, point-to-point communication is primarily used in two situations. First, thousands of organizations have one or more LANs, each with some number of hosts (personal computers, user workstations, servers, and so on) along with a router (or a bridge, which is functionally similar). Often, the routers are interconnected by a backbone LAN. Typically, all connections to the outside world go through one or two routers that have point-to-point leased lines to distant routers. It is these routers and their leased lines that make up the communication subnets on which the Internet is built.

The second situation where point-to-point lines play a major role in the Internet is the millions of individuals who have home connections to the Internet using modems and dial-up telephone lines. Usually, what happens is that the user's home PC calls up an **Internet provider**, which includes commercial companies like America Online, CompuServe, and the Microsoft Network, but also many universities and companies that provide home Internet connectivity to their students and employees. Sometimes the home PC just functions as a character-oriented terminal logged into the Internet service provider's timesharing system. In this mode, the user can type commands and run programs, but the graphical Internet services, such as the World Wide Web, are not available. This way of working is called having a **shell account**.

Alternatively, the home PC can call an Internet service provider's router and then act like a full-blown Internet host. This method of operation is no different than having a leased line between the PC and the router, except that the connection is terminated when the user ends the session. With this approach, all Internet services, including the graphical ones, become available. A home PC calling an Internet service provider is illustrated in Fig. 3-26.

For both the router-router leased line connection and the dial-up host-router connection, some point-to-point data link protocol is required on the line for framing, error control, and the other data link layer functions we have studied in this chapter. Two such protocols are widely used in the Internet, SLIP and PPP. We will now examine each of these in turn.

SLIP—Serial Line IP

SLIP is the older of the two protocols. It was devised by Rick Adams in 1984 to connect Sun workstations to the Internet over a dial-up line using a modem. The protocol, which is described in RFC 1055, is very simple. The workstation

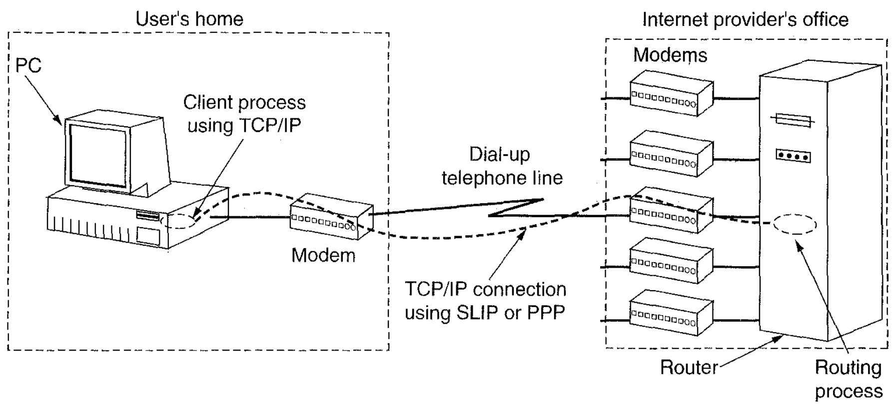

Fig. 3-26. A home personal computer acting as an Internet host.

just sends raw IP packets over the line, with a special flag byte (0xC0) at the end for framing. If the flag byte occurs inside the IP packet, a form of character stuffing is used, and the two byte sequence (0xDB, 0xDC) is sent in its place. If 0xDB occurs inside the IP packet, it, too, is stuffed. Some SLIP implementations attach a flag byte to both the front and back of each IP packet sent.

More recent versions of SLIP do some TCP and IP header compression. What they do is take advantage of the fact that consecutive packets often have many header fields in common. These are compressed by omitting those fields that are the same as the corresponding fields in the previous IP packet. Furthermore, the fields that do differ are not sent in their entirety, but as increments to the previous value. These optimizations are described in RFC 1144.

Although it is still widely used, SLIP has some serious problems. First, it does not do any error detection or correction, so it is up to higher layers to detect and recover from lost, damaged, or merged frames.

Second, SLIP supports only IP. With the growth of the Internet to encompass networks that do not use IP as their native language (e.g., Novell LANs), this restriction is becoming increasingly serious.

Third, each side must know the other's IP address in advance; neither address can be dynamically assigned during setup. Given the current shortage of IP addresses, this limitation is a major issue as it is impossible to give each home Internet user a unique IP address.

Fourth, SLIP does not provide any form of authentication, so neither party knows whom it is really talking to. With leased lines, this is not an issue, but with dial-up lines it is.

Fifth, SLIP is not an approved Internet Standard, so many different (and incompatible) versions exist. This situation does not make interworking easier.

PPP—Point-to-Point Protocol

To improve the situation, the IETF set up a group to devise a data link protocol for point-to-point lines that solved all these problems and that could become an official Internet Standard. This work culminated in **PPP (Point-to-Point Protocol)**, which is defined in RFC 1661 and further elaborated on in several other RFCs (e.g., RFCs 1662 and 1663). PPP handles error detection, supports multiple protocols, allows IP addresses to be negotiated at connection time, permits authentication, and has many other improvements over SLIP. While many Internet service providers still support both SLIP and PPP, the future clearly lies with PPP, not only for dial-up lines, but also for leased router-router lines.

PPP provides three things:

1. A framing method that unambiguously delineates the end of one frame and the start of the next one. The frame format also handles error detection.

2. A link control protocol for bringing lines up, testing them, negotiating options, and bringing them down again gracefully when they are no longer needed. This protocol is called **LCP (Link Control Protocol)**.

3. A way to negotiate network-layer options in a way that is independent of the network layer protocol to be used. The method chosen is to have a different **NCP (Network Control Protocol)** for each network layer supported.

To see how these pieces fit together, let us consider the typical scenario of a home user calling up an Internet service provider to make a home PC a temporary Internet host. The PC first calls the provider's router via a modem. After the router's modem has answered the phone and established a physical connection, the PC sends the router a series of LCP packets in the payload field of one or more PPP frames. These packets, and their responses, select the PPP parameters to be used.

Once these have been agreed upon, a series of NCP packets are sent to configure the network layer. Typically, the PC wants to run a TCP/IP protocol stack, so it needs an IP address. There are not enough IP addresses to go around, so normally each Internet provider gets a block of them and then dynamically assigns one to each newly attached PC for the duration of its login session. If a provider owns n IP addresses, it can have up to n machines logged in simultaneously, but its total customer base may be many times that. The NCP for IP is used to do the IP address assignment.

At this point, the PC is now an Internet host and can send and receive IP packets, just as hardwired hosts can. When the user is finished, NCP is used to tear down the network layer connection and free up the IP address. Then LCP is used to shut down the data link layer connection. Finally, the computer tells the modem to hang up the phone, releasing the physical layer connection.

The PPP frame format was chosen to closely resemble the HDLC frame format, since there was no reason to reinvent the wheel. The major difference between PPP and HDLC is that the former is character oriented rather than bit oriented. In particular, PPP, like, SLIP, uses character stuffing on dial-up modem lines, so all frames are an integral number of bytes. It is not possible to send a frame consisting of 30.25 bytes, as it is with HDLC. Not only can PPP frames be sent over dial-up telephone lines, but they can also be sent over SONET or true bit-oriented HDLC lines (e.g., for router-router connections). The PPP frame format is shown in Fig. 3-27.

Bytes	1	1	1	1 or 2	Variable	2 or 4	1
	Flag 01111110	Address 11111111	Control 00000011	Protocol	Payload	Checksum	Flag 01111110

Fig. 3-27. The PPP full frame format for unnumbered mode operation.

All PPP frames begin with the standard HDLC flag byte (01111110), which is character stuffed if it occurs within the payload field. Next comes the *Address* field, which is always set to the binary value 11111111 to indicate that all stations are to accept the frame. Using this value avoids the issue of having to assign data link addresses.

The *Address* field is followed by the *Control* field, the default value of which is 00000011. This value indicates an unnumbered frame. In other words, PPP does not provide reliable transmission using sequence numbers and acknowledgements as the default. In noisy environments, such as wireless networks, reliable transmission using numbered mode can be used. The exact details are defined in RFC 1663.

Since the *Address* and *Control* fields are always constant in the default configuration, LCP provides the necessary mechanism for the two parties to negotiate an option to just omit them altogether and save 2 bytes per frame.

The fourth PPP field is the *Protocol* field. Its job is to tell what kind of packet is in the *Payload* field. Codes are defined for LCP, NCP, IP, IPX, AppleTalk, and other protocols. Protocols starting with a 0 bit are network layer protocols such as IP, IPX, OSI CLNP, XNS. Those starting with a 1 bit are used to negotiate other protocols. These include LCP and a different NCP for each network layer protocol supported. The default size of the *Protocol* field is 2 bytes, but it can be negotiated down to 1 byte using LCP.

The *Payload* field is variable length, up to some negotiated maximum. If the length is not negotiated using LCP during line setup, a default length of 1500 bytes is used. Padding may follow the payload if need be.

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After the *Payload* field comes the *Checksum* field, which is normally 2 bytes, but a 4-byte checksum can be negotiated.

In summary, PPP is a multiprotocol framing mechanism suitable for use over modems, HDLC bit-serial lines, SONET, and other physical layers. It supports error detection, option negotiation, header compression, and optionally, reliable transmission using HDLC framing.

Let us now turn from the PPP frame format to the way lines are brought up and down. The (simplified) diagram of Fig. 3-28 shows the phases that a line goes through when it is brought up, used, and taken down again. This sequence applies both to modem connections and to router-router connections.

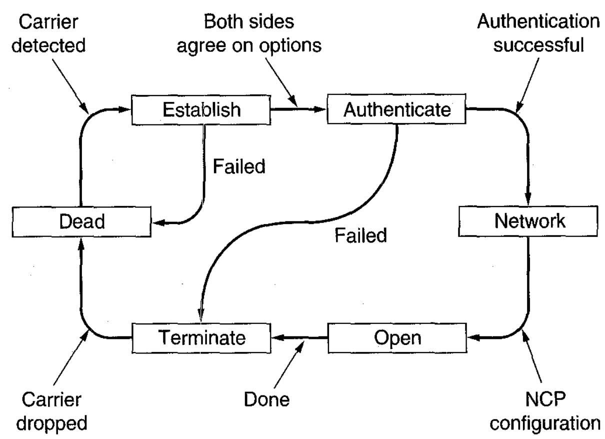

Fig. 3-28. A simplified phase diagram for bringing a line up and down.

When the line is *DEAD*, no physical layer carrier is present and no physical layer connection exists. After physical connection is established, the line moves to *ESTABLISHED*. At that point LCP option negotiation begins, which, if successful, leads to *AUTHENTICATE*. Now the two parties can check on each other's identities, if desired. When the *NETWORK* phase is entered, the appropriate NCP protocol is invoked to configure the network layer. If the configuration is successful, *OPEN* is reached and data transport can take place. When data transport is finished, the line moves into the *TERMINATE* phase, and from there, back to *DEAD* when the carrier is dropped.

LCP is used to negotiate data link protocol options during the *ESTABLISH* phase. The LCP protocol is not actually concerned with the options themselves, but with the mechanism for negotiation. It provides a way for the initiating process to make a proposal and for the responding process to accept or reject it, in whole or in part. It also provides a way for the two processes to test the line

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quality, to see if they consider it good enough to set up a connection. Finally, the LCP protocol also allows lines to be taken down when they are no longer needed.

Eleven types of LCP packets are defined in RFC 1661. These are listed in Fig. 3-29. The four *Configure-* types allow the initiator (I) to propose option values and the responder (R) to accept or reject them. In the latter case, the responder can make an alternative proposal or announce that it is not willing to negotiate certain options at all. The options being negotiated and their proposed values are part of the LCP packets.

Name	Direction	Description
Configure-request	I → R	List of proposed options and values
Configure-ack	I ← R	All options are accepted
Configure-nak	I ← R	Some options are not accepted
Configure-reject	I ← R	Some options are not negotiable
Terminate-request	I → R	Request to shut the line down
Terminate-ack	I ← R	OK, line shut down
Code-reject	I ← R	Unknown request received
Protocol-reject	I ← R	Unknown protocol requested
Echo-request	I → R	Please send this frame back
Echo-reply	I ← R	Here is the frame back
Discard-request	I → R	Just discard this frame (for testing)

Fig. 3-29. The LCP packet types.

The *Terminate-* codes are used to shut a line down when it is no longer needed. The *Code-reject* and *Protocol-reject* codes are used by the responder to indicate that it got something that it does not understand. This situation could mean that an undetected transmission error has occurred, but more likely it means that the initiator and responder are running different versions of the LCP protocol. The *Echo-* types are used to test the line quality. Finally, *Discard-request* is used for debugging. If either end is having trouble getting bits onto the wire, the programmer can use this type for testing. If it manages to get through, the receiver just throws it away, rather than taking some other action, which might confuse the person doing the testing.

The options that can be negotiated include setting the maximum payload size for data frames, enabling authentication and choosing a protocol to use, enabling line quality monitoring during normal operation, and selecting various header compression options.

There is little to say about the NCP protocols in a general way. Each one is specific to some network layer protocol and allows configuration requests to be made that are specific to that protocol. For IP, for example, dynamic address assignment is the most important possibility.

3.6.3. The Data Link Layer in ATM

It is now time to begin our journey up through the ATM protocol layers of Fig. 1-30. The ATM physical layer covers roughly the OSI physical and data link layers, with the physical medium dependent sublayer being functionally like the OSI physical layer and the transmission convergence (TC) sublayer having data link functionality. There are no physical layer characteristics specific to ATM. Instead, ATM cells are carried by SONET, FDDI, and other transmission systems. Therefore we will concentrate here on the data link functionality of the TC sublayer, but we will discuss some aspects of the interface with the lower sublayer later on.

When an application program produces a message to be sent, that message works its way down the ATM protocol stack, having headers and trailers added and undergoing segmentation into cells. Eventually, the cells reach the TC sublayer for transmission. Let us see what happens to them on the way out the door.

Cell Transmission

The first step is header checksumming. Each cell contains a 5-byte header consisting of 4 bytes of virtual circuit and control information followed by a 1-byte checksum. Although the contents of the header are not relevant to the TC sublayer, curious readers wishing a sneak preview should turn to Fig. 5-62. The checksum only covers the first four header bytes, not the payload field. It consists of the remainder after the 32 header bits have been divided by the polynomial $x^8 + x^2 + x + 1$. To this the constant 01010101 is added, to provide robustness in the face of headers containing mostly 0 bits.

The decision to checksum only the header was made to reduce the probability of cells being delivered incorrectly due to a header error, but to avoid paying the price of checksumming the much larger payload field. It is up to higher layers to perform this function, if they so desire. For many real-time applications, such as voice and video, losing a few bits once in a while is acceptable (although for some compression schemes, all frames are equal but some frames are more equal). Because it covers only the header, the 8-bit checksum field is called the **HEC** (**Header Error Control**).

A factor that played a major role in this checksumming scheme is the fact that ATM was designed for use over fiber, and fiber is highly reliable. Furthermore, a major study of the U.S. telephone network has shown that during normal operation 99.64 percent of all errors on fiber optic lines are single-bit errors (AT&T and Bellcore, 1989). The HEC scheme corrects all single-bit errors and detects many multibit errors as well. If we assume that the probability of a single-bit error is 10^{-8}, then the probability of a cell containing a detectable multibit header error is about 10^{-13}. The probability of a cell slipping through with an undetected header error is about 10^{-20}, which means that at OC-3 speed, one bad cell header will get through every 90,000 years. Although this may sound like a long time, once the earth has, say, 1 billion ATM telephones, each used 10 percent of the time, over 1000 bad cell headers per year will go undetected.

For applications that need reliable transmission in the data link layer, Shacham and McKenney (1990) have developed a scheme in which a sequence of consecutive cells are EXCLUSIVE ORed together. The result, an entire cell, is appended to the sequence. If one cell is lost or badly garbled, it can be reconstructed from the available information.

Once the HEC has been generated and inserted into the cell header, the cell is ready for transmission. Transmission media come in two categories: asynchronous and synchronous. When an asynchronous medium is used, a cell can be sent whenever it is ready to go. No timing restrictions exist.

With a synchronous medium, cells must be transmitted according to a predefined timing pattern. If no data cell is available when needed, the TC sublayer must invent one. These are called **idle cells**.

Another kind of nondata cell is the **OAM** (**Operation And Maintenance**) cell. OAM cells are also used by the ATM switches for exchanging control and other information necessary for keeping the system running. OAM cells also have some other special functions. For example, the 155.52-Mbps OC-3 speed matches the gross data rate of SONET, but an STM-1 frame has a total of 10 columns of overhead out of 270, so the SONET payload is only $260/270 \times 155.52$ Mbps or 149.76 Mbps. To keep from swamping SONET, an ATM source using SONET would normally put out an OAM cell as every 27th cell, to slow the data rate down to 26/27 of 155.52 Mbps and thus match SONET exactly. The job of matching the ATM output rate to the rate of the underlying transmission system is an important task of the TC sublayer.

On the receiver's side, idle cells are processed in the TC sublayer, but OAM cells are given to the ATM layer. OAM cells are distinguished from data cells by having the first three header bytes be all zeros, something not allowed for data cells. The fourth byte describes the nature of the OAM cell.

Another important task of the TC sublayer is generating the framing information for the underlying transmission system, if any. For example, an ATM video camera might just produce a sequence of cells on the wire, but it might also produce SONET frames with the ATM cells embedded inside the SONET payload. In the latter case, the TC sublayer would generate the SONET framing and pack the ATM cells inside, not entirely a trivial business since a SONET payload does not hold an integral number of 53-byte cells.

Although the telephone companies clearly intend to use SONET as the underlying transmission system for ATM, mappings from ATM onto the payload fields of other systems have also been defined, and new ones are being worked on. In particular, mappings onto T1, T3, and FDDI also exist.

Cell Reception

On output, the job of the TC sublayer is to take a sequence of cells, add a HEC to each one, convert the result to a bit stream, and match the bit stream to the speed of the underlying physical transmission system by inserting OAM cells as filler. On input, the TC sublayer does exactly the reverse. It takes an incoming bit stream, locates the cell boundaries, verifies the headers (discarding cells with invalid headers), processes the OAM cells, and passes the data cells up to the ATM layer.

The hardest part is locating the cell boundaries in the incoming bit stream. At the bit level, a cell is just a sequence of $53 \times 8 = 424$ bits. No 01111110 flag bytes are present to mark the start and end of a cell, as they are in HDLC. In fact, there are no markers at all. How can cell boundaries be recognized under these circumstances?

In some case, the underlying physical layer provides help. With SONET, for example, cells can be aligned with the synchronous payload envelope, so the SPE pointer in the SONET header points to the start of the first full cell. However, sometimes the physical layer provides no assistance in framing. What then?

The trick is to use the HEC. As the bits come in, the TC sublayer maintains a 40-bit shift register, with bits entering on the left and exiting on the right. The TC sublayer then inspects the 40 bits to see if it is potentially a valid cell header. If it is, the rightmost 8 bits will be valid HEC over the leftmost 32 bits. If this condition does not hold, the buffer does not hold a valid cell, in which case all the bits in the buffer are shifted right one bit, causing one bit to fall off the end, and a new input bit is inserted at the left end. This process is repeated until a valid HEC is located. At that point, the cell boundary is known because the shift register contains a valid header.

The trouble with this heuristic is that the HEC is only 8 bits wide. For any given shift register, even one containing random bits, the probability of finding a valid HEC is 1/256, a moderately large value. Used by itself, this procedure would incorrectly detect cell headers far too often.

To improve the accuracy of the recognition algorithm, the finite state machine of Fig. 3-30 is used. Three states are used: *HUNT*, *PRESYNCH*, and *SYNCH*. In the *HUNT* state, the TC sublayer is shifting bits into the shift registers one at a time looking for a valid HEC. As soon as one is found, the finite state machine switches to *PRESYNCH* state, meaning that it has tentatively located a cell boundary. It now shifts in the next 424 bits (53 bytes) without examining them. If its guess about the cell boundary was correct, the shift register should now contain another valid cell header, so it once again runs the HEC algorithm. If the HEC is incorrect, the TC goes back to the *HUNT* state and continues to search bit-by-bit for a header whose HEC is correct.

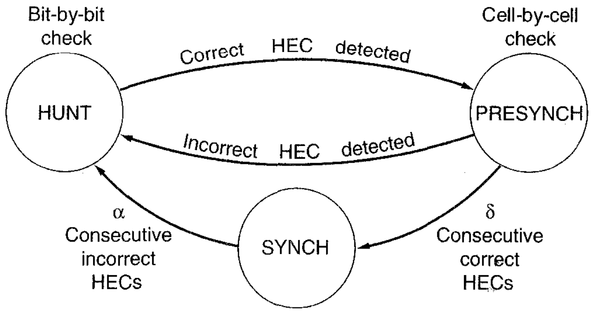

Fig. 3-30. The cell delineation heuristic.

On the other hand, if the second HEC is also correct, the TC may be onto something, so it shifts in another 424 bits and tries again. It continues inspecting headers in this fashion until it has found δ correct headers in a row, at which time it assumes that it is synchronized and moves into the *SYNCH* state to start normal operation. Note that the probability of getting into *SYNCH* state by accident with a purely random bit stream is $2^{-8\delta}$, which can be made arbitrarily small by choosing a large enough δ. The price paid for a large δ, however, is a longer time to synchronize.

In addition to resynchronizing after losing synchronization (or at startup), the TC sublayer needs a heuristic to determine when it has lost synchronization, for example after a bit has been inserted or deleted from the bit stream. It would be unwise to give up if just one HEC was incorrect, since most errors are bit inversions, not insertions or deletions. The wisest course here is just to discard the cell with the bad header and hope the next one is good. However, if α HECs in a row are bad, the TC sublayer has to conclude that it has lost synchronization and must return to the *HUNT* state.

Although unlikely, it is conceivable that a malicious user could try to spoof the TC sublayer by inserting a data pattern into the payload field of many consecutive cells that imitates the HEC algorithm. Then, if synchronization were ever lost, it might be regained in the wrong place. To make this trick much harder, the payload bits are scrambled on transmission and descrambled on reception.

Before leaving the TC sublayer, one comment is in order. The mechanism chosen for cell delineation requires the TC sublayer to understand and use the header of the ATM layer above it. Having one layer make use of the header of a higher layer is in complete violation of the basic rules of protocol engineering. The idea of having layered protocols is to make each layer be independent of the ones above it. It should be possible, for example, to change the header format of the ATM layer without affecting the TC sublayer. However due to the way cell delineation is accomplished, making such a change is not possible.

3.7. SUMMARY

The task of the data link layer is to convert the raw bit stream offered by the physical layer into a stream of frames for use by the network layer. Various framing methods are used, including character count, character stuffing, and bit stuffing. Data link protocols can provide error control to retransmit damaged or lost frames. To prevent a fast sender from overrunning a slow receiver, the data link protocol can also provide flow control. The sliding window mechanism is widely used to integrate error control and flow control in a convenient way.

Sliding window protocols can be categorized by the size of the sender's window and the size of the receiver's window. When both are equal to 1, the protocol is stop-and-wait. When the sender's window is greater than 1, for example to prevent the sender from blocking on a circuit with a long propagation delay, the receiver can be programmed either to discard all frames other than the next one in sequence (protocol 5) or buffer out of order frames until they are needed (protocol 6).

Protocols can be modeled using various techniques to help demonstrate their correctness (or lack thereof). Finite state machine models and Petri net models are commonly used for this purpose.

Many networks use one of the bit-oriented protocols—SDLC, HDLC, ADCCP, or LAPB—at the data link level. All of these protocols use flag bytes to delimit frames, and bit stuffing to prevent flag bytes from occurring in the data. All of them also use a sliding window for flow control. The Internet uses SLIP and PPP as data link protocols. ATM systems have their own simple protocol, which does a bare minimum of error checking and no flow control.

PROBLEMS

1. An upper layer message is split into 10 frames, each of which has an 80 percent chance of arriving undamaged. If no error control is done by the data link protocol, how many times must the message be sent on the average to get the entire thing through?

2. The following data fragment occurs in the middle of a data stream for which the character-stuffing algorithm described in the text is used: DLE, STX, A, DLE, B, DLE, ETX. What is the output after stuffing?

3. If the bit string 0111101111101111110 is bit stuffed, what is the output string?

4. When bit stuffing is used, is it possible for the loss, insertion, or modification of a single bit to cause an error not detected by the checksum? If not, why not? If so, how? Does the checksum length play a role here?

5. Can you think of any circumstances under which an open-loop protocol, (e.g., a Hamming code) might be preferable to the feedback type protocols discussed throughout this chapter?

6. To provide more reliability than a single parity bit can give, an error-detecting coding scheme uses one parity bit for checking all the odd numbered bits and a second parity bit for all the even numbered bits. What is the Hamming distance of this code?

7. One way of detecting errors is to transmit data as a block of n rows of k bits per row and adding parity bits to each row and each column. Will this scheme detect all single errors? Double errors? Triple errors?

8. A block of bits with n rows and k columns uses horizontal and vertical parity bits for error detection. Suppose that exactly 4 bits are inverted due to transmission errors. Derive an expression for the probability that the error will be undetected.

9. What is the remainder obtained by dividing $x^7 + x^5 + 1$ by the generator polynomial $x^3 + 1$?

10. Data link protocols almost always put the CRC in a trailer, rather than in a header. Why?

11. A channel has a bit rate of 4 kbps and a propagation delay of 20 msec. For what range of frame sizes does stop-and-wait give an efficiency of at least 50 percent?

12. A 3000-km long T1 trunk is used to transmit 64-byte frames using protocol 5. If the propagation speed is 6 μsec/km, how many bits should the sequence numbers be?

13. Imagine a sliding window protocol using so many bits for sequence numbers that wraparound never occurs. What relations must hold among the four window edges and the window size?

14. If the procedure *between* in protocol 5 checked for the condition $a \le b \le c$ instead of the condition $a \le b < c$, would that have any effect on the protocol's correctness or efficiency? Explain your answer.

15. In protocol 6, when a data frame arrives, a check is made to see if the sequence number differs from the one expected and *NoNak* is true. If both conditions hold, a NAK is sent. Otherwise, the auxiliary timer is started. Suppose that the else clause were omitted. Would this change affect the protocol's correctness?

16. Suppose that the three-statement `while` loop near the end of protocol 6 were removed from the code. Would this affect the correctness of the protocol or just the performance? Explain your answer.

17. Suppose that the case for checksum errors were removed from the `switch` statement of protocol 6. How would this change affect the operation of the protocol?

18. In protocol 6 the code for *FrameArrival* has a section used for NAKs. This section is invoked if the incoming frame is a NAK and another condition is met. Give a scenario where the presence of this other condition is essential.

19. Imagine that you are writing the data link layer software for a line used to send data to you, but not from you. The other end uses HDLC, with a 3-bit sequence number and a window size of seven frames. You would like to buffer as many out of sequence frames as possible to enhance efficiency, but you are not allowed to modify the software on the sending side. Is it possible to have a receiver window greater than one, and still guarantee that the protocol will never fail? If so, what is the largest window that can be safely used?

20. Consider the operation of protocol 6 over a 1-Mbps error-free line. The maximum frame size is 1000 bits. New packets are generated about 1 second apart. The timeout interval is 10 msec. If the special acknowledgement timer were eliminated, unnecessary timeouts would occur. How many times would the average message be transmitted?

21. In protocol 6 $MaxSeq = 2^n - 1$. While this condition is obviously desirable to make efficient use of header bits, we have not demonstrated that it is essential. Does the protocol work correctly for $MaxSeq = 4$, for example?

22. Frames of 1000 bits are sent over a 1-Mbps satellite channel. Acknowledgements are always piggybacked onto data frames. The headers are very short. Three-bit sequence numbers are used. What is the maximum achievable channel utilization for
(a) Stop-and-wait.
(b) Protocol 5.
(c) Protocol 6.

23. Compute the fraction of the bandwidth that is wasted on overhead (headers and retransmissions) for protocol 6 on a heavily loaded 50-kbps satellite channel with data frames consisting of 40 header and 3960 data bits. ACK frames never occur. NAK frames are 40 bits. The error rate for data frames is 1 percent, and the error rate for NAK frames is negligible. The sequence numbers are 8 bits.

24. Consider an error-free 64-kbps satellite channel used to send 512-byte data frames in one direction, with very short acknowledgements coming back the other way. What is the maximum throughput for window sizes of 1, 7, 15, and 127?

25. A 100 km long cable runs at the T1 data rate. The propagation speed in the cable is 2/3 the speed of light. How many bits fit in the cable?

26. Redraw Fig. 3-21 for a full-duplex channel that never loses frames. Is the protocol failure still possible?

27. Give the firing sequence for the Petri net of Fig. 3-23 corresponding to the state sequence (000), (01A), (01—), (010), (01A) in Fig. 3-20. Explain in words what the sequence represents.

28. Given the transition rules $AC \rightarrow B$, $B \rightarrow AC$, $CD \rightarrow E$, and $E \rightarrow CD$, draw the Petri net described. From the Petri net, draw the finite state graph reachable from the initial state ACD. What well-known computer science concept do these transition rules model?

29. PPP is based closely on HDLC, which uses bit stuffing to prevent accidental flag bytes within the payload from causing confusion. Give at least one reason why PPP uses character stuffing instead.

30. What is the minimum overhead in sending an IP packet using PPP? Count only the overhead introduced by PPP itself, not the IP header overhead.

31. Consider the ATM cell delineation heuristic with $\alpha = 5$, $\delta = 6$, and a per-bit error rate of 10^{-5}. Once the system is synchronized, how long will it remain so, despite occasional header bit errors? Assume the line is running at OC-3.

32. Write a program to stochastically simulate the behavior of a Petri net. The program should read in the transition rules as well as a list of states corresponding to the network link layer issuing a new packet or the accepting a new packet. From the initial state, also read in, the program should pick enabled transitions at random and fire them, checking to see if a host ever accepts two messages without the other host emitting a new one in between.

4

THE MEDIUM ACCESS SUBLAYER

As we pointed out in Chap. 1, networks can be divided into two categories: those using point-to-point connections and those using broadcast channels. This chapter deals with broadcast networks and their protocols.

In any broadcast network, the key issue is how to determine who gets to use the channel when there is competition for it. To make this point clearer, consider a conference call in which six people, on six different telephones, are all connected together so that each one can hear and talk to all the others. It is very likely that when one of them stops speaking, two or more will start talking at once, leading to chaos. In a face-to-face meeting, chaos is avoided by external means, for example, at a meeting, people raise their hands to request permission to speak. When only a single channel is available, determining who should go next is much harder. Many protocols for solving the problem are known and form the contents of this chapter. In the literature, broadcast channels are sometimes referred to as **multiaccess channels** or **random access channels**.

The protocols used to determine who goes next on a multiaccess channel belong to a sublayer of the data link layer called the **MAC** (**Medium Access Control**) sublayer. The MAC sublayer is especially important in LANs, nearly all of which use a multiaccess channel as the basis of their communication. WANs, in contrast, use point-to-point links, except for satellite networks. Because multiaccess channels and LANs are so closely related, in this chapter we will discuss LANs in general, as well as satellite and some other broadcast networks.

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Technically, the MAC sublayer is the bottom part of the data link layer, so logically we should have studied it before examining all the point-to-point protocols in Chap. 3. Nevertheless, for most people, understanding protocols involving multiple parties is easier after two-party protocols are well understood. For that reason we have deviated slightly from a strict bottom-up order of presentation.

4.1. THE CHANNEL ALLOCATION PROBLEM

The central theme of this chapter is how to allocate a single broadcast channel among competing users. We will first look at static and dynamic schemes in general. Then we will examine a number of specific algorithms.

4.1.1. Static Channel Allocation in LANs and MANs

The traditional way of allocating a single channel, such as a telephone trunk, among multiple competing users is Frequency Division Multiplexing (FDM). If there are N users, the bandwidth is divided into N equal sized portions (see Fig. 2-24), each user being assigned one portion. Since each user has a private frequency band, there is no interference between users. When there is only a small and fixed number of users, each of which has a heavy (buffered) load of traffic (e.g., carriers' switching offices), FDM is a simple and efficient allocation mechanism.

However, when the number of senders is large and continuously varying, or the traffic is bursty, FDM presents some problems. If the spectrum is cut up into N regions, and fewer than N users are currently interested in communicating, a large piece of valuable spectrum will be wasted. If more than N users want to communicate, some of them will be denied permission, for lack of bandwidth, even if some of the users who have been assigned a frequency band hardly ever transmit or receive anything.

However, even assuming that the number of users could somehow be held constant at N, dividing the single available channel into static subchannels is inherently inefficient. The basic problem is that when some users are quiescent, their bandwidth is simply lost. They are not using it, and no one else is allowed to use it either. Furthermore, in most computer systems, data traffic is extremely bursty (peak traffic to mean traffic ratios of 1000:1 are common). Consequently, most of the channels will be idle most of the time.

The poor performance of static FDM can easily be seen from a simple queueing theory calculation. Let us start with the mean time delay, T, for a channel of capacity C bps, with an arrival rate of λ frames/sec, each frame having a length drawn from an exponential probability density function with mean $1/\mu$ bits/frame:

$$T = \frac{1}{\mu C - \lambda}$$

Now let us divide the single channel up into N independent subchannels, each with capacity C/N bps. The mean input rate on each of the subchannels will now be λ/N. Recomputing T we get

$$T_{\mathrm{FDM}} = \frac{1}{\mu(C/N) - (\lambda/N)} = \frac{N}{\mu C - \lambda} = NT \tag{4-1}$$

The mean delay using FDM is N times worse than if all the frames were somehow magically arranged orderly in a big central queue.

Precisely the same arguments that apply to FDM also apply to time division multiplexing (TDM). Each user is statically allocated every Nth time slot. If a user does not use the allocated slot, it just lies fallow. Since none of the traditional static channel allocation methods work well with bursty traffic, we will now explore dynamic methods.

4.1.2. Dynamic Channel Allocation in LANs and MANs

Before we get into the first of the many channel allocation methods to be discussed in this chapter, it is worthwhile carefully formulating the allocation problem. Underlying all the work done in this area are five key assumptions, described below.

1. **Station model**. The model consists of N independent **stations** (computers, telephones, personal communicators, etc.), each with a program or user that generates frames for transmission. The probability of a frame being generated in an interval of length Δt is $\lambda \Delta t$, where λ is a constant (the arrival rate of new frames). Once a frame has been generated, the station is blocked and does nothing until the frame has been successfully transmitted.

2. **Single Channel Assumption**. A single channel is available for all communication. All stations can transmit on it and all can receive from it. As far as the hardware is concerned, all stations are equivalent, although protocol software may assign priorities to them.

3. **Collision Assumption**. If two frames are transmitted simultaneously, they overlap in time and the resulting signal is garbled. This event is called a **collision**. All stations can detect collisions. A collided frame must be transmitted again later. There are no errors other than those generated by collisions.

4a. **Continuous Time**. Frame transmission can begin at any instant. There is no master clock dividing time into discrete intervals.

4b. **Slotted Time**. Time is divided into discrete intervals (slots). Frame transmissions always begin at the start of a slot. A slot may contain 0, 1, or more frames, corresponding to an idle slot, a successful transmission, or a collision, respectively.

5a. **Carrier Sense**. Stations can tell if the channel is in use before trying to use it. If the channel is sensed as busy, no station will attempt to use it until it goes idle.

5b. **No Carrier Sense**. Stations cannot sense the channel before trying to use it. They just go ahead and transmit. Only later can they determine whether or not the transmission was successful.

Some discussion of these assumptions is in order. The first one says that stations are independent, and that work is generated at a constant rate. It also implicitly assumes that each station only has one program or user, so while the station is blocked, no new work is generated. More sophisticated models allow multiprogrammed stations that can generate work while a station is blocked, but the analysis of these stations is much more complex.

The single channel assumption is the heart of the matter. There are no external ways to communicate. Stations cannot raise their hands to request that the teacher call on them.

The collision assumption is also basic, although in some systems (notably spread spectrum), this assumption is relaxed, with surprising results. Also, some LANs, such as token rings, use a mechanism for contention elimination that eliminates collisions.

There are two alternative assumptions about time. Either it is continuous or it is slotted. Some systems use one and some systems use the other, so we will discuss and analyze both. Obviously, for a given system, only one of them holds.

Similarly, a network can either have carrier sensing or not have it. LANs generally have carrier sense, but satellite networks do not (due to the long propagation delay). Stations on carrier sense networks can terminate their transmission prematurely if they discover that it is colliding with another transmission. Note that the word "carrier" in this sense refers to an electrical signal on the cable and has nothing to do with the common carriers (e.g., telephone companies) that date back to the Pony Express days.

4.2. MULTIPLE ACCESS PROTOCOLS

Many algorithms for allocating a multiple access channel are known. In the following sections we will study a representative sample of the more interesting ones and give some examples of their use.

4.2.1. ALOHA

In the 1970s, Norman Abramson and his colleagues at the University of Hawaii devised a new and elegant method to solve the channel allocation problem. Their work has been extended by many researchers since then (Abramson, 1985). Although Abramson's work, called the ALOHA system, used ground-based radio broadcasting, the basic idea is applicable to any system in which uncoordinated users are competing for the use of a single shared channel.

We will discuss two versions of ALOHA here: pure and slotted. They differ with respect to whether or not time is divided up into discrete slots into which all frames must fit. Pure ALOHA does not require global time synchronization; slotted ALOHA does.

Pure ALOHA

The basic idea of an ALOHA system is simple: let users transmit whenever they have data to be sent. There will be collisions, of course, and the colliding frames will be destroyed. However, due to the feedback property of broadcasting, a sender can always find out whether or not its frame was destroyed by listening to the channel, the same way other users do. With a LAN, the feedback is immediate; with a satellite, there is a delay of 270 msec before the sender knows if the transmission was successful. If the frame was destroyed, the sender just waits a random amount of time and sends it again. The waiting time must be random or the same frames will collide over and over, in lockstep. Systems in which multiple users share a common channel in a way that can lead to conflicts are widely known as **contention** systems.

A sketch of frame generation in an ALOHA system is given in Fig. 4-1. We have made the frames all the same length because the throughput of ALOHA systems is maximized by having a uniform frame size rather than allowing variable length frames.

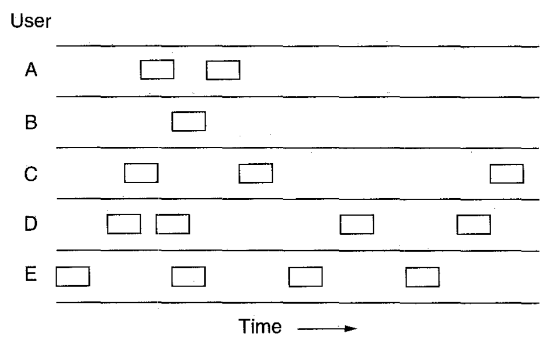

Fig. 4-1. In pure ALOHA, frames are transmitted at completely arbitrary times.

Whenever two frames try to occupy the channel at the same time, there will be a collision and both will be garbled. If the first bit of a new frame overlaps with just the last bit of a frame almost finished, both frames will be totally

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destroyed, and both will have to be retransmitted later. The checksum cannot (and should not) distinguish between a total loss and a near miss. Bad is bad.

A most interesting question is: What is the efficiency of an ALOHA channel? That is, what fraction of all transmitted frames escape collisions under these chaotic circumstances? Let us first consider an infinite collection of interactive users sitting at their computers (stations). A user is always in one of two states: typing or waiting. Initially, all users are in the typing state. When a line is finished, the user stops typing, waiting for a response. The station then transmits a frame containing the line and checks the channel to see if it was successful. If so, the user sees the reply and goes back to typing. If not, the user continues to wait and the frame is retransmitted over and over until it has been successfully sent.

Let the "frame time" denote the amount of time needed to transmit the standard, fixed-length frame (i.e., the frame length divided by the bit rate). At this point we assume that the infinite population of users generates new frames according to a Poisson distribution with mean S frames per frame time. (The infinite-population assumption is needed to ensure that S does not decrease as users become blocked.) If $S > 1$, the user community is generating frames at a higher rate than the channel can handle, and nearly every frame will suffer a collision. For reasonable throughput we would expect $0 < S < 1$.

In addition to the new frames, the stations also generate retransmissions of frames that previously suffered collisions. Let us further assume that the probability of k transmission attempts per frame time, old and new combined, is also Poisson, with mean G per frame time. Clearly, $G \geq S$. At low load (i.e., $S \approx 0$), there will be few collisions, hence few retransmissions, so $G \approx S$. At high load there will be many collisions, so $G > S$. Under all loads, the throughput is just the offered load, G, times the probability of a transmission being successful—that is, $S = GP_0$, where P_0 is the probability that a frame does not suffer a collision.

A frame will not suffer a collision if no other frames are sent within one frame time of its start, as shown in Fig. 4-2. Under what conditions will the shaded frame arrive undamaged? Let t be the time required to send a frame. If any other user has generated a frame between time t_0 and $t_0 + t$, the end of that frame will collide with the beginning of the shaded one. In fact, the shaded frame's fate was already sealed even before the first bit was sent, but since in pure ALOHA a station does not listen to the channel before transmitting, it has no way of knowing that another frame was already underway. Similarly, any other frame started between $t_0 + t$ and $t_0 + 2t$ will bump into the end of the shaded frame.

The probability that k frames are generated during a given frame time is given by the Poisson distribution:

$$\Pr[k] = \frac{G^k e^{-G}}{k!} \qquad (4\text{-}2)$$

so the probability of zero frames is just e^{-G}. In an interval two frame times long, the mean number of frames generated is $2G$. The probability of no other traffic

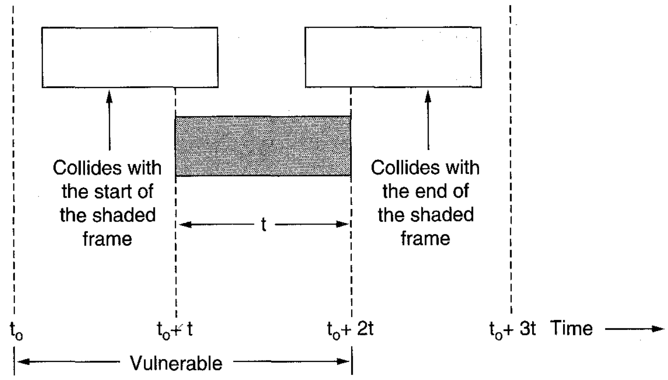

Fig. 4-2. Vulnerable period for the shaded frame.

being initiated during the entire vulnerable period is thus given by $P_0 = e^{-2G}$. Using $S = GP_0$, we get

$$S = Ge^{-2G}$$

The relation between the offered traffic and the throughput is shown in Fig. 4-3. The maximum throughput occurs at $G = 0.5$, with $S = 1/2e$, which is about 0.184. In other words, the best we can hope for is a channel utilization of 18 percent. This result is not very encouraging, but with everyone transmitting at will, we could hardly have expected a 100 percent success rate.

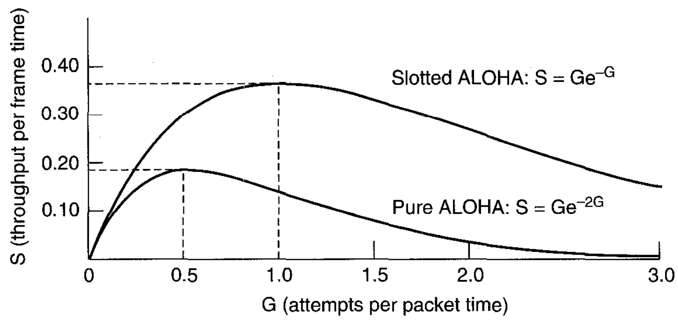

Fig. 4-3. Throughput versus offered traffic for ALOHA systems.

Slotted ALOHA

In 1972, Roberts published a method for doubling the capacity of an ALOHA system (Roberts, 1972). His proposal was to divide time up into discrete intervals, each interval corresponding to one frame. This approach requires the users to agree of slot boundaries. One way to achieve synchronization would be to have one special station emit a pip at the start of each interval, like a clock.

In Roberts' method, which has come to be known as **slotted ALOHA**, in contrast to Abramson's **pure ALOHA**, a computer is not permitted to send whenever a carriage return is typed. Instead, it is required to wait for the beginning of the next slot. Thus the continuous pure ALOHA is turned into a discrete one. Since the vulnerable period is now halved, the probability of no other traffic during the same slot as our test frame is e^{-G} which leads to

$$S = Ge^{-G} \tag{4-3}$$

As you can see from Fig. 4-3, slotted ALOHA peaks at $G = 1$, with a throughput of $S = 1/e$ or about 0.368, twice that of pure ALOHA. If the system is operating at $G = 1$, the probability of an empty slot is 0.368 (from Eq. 4-2). The best we can hope for using slotted ALOHA is 37 percent of the slots empty, 37 percent successes, and 26 percent collisions. Operating at higher values of G reduces the number of empties but increases the number of collisions exponentially. To see how this rapid growth of collisions with G comes about, consider the transmission of a test frame. The probability that it will avoid a collision is e^{-G}, the probability that all the other users are silent in that slot. The probability of a collision is then just $1 - e^{-G}$. The probability of a transmission requiring exactly k attempts, (i.e., $k - 1$ collisions followed by one success) is

$$P_k = e^{-G}(1 - e^{-G})^{k-1}$$

The expected number of transmissions, E, per carriage return typed is then

$$E = \sum_{k=1}^{\infty} kP_k = \sum_{k=1}^{\infty} ke^{-G}(1 - e^{-G})^{k-1} = e^G$$

As a result of the exponential dependence of E upon G, small increases in the channel load can drastically reduce its performance.

4.2.2. Carrier Sense Multiple Access Protocols

With slotted ALOHA the best channel utilization that can be achieved is $1/e$. This is hardly surprising, since with stations transmitting at will, without paying attention to what the other stations are doing, there are bound to be many collisions. In local area networks, however it is possible for stations to detect what other stations are doing, and adapt their behavior accordingly. These networks can achieve a much better utilization than $1/e$. In this section we will discuss some protocols for improving performance.

Protocols in which stations listen for a carrier (i.e., a transmission) and act accordingly are called **carrier sense protocols**. A number of them have been proposed. Kleinrock and Tobagi (1975) have analyzed several such protocols in detail. Below we will mention several versions of the carrier sense protocols.

Persistent and Nonpersistent CSMA

The first carrier sense protocol that we will study here is called **1-persistent CSMA** (Carrier Sense Multiple Access). When a station has data to send, it first listens to the channel to see if anyone else is transmitting at that moment. If the channel is busy, the station waits until it becomes idle. When the station detects an idle channel, it transmits a frame. If a collision occurs, the station waits a random amount of time and starts all over again. The protocol is called 1-persistent because the station transmits with a probability of 1 whenever it finds the channel idle.

The propagation delay has an important effect on the performance of the protocol. There is a small chance that just after a station begins sending, another station will become ready to send and sense the channel. If the first station's signal has not yet reached the second one, the latter will sense an idle channel and will also begin sending, resulting in a collision. The longer the propagation delay, the more important this effect becomes, and the worse the performance of the protocol.

Even if the propagation delay is zero, there will still be collisions. If two stations become ready in the middle of a third station's transmission, both will wait politely until the transmission ends and then both will begin transmitting exactly simultaneously, resulting in a collision. If they were not so impatient, there would be fewer collisions. Even so, this protocol is far better than pure ALOHA, because both stations have the decency to desist from interfering with the third station's frame. Intuitively, this will lead to a higher performance than pure ALOHA. Exactly the same holds for slotted ALOHA.

A second carrier sense protocol is **nonpersistent CSMA**. In this protocol, a conscious attempt is made to be less greedy than in the previous one. Before sending, a station senses the channel. If no one else is sending, the station begins doing so itself. However, if the channel is already in use, the station does not continually sense it for the purpose of seizing it immediately upon detecting the end of the previous transmission. Instead, it waits a random period of time and then repeats the algorithm. Intuitively this algorithm should lead to better channel utilization and longer delays than 1-persistent CSMA.

The last protocol is **p-persistent CSMA**. It applies to slotted channels and works as follows. When a station becomes ready to send, it senses the channel. If it is idle, it transmits with a probability p. With a probability $q = 1 - p$ it defers until the next slot. If that slot is also idle, it either transmits or defers again, with probabilities p and q. This process is repeated until either the frame has been transmitted or another station has begun transmitting. In the latter case, it acts as if there had been a collision (i.e., it waits a random time and starts again). If the station initially senses the channel busy, it waits until the next slot and applies the above algorithm. Figure 4-4 shows the throughput versus offered traffic for all three protocols, as well as pure and slotted ALOHA.

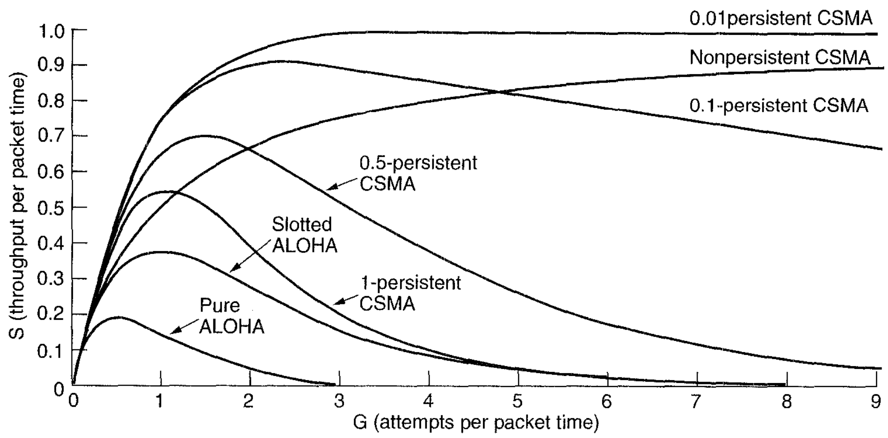

Fig. 4-4. Comparison of the channel utilization versus load for various random access protocols.

CSMA with Collision Detection

Persistent and nonpersistent CSMA protocols are clearly an improvement over ALOHA because they ensure that no station begins to transmit when it senses the channel busy. Another improvement is for stations to abort their transmissions as soon as they detect a collision. In other words, if two stations sense the channel to be idle and begin transmitting simultaneously, they will both detect the collision almost immediately. Rather than finish transmitting their frames, which are irretrievably garbled anyway, they should abruptly stop transmitting as soon as the collision is detected. Quickly terminating damaged frames saves time and bandwidth. This protocol, known as **CSMA/CD** (**Carrier Sense Multiple Access with Collision Detection**), is widely used on LANs in the MAC sublayer.

CSMA/CD, as well as many other LAN protocols, uses the conceptual model of Fig. 4-5. At the point marked t_0, a station has finished transmitting its frame. Any other station having a frame to send may now attempt to do so. If two or more stations decide to transmit simultaneously, there will be a collision. Collisions can be detected by looking at the power or pulse width of the received signal and comparing it to the transmitted signal.

After a station detects a collision, it aborts its transmission, waits a random period of time, and then tries again, assuming that no other station has started transmitting in the meantime. Therefore, our model for CSMA/CD will consist of alternating contention and transmission periods, with idle periods occurring when all stations are quiet (e.g., for lack of work).

Now let us look closely at the details of the contention algorithm. Suppose

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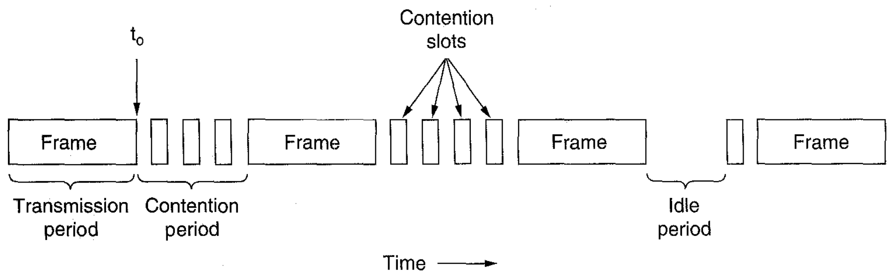

Fig. 4-5. CSMA/CD can be in one of three states: contention, transmission, or idle.

that two stations both begin transmitting at exactly time t_0. How long will it take them to realize that there has been a collision? The answer to this question is vital to determining the length of the contention period, and hence what the delay and throughput will be. The minimum time to detect the collision is then just the time it takes the signal to propagate from one station to the other.

Based on this reasoning, you might think that a station not hearing a collision for a time equal to the full cable propagation time after starting its transmission could be sure it had seized the cable. By "seized," we mean that all other stations knew it was transmitting and would not interfere. This conclusion is wrong. Consider the following worst-case scenario. Let the time for a signal to propagate between the two farthest stations be τ. At t_0, one station begins transmitting. At $\tau - \varepsilon$, an instant before the signal arrives at the most distant station, that station also begins transmitting. Of course, it detects the collision almost instantly and stops, but the little noise burst caused by the collision does not get back to the original station until time $2\tau - \varepsilon$. In other words, in the worst case a station cannot be sure that it has seized the channel until it has transmitted for 2τ without hearing a collision. For this reason we will model the contention interval as a slotted ALOHA system with slot width 2τ. On a 1-km long coaxial cable, $\tau \approx 5$ µsec. For simplicity we will assume that each slot contains just 1 bit. Once the channel has been seized, a station can transmit at any rate it wants to, of course, not just at 1 bit per 2τ sec.

It is important to realize that collision detection is an *analog* process. The station's hardware must listen to the cable while it is transmitting. If what it reads back is different from what it is putting out, it knows a collision is occurring. The implication is that the signal encoding must allow collisions to be detected (e.g., a collision of two 0-volt signals may well be impossible to detect). For this reason, special encoding is commonly used.

CSMA/CD is an important protocol. Later in this chapter we will study one version of it, IEEE 802.3 (Ethernet), which is an international standard.

To avoid any misunderstanding, it is worth noting that no MAC-sublayer protocol guarantees reliable delivery. Even in the absence of collisions, the receiver may not have copied the frame correctly due to various reasons (e.g., lack of buffer space or a missed interrupt).

4.2.3. Collision-Free Protocols

Although collisions do not occur with CSMA/CD once a station has unambiguously seized the channel, they can still occur during the contention period. These collisions adversely affect the system performance, especially when the cable is long (i.e., large τ) and the frames short. As very long, high-bandwidth fiber optic networks come into use, the combination of large τ and short frames will become an increasingly serious problem. In this section, we will examine some protocols that resolve the contention for the channel without any collisions at all, not even during the contention period.

In the protocols to be described, we make the assumption that there are N stations, each with a unique address from 0 to $N-1$ "wired" into it. That some stations may be inactive part of the time does not matter. The basic question remains: Which station gets the channel after a successful transmission? We continue using the model of Fig. 4-5 with its discrete contention slots.

A Bit-Map Protocol

In our first collision-free protocol, the **basic bit-map method**, each contention period consists of exactly N slots. If station 0 has a frame to send, it transmits a 1 bit during the zeroth slot. No other station is allowed to transmit during this slot. Regardless of what station 0 does, station 1 gets the opportunity to transmit a 1 during slot 1, but only if it has a frame queued. In general, station j may announce the fact that it has a frame to send by inserting a 1 bit into slot j. After all N slots have passed by, each station has complete knowledge of which stations wish to transmit. At that point, they begin transmitting in numerical order (see Fig. 4-6).

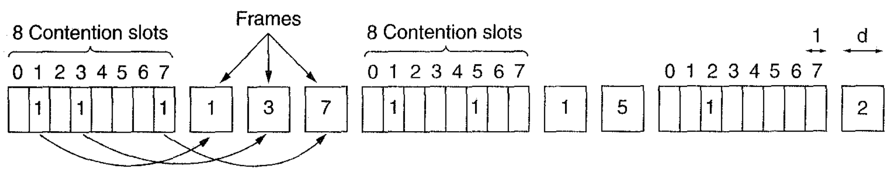

Fig. 4-6. The basic bit-map protocol.

Since everyone agrees on who goes next, there will never be any collisions. After the last ready station has transmitted its frame, an event all stations can easily monitor, another N bit contention period is begun. If a station becomes ready just after its bit slot has passed by, it is out of luck and must remain silent until every station has had a chance and the bit map has come around again. Protocols like this in which the desire to transmit is broadcast before the actual transmission are called **reservation protocols**.

Let us briefly analyze the performance of this protocol. For convenience, we will measure time in units of the contention bit slot, with data frames consisting of d time units. Under conditions of low load, the bit map will simply be repeated over and over, for lack of data frames.

Consider the situation from the point of view of a low-numbered station, such as 0 or 1. Typically, when it becomes ready to send, the "current" slot will be somewhere in the middle of the bit map. On the average, the station will have to wait $N/2$ slots for the current scan to finish and another full N slots for the following scan to run to completion before it may begin transmitting.

The prospects for high-numbered stations are brighter. Generally, these will only have to wait half a scan ($N/2$ bit slots) before starting to transmit. High-numbered stations rarely have to wait for the next scan. Since low-numbered stations must wait on the average $1.5N$ slots and high-numbered stations must wait on the average $0.5N$ slots, the mean for all stations is N slots. The channel efficiency at low load is easy to compute. The overhead per frame is N bits, and the amount of data is d bits, for an efficiency of $d/(N + d)$.

At high load, when all the stations have something to send all the time, the N bit contention period is prorated over N frames, yielding an overhead of only 1 bit per frame, or an efficiency of $d/(d + 1)$. The mean delay for a frame is equal to the sum of the time it queues inside its station, plus an additional $N(d + 1)/2$ once it gets to the head of its internal queue.

Binary Countdown

A problem with the basic bit-map protocol is that the overhead is 1 bit per station. We can do better than that by using binary station addresses. A station wanting to use the channel now broadcasts its address as a binary bit string, starting with the high-order bit. All addresses are assumed to be the same length. The bits in each address position from different stations are BOOLEAN ORed together. We will call this protocol **binary countdown**. It is used in Datakit (Fraser, 1987).

To avoid conflicts, an arbitration rule must be applied: as soon as a station sees that a high-order bit position that is 0 in its address has been overwritten with a 1, it gives up. For example, if stations 0010, 0100, 1001, and 1010 are all trying to get the channel, in the first bit time the stations transmit 0, 0, 1, and 1, respectively. These are ORed together to form a 1. Stations 0010 and 0100 see the 1 and know that a higher-numbered station is competing for the channel, so they give up for the current round. Stations 1001 and 1010 continue.

The next bit is 0, and both stations continue. The next bit is 1, so station 1001 gives up. The winner is station 1010, because it has the highest address. After winning the bidding, it may now transmit a frame, after which another bidding cycle starts. The protocol is illustrated in Fig. 4-7.

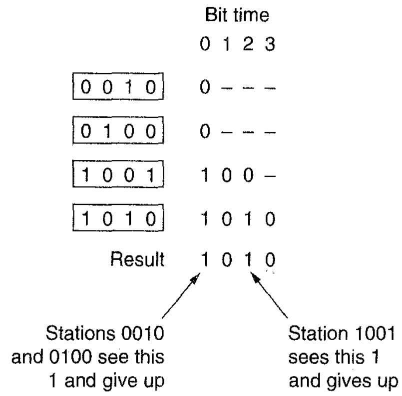

Fig. 4-7. The binary countdown protocol. A dash indicates silence.

The channel efficiency of this method is $d/(d + \ln N)$. If, however, the frame format has been cleverly chosen so that the sender's address is the first field in the frame, even these $\ln N$ bits are not wasted, and the efficiency is 100 percent.

Mok and Ward (1979) have described a variation of binary countdown using a parallel rather than a serial interface. They also suggest using virtual station numbers, with the virtual station numbers from 0 up to and including the successful station being circularly permuted after each transmission, in order to give higher priority to stations that have been silent unusually long. For example, if stations *C*, *H*, *D*, *A*, *G*, *B*, *E*, *F* have priorities 7, 6, 5, 4, 3, 2, 1, and 0, respectively, then a successful transmission by *D* puts it at the end of the list, giving a priority order of *C*, *H*, *A*, *G*, *B*, *E*, *F*, *D*. Thus *C* remains virtual station 7, but *A* moves up from 4 to 5 and *D* drops from 5 to 0. Station *D* will now only be able to acquire the channel if no other station wants it.

4.2.4. Limited-Contention Protocols

We have now considered two basic strategies for channel acquisition in a cable network: contention, as in CSMA, and collision-free methods. Each strategy can be rated as to how well it does with respect to the two important performance measures, delay at low load and channel efficiency at high load. Under conditions of light load, contention (i.e., pure or slotted ALOHA) is preferable due to its low delay. As the load increases, contention becomes increasingly less

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attractive, because the overhead associated with channel arbitration becomes greater. Just the reverse is true for the collision-free protocols. At low load, they have high delay, but as the load increases, the channel efficiency improves rather than gets worse as it does for contention protocols.

Obviously, it would be nice if we could combine the best properties of the contention and collision-free protocols, arriving at a new protocol that used contention at low loads to provide low delay, but used a collision-free technique at high load to provide good channel efficiency. Such protocols, which we will call **limited contention protocols**, do, in fact, exist, and will conclude our study of carrier sense networks.

Up until now the only contention protocols we have studied have been symmetric, that is, each station attempts to acquire the channel with some probability, p, with all stations using the same p. Interestingly enough, the overall system performance can sometimes be improved by using a protocol that assigns different probabilities to different stations.

Before looking at the asymmetric protocols, let us quickly review the performance of the symmetric case. Suppose that k stations are contending for channel access. Each has a probability p of transmitting during each slot. The probability that some station successfully acquires the channel during a given slot is then $kp(1-p)^{k-1}$. To find the optimal value of p, we differentiate with respect to p, set the result to zero, and solve for p. Doing so, we find that the best value of p is $1/k$. Substituting $p = 1/k$ we get

$$\text{Pr[success with optimal } p] = \left(\frac{k-1}{k}\right)^{k-1} \tag{4-4}$$

This probability is plotted in Fig. 4-8. For small numbers of stations, the chances of success are good, but as soon as the number of stations reaches even five, the probability has dropped close to its asymptotic value of $1/e$.

From Fig. 4-8, it is fairly obvious that the probability of some station acquiring the channel can be increased only by decreasing the amount of competition. The limited-contention protocols do precisely that. They first divide the stations up into (not necessarily disjoint) groups. Only the members of group 0 are permitted to compete for slot 0. If one of them succeeds, it acquires the channel and transmits its frame. If the slot lies fallow or if there is a collision, the members of group 1 contend for slot 1, etc. By making an appropriate division of stations into groups, the amount of contention for each slot can be reduced, thus operating each slot near the left end of Fig. 4-8.

The trick is how to assign stations to slots. Before looking at the general case, let us consider some special cases. At one extreme, each group has but one member. Such an assignment guarantees that there will never be collisions, because at most one station is contending per slot. We have seen such protocols before (e.g., binary countdown). The next special case is to assign two stations per group. The probability that both will try to transmit during a slot is p^2, which

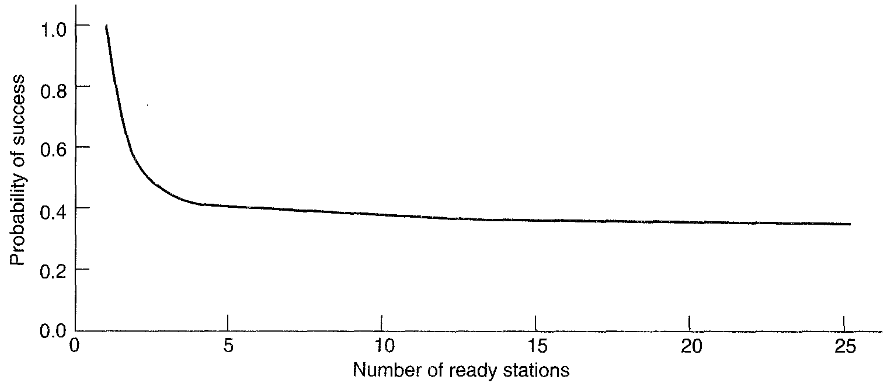

Fig. 4-8. Acquisition probability for a symmetric contention channel.

for small p is negligible. As more and more stations are assigned to the same slot, the probability of a collision grows, but the length of the bit-map scan needed to give everyone a chance shrinks. The limiting case is a single group containing all stations (i.e., slotted ALOHA). What we need is a way to assign stations to slots dynamically, with many stations per slot when the load is low and few (or even just one) station per slot when the load is high.

The Adaptive Tree Walk Protocol

One particularly simple way of performing the necessary assignment is to use the algorithm devised by the U.S. Army for testing soldiers for syphilis during World War II (Dorfman, 1943). In short, the Army took a blood sample from N soldiers. A portion of each sample was poured into a single test tube. This mixed sample was then tested for antibodies. If none were found, all the soldiers in the group were declared healthy. If antibodies were present, two new mixed samples were prepared, one from soldiers 1 through $N/2$ and one from the rest. The process was repeated recursively until the infected soldiers were determined.

For the computer version of this algorithm (Capetanakis, 1979) it is convenient to think of the stations as the leaves of a binary tree, as illustrated in Fig. 4-9. In the first contention slot following a successful frame transmission, slot 0, all stations are permitted to try to acquire the channel. If one of them does so, fine. If there is a collision, then during slot 1 only those stations falling under node 2 in the tree may compete. If one of them acquires the channel, the slot following the frame is reserved for those stations under node 3. If, on the other hand, two or more stations under node 2 want to transmit, there will be a collision during slot 1, in which case it is node 4's turn during slot 2.

In essence, if a collision occurs during slot 0, the entire tree is searched, depth

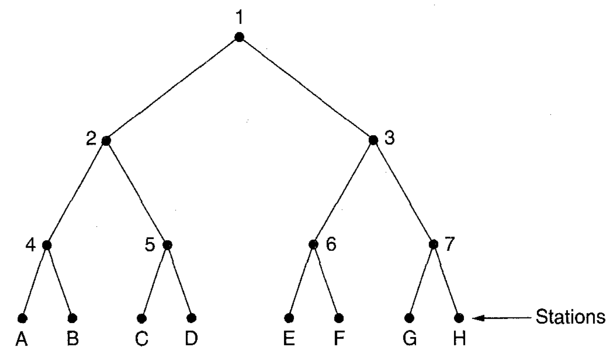

Fig. 4-9. The tree for eight stations.

first, to locate all ready stations. Each bit slot is associated with some particular node in the tree. If a collision occurs, the search continues recursively with the node's left and right children. If a bit slot is idle or if there is only one station that transmits in it, the searching of its node can stop, because all ready stations have been located. (Were there more than one, there would have been a collision.)

When the load on the system is heavy, it is hardly worth the effort to dedicate slot 0 to node 1, because that makes sense only in the unlikely event that precisely one station has a frame to send. Similarly, one could argue that nodes 2 and 3 should be skipped as well for the same reason. Put in more general terms, at what level in the tree should the search begin? Clearly, the heavier the load, the farther down the tree the search should begin. We will assume that each station has a good estimate of the number of ready stations, q, for example, from monitoring recent traffic.

To proceed, let us number the levels of the tree from the top, with node 1 in Fig. 4-9 at level 0, nodes 2 and 3 at level 1, etc. Notice that each node at level i has a fraction 2^{-i} of the stations below it. If the q ready stations are uniformly distributed, the expected number of them below a specific node at level i is just $2^{-i}q$. Intuitively, we would expect the optimal level to begin searching the tree as the one at which the mean number of contending stations per slot is 1, that is, the level at which $2^{-i}q = 1$. Solving this equation we find that $i = \log_2 q$.

Numerous improvements to the basic algorithm have been discovered and are discussed in some detail by Bertsekas and Gallager (1992). For example, consider the case of stations G and H being the only ones wanting to transmit. At node 1 a collision will occur, so 2 will be tried and discovered idle. It is pointless to probe node 3 since it is guaranteed to have a collision (we know that two or more stations under 1 are ready and none of them are under 2 so they must all be under 3). The probe of 3 can be skipped and 6 tried next. When this probe also turns up nothing, 7 can be skipped and node G tried next.

4.2.5. Wavelength Division Multiple Access Protocols

A different approach to channel allocation is to divide the channel into subchannels using FDM, TDM, or both, and dynamically allocate them as needed. Schemes like this are commonly used on fiber optic LANs in order to permit different conversations to use different wavelengths (i.e., frequencies) at the same time. In this section we will examine one such protocol (Humblet et al., 1992).

A simple way to build an all-optical LAN is to use a passive star coupler (see Fig. 2-10). In effect, two fibers from each station are fused to a glass cylinder. One fiber is for output to the cylinder and one is for input from the cylinder. Light output by any station illuminates the cylinder and can be detected by all the other stations. Passive stars can handle hundreds of stations.

To allow multiple transmissions at the same time, the spectrum is divided up into channels (wavelength bands), as shown in Fig. 2-24. In this protocol, **WDMA (Wavelength Division Multiple Access)**, each station is assigned two channels. A narrow channel is provided as a control channel to signal the station, and a wide channel is provided so the station can output data frames.

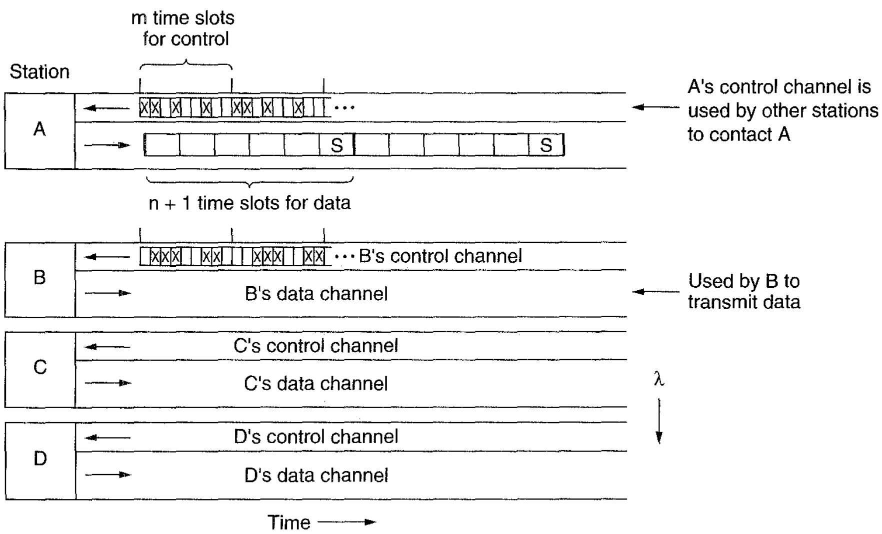

Fig. 4-10. Wavelength division multiple access.

Each channel is divided up into groups of time slots, as depicted in Fig. 4-10. Let us call the number of slots in the control channel m and the number of slots in the data channel $n + 1$, where n of these are for data and the last one is used by the station to report on its status (mainly, which slots on both channels are free). On both channels, the sequence of slots repeats endlessly, with slot 0 being marked in a special way so latecomers can detect it. All channels are synchronized by a single global clock.

The protocol supports three classes of traffic: (1) constant data rate connection-oriented traffic, such as uncompressed video, (2) variable data rate connection-oriented traffic, such as file transfer, and (3) datagram traffic, such as UDP packets. For the two connection-oriented protocols, the idea is that for *A* to communicate with *B*, it must first insert a CONNECTION REQUEST frame in a free slot on *B*'s control channel. If *B* accepts, communication can take place on *A*'s data channel.

Each station has two transmitters and two receivers, as follows:

1. A fixed-wavelength receiver for listening to its own control channel.
2. A tunable transmitter for sending on other station's control channel.
3. A fixed-wavelength transmitter for outputting data frames.
4. A tunable receiver for selecting a data transmitter to listen to.

In other words, every station listens to its own control channel for incoming requests but has to tune to the transmitter's wavelength to get the data. Wavelength tuning is done by a Fabry-Perot or Mach-Zehnder interferometer that filters out all wavelengths except the desired wavelength band.

Let us now consider how station *A* sets up a class 2 communication channel with station *B* for, say, file transfer. First, *A* tunes its data receiver to *B*'s data channel and waits for the status slot. This slot tells which control slots are currently assigned and which are free. In Fig. 4-10, for example, we see that of *B*'s eight control slots, 0, 4, and 5 are free. The rest are occupied (indicated by crosses).

A picks one of the free control slots, say, 4, and inserts its CONNECTION REQUEST message there. Since *B* constantly monitors its control channel, it sees the request and grants it by assigning slot 4 to *A*. This assignment is announced in the status slot of the control channel. When *A* sees the announcement, it knows it has a unidirectional connection. If *A* asked for a two-way connection, *B* now repeats the same algorithm with *A*.

It is possible that at the same time *A* tried to grab *B*'s control slot 4, *C* did the same thing. Neither will get it, and both will notice the failure by monitoring the status slot in *B*'s control channel. They now each wait a random amount of time and try again later.

At this point, each party has a conflict-free way to send short control messages to the other one. To perform the file transfer, *A* now sends *B* a control message saying, for example, "Please watch my next data output slot 3. There is a data frame for you in it." When *B* gets the control message, it tunes its receiver to *A*'s output channel to read the data frame. Depending on the higher-layer protocol, *B* can use the same mechanism to send back an acknowledgement if it wishes.

Note that a problem arises if both A and C have connections to B and each of them suddenly tells B to look at slot 3. B will pick one of these at random, and the other transmission will be lost.

For constant rate traffic, a variation of this protocol is used. When A asks for a connection, it simultaneously says something like: Is it all right if I send you a frame in every occurrence of slot 3? If B is able to accept (i.e., has no previous commitment for slot 3), a guaranteed bandwidth connection is established. If not, A can try again with a different proposal, depending on which output slots it has free.

Class 3 (datagram) traffic uses yet another variation. Instead of writing a CONNECTION REQUEST message into the control slot it just found (4), it writes a DATA FOR YOU IN SLOT 3 message. If B is free during the next data slot 3, the transmission will succeed. Otherwise, the data frame is lost. In this manner, no connections are ever needed.

Several variants of the entire protocol are possible. For example, instead of giving each station its own control channel, a single control channel can be shared by all stations. Each station is assigned a block of slots in each group, effectively multiplexing multiple virtual channels onto one physical one.

It is also possible to make do with a single tunable transmitter and a single tunable receiver per station by having each station's channel be divided up into m control slots followed by $n + 1$ data slots. The disadvantage here is that senders have to wait longer to capture a control slot and consecutive data frames are further apart because some control information is in the way.

Numerous other WDMA protocols have been proposed, differing in the details. Some have one control channel, some have multiple control channels. Some take propagation delay into account, others do not; some make tuning time an explicit part of the model, others ignore it. The protocols also differ in terms of processing complexity, throughput and scalability. For more information see (Bogineni et al., 1993; Chen, 1994; Chen and Yum, 1991; Jia and Mukherjee, 1993; Levine and Akyildiz, 1995; and Williams et al., 1993).

4.2.6. Wireless LAN Protocols

As the number of portable computing and communication devices grows, so does the demand to connect them to the outside world. Even the very first portable telephones had the ability to connect to other telephones. The first portable computers did not have this capability, but soon afterward, modems became commonplace. To go on-line, these computers had to be plugged into a telephone wall socket. Requiring a wired connection to the fixed network meant that the computers were portable, but not mobile.

To achieve true mobility, portable computers need to use radio (or infrared) signals for communication. In this manner, dedicated users can read and send email while driving or boating. A system of portable computers that communicate by radio can be regarded as a wireless LAN. These LANs have somewhat different properties than conventional LANs and require special MAC sublayer protocols. In this section we will examine some of these protocols. More information about wireless LANs can be found in (Davis and McGuffin, 1995; and Nemzow, 1995).

A common configuration for a wireless LAN is an office building with base stations strategically placed around the building. All the base stations are wired together using copper or fiber. If the transmission power of the base stations and portables is adjusted to have a range of 3 or 4 meters, then each room becomes a single cell, and the entire building becomes a large cellular system, as in the traditional cellular telephony systems we studied in Chap. 2. Unlike cellular telephone systems, each cell has only one channel, covering the entire available bandwidth. Typically its bandwidth is 1 to 2 Mbps.

In our discussions below, we will make the simplifying assumption that all radio transmitters have some fixed range. When a receiver is within range of two active transmitters, the resulting signal will generally be garbled and useless (but with certain exceptions to be discussed later). It is important to realize that in some wireless LANs, not all stations are within range of one another, which leads to a variety of complications. Furthermore, for indoor wireless LANs, the presence of walls between stations can have a major impact on the effective range of each station.

A naive approach to using a wireless LAN might be to try CSMA: just listen for other transmissions and only transmit if no one else is doing so. The trouble is, this protocol is not really appropriate because what matters is interference at the receiver, not at the sender. To see the nature of the problem, consider Fig. 4-11, where four wireless stations are illustrated. For our purposes, it does not matter which are base stations and which are portables. The radio range is such that A and B are within each other's range and can potentially interfere with one another. C can also potentially interfere with both B and D, but not with A.

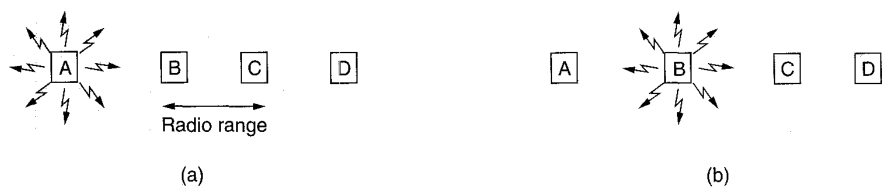

Fig. 4-11. A wireless LAN. (a) A transmitting. (b) B transmitting.

First consider what happens when A is transmitting to B, as depicted in Fig. 4-11(a). If C senses the medium, it will not hear A because A is out of range, and thus falsely conclude that it can transmit. If C does start transmitting, it will interfere at B, wiping out the frame from A. The problem of a station not being

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able to detect a potential competitor for the medium because the competitor is too far away is sometimes called the **hidden station problem**.

Now let us consider the reverse situation: *B* transmitting to *A*, as shown in Fig. 4-11(b). If *C* senses the medium, it will hear an ongoing transmission and falsely conclude that it may not send to *D*, when in fact such a transmission would cause bad reception only in the zone between *B* and *C*, where neither of the intended receivers is located. This situation is sometimes called the **exposed station problem**.

The problem is that before starting a transmission, a station really wants to know whether or not there is activity around the receiver. CSMA merely tells it whether or not there is activity around the station sensing the carrier. With a wire, all signals propagate to all stations so only one transmission can take place at once anywhere in the system. In a system based on short-range radio waves, multiple transmissions can occur simultaneously if they all have different destinations and these destinations are out of range of one another.

Another way to think about this problem is to imagine an office building in which every employee has a wireless portable computer. Suppose that Linda wants to send a message to Milton. Linda's computer senses the local environment and, detecting no activity, starts sending. However, there may still be a collision in Milton's office because a third party may currently be sending to him from a location so far from Linda that her computer could not detect it.

MACA and MACAW

An early protocol designed for wireless LANs is **MACA (Multiple Access with Collision Avoidance)** (Karn, 1990). It was used as the basis for the IEEE 802.11 wireless LAN standard. The basic idea behind it is for the sender to stimulate the receiver into outputting a short frame, so stations nearby can detect this transmission and avoid transmitting themselves for the duration of the upcoming (large) data frame. MACA is illustrated in Fig. 4-12.

Let us consider how *A* sends a frame to *B*. *A* starts by sending an RTS (Request To Send) frame to *B*, as shown in Fig. 4-12(a). This short frame (30 bytes) contains the length of the data frame that will eventually follow. Then *B* replies with a CTS (Clear To Send) frame, as shown in Fig. 4-12(b). The CTS frame contains the data length (copied from the RTS frame). Upon receipt of the CTS frame, *A* begins transmission.

Now let us see how stations overhearing either of these frames react. Any station hearing the RTS is clearly close to *A* and must remain silent long enough for the CTS to be transmitted back to *A* without conflict. Any station hearing the CTS is clearly close to *B* and must remain silent during the upcoming data transmission, whose length it can tell by examining the CTS frame.

In Fig. 4-12, *C* is within range of *A* but not within range of *B*. Therefore it hears the RTS from *A* but not the CTS from *B*. As long as it does not interfere with

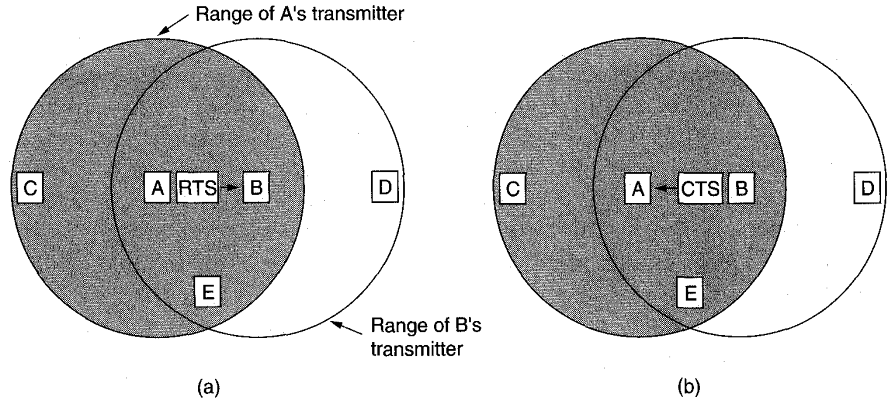

Fig. 4-12. The MACA protocol. (a) *A* sending an RTS to *B*. (b) *B* responding with a CTS to *A*.

the CTS, it is free to transmit while the data frame is being sent. In contrast, *D* is within range of *B* but not *A*. It does not hear the RTS but does hear the CTS. Hearing the CTS tips it off that it is close to a station that is about to receive a frame, so it defers from sending anything until that frame is expected to be finished. Station *E* hears both control messages, and like *D*, must be silent until the data frame is complete.

Despite these precautions, collisions can still occur. For example, *B* and *C* could both send RTS frames to *A* at the same time. These will collide and be lost. In the event of a collision, an unsuccessful transmitter (i.e., one that does not hear a CTS within the expected time interval) waits a random amount of time and tries again later. The algorithm used is binary exponential backoff, which we will study when we come to the IEEE 802.3 LAN.

Based on simulation studies of MACA, Bharghavan et al. (1994) fine tuned MACA to improve its performance and renamed their new protocol **MACAW**. To start with, they noticed that without data link layer acknowledgements, lost frames were not retransmitted until the transport layer noticed their absence, much later. They solved this problem by introducing an ACK frame after each successful data frame. They also observed that CSMA has some utility—namely to keep a station from transmitting an RTS at the same time another nearby station is also doing so to the same destination, so carrier sensing was added. In addition, they decided to run the backoff algorithm separately for each data stream (source-destination pair), rather than for each station. This change improves the fairness of the protocol. Finally, they added a mechanism for stations to exchange information about congestion, and a way to make the backoff algorithm react less violently to temporary problems, to improve system performance.

4.2.7. Digital Cellular Radio

A second form of wireless networking is digital cellular radio, the successor to the AMPS system we studied in Chap. 2. Digital cellular radio presents a somewhat different environment than do wireless LANs and uses different protocols. In particular, it is oriented toward telephony, which requires connections lasting for minutes, rather than milliseconds, so it is more efficient to do channel allocation per call rather than per frame. Nevertheless, the techniques are equally valid for data traffic. In this section we will look at three radically different approaches to channel allocation for wireless digital radio systems, GSM, CDPD, and CDMA.

GSM—Global System for Mobile Communications

The first generation of cellular phones were analog, as described in Chap. 2, but the current generation is digital, using packet radio. Digital transmission has several advantages over analog for mobile communication. First, voice, data, and fax, can be integrated into a single system. Second, as better speech compression algorithms are discovered, less bandwidth will be needed per channel. Third, error-correcting codes can be used to improve transmission quality. Finally, digital signals can be encrypted for security.

Although it might have been nice if the whole world had adopted the same digital standard, such is not the case. The U.S. system, IS-54, and the Japanese system, JDC, have been designed to be compatible with each country's existing analog system, so each AMPS channel could be used either for analog or digital communication.

In contrast, the European digital system, **GSM** (**Global System for Mobile communications**), has been designed from scratch as a fully digital system, without any compromises for the sake of backward compatibility (e.g., having to use the existing frequency slots). Since GSM is also further along than the U.S. system and is currently in use in over 50 countries, inside and outside of Europe, we will use it as an example of digital cellular radio.

GSM was originally designed for use in the 900-MHz band. Later, frequencies were allocated at 1800 MHz, and a second system, closely patterned on GSM, was set up there. The latter is called **DCS 1800**, but it is essentially GSM.

The GSM standard is over 5000 [sic] pages long. A large fraction of this material relates to engineering aspects of the system, especially the design of receivers to handle multipath signal propagation, and synchronizing transmitters and receivers.

A GSM system has up to a maximum of 200 full-duplex channels per cell. Each channel consists of a downlink frequency (from the base station to the mobile stations) and an uplink frequency (from the mobile stations to the base station). Each frequency band is 200 kHz wide as shown in Fig. 4-13.

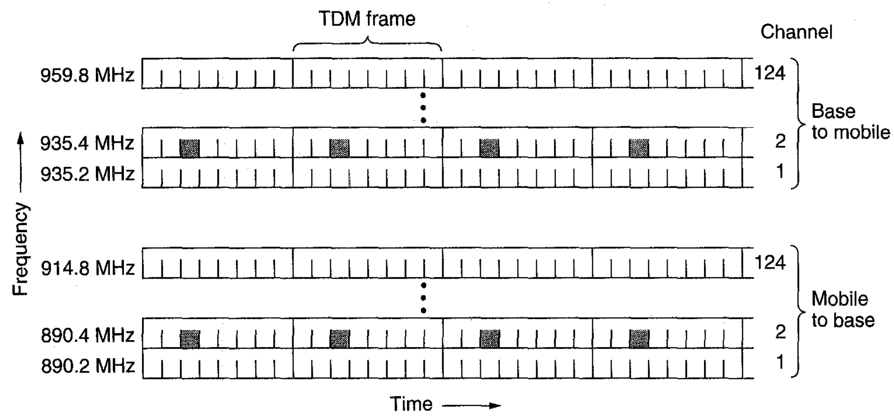

Fig. 4-13. GSM uses 124 frequency channels, each of which use an eight-slot TDM system.

Each of the 124 frequency channels supports eight separate connections using time division multiplexing. Each currently active station is assigned one time slot on one channel. Theoretically, 992 channels can be supported in each cell, but many of them are not available, to avoid frequency conflicts with neighboring cells. In Fig. 4-13, the eight shaded time slots all belong to the same channel, four of them in each direction. If the mobile station assigned to 890.4/935.4 MHz and slot 2 wanted to transmit to the base station, it would use the lower four shaded slots (and the ones following them in time), putting some data in each slot until all the data had been sent.

The TDM slots shown in Fig. 4-13 are part of a complex framing hierarchy. Each TDM slot has a specific structure, and groups of TDM slots form multiframes, also with a specific structure. A simplified version of this hierarchy is shown in Fig. 4-14. Here we can see that each TDM slot consists of a 148-bit data frame. Each data frame starts and ends with three 0 bits, for frame delineation purposes. It also contains two 57-bit *Information* fields, each one having a control bit that indicates whether the following *Information* field is for voice or data. Between the *Information* fields is a 26-bit *Sync* (training) field that is used by the receiver to synchronize to the sender's frame boundaries. A data frame is transmitted in 547 μsec, but a transmitter is only allowed to send one data frame every 4.615 msec, since it is sharing the channel with seven other stations. The gross rate of each channel is 270,833 bps, divided among eight users. Discounting all the overhead, each connection can send one compressed voice signal or 9600 bps of data.

As can be seen from Fig. 4-14, eight data frames make up a TDM frame, and 26 TDM frames make up a 120-msec multiframe. Of the 26 TDM frames in a

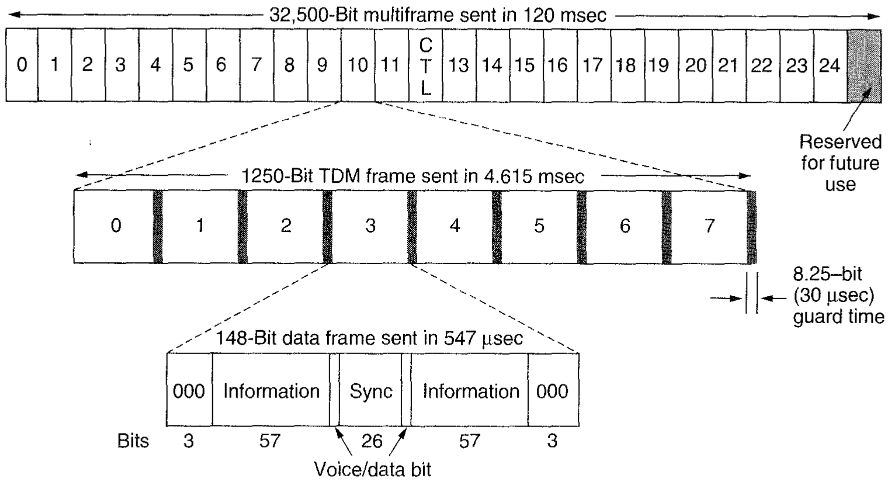

Fig. 4-14. A portion of the GSM framing structure.

multiframe, slot 12 is used for control and slot 25 is reserved for future use, so only 24 are available for user traffic.

However, in addition to the 26-slot multiframe shown in Fig. 4-14, a 51-slot multiframe (not shown) is also used. Some of these slots are used to hold several control channels used to manage the system. The **broadcast control channel** is a continuous stream of output from the base station containing its identity and the channel status. All mobile stations monitor its signal strength to see when they have moved into a new cell.

The **dedicated control channel** is used for location updating, registration, and call setup. In particular, each base station maintains a database of mobile stations currently under its jurisdiction. Information needed to maintain this database is sent on the dedicated control channel.

Finally, there is the **common control channel**, which is split up into three logical subchannels. The first of these subchannels is the **paging channel**, which the base station uses to announce incoming calls. Each mobile station monitors it continuously to watch for calls it should answer. The second is the **random access channel**, which runs a slotted ALOHA system to allow a mobile station to request a slot on the dedicated control channel. Using this slot, the station can set up a call. The assigned slot is announced on the third subchannel, the **access grant channel**.

All in all, GSM is a fairly complex system. It handles channel access using a combination of slotted ALOHA, FDM and TDM. For more information about GSM, including aspects of the system that we have not discussed, for example, the protocol layering architecture, see (Rahnema, 1993).

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CDPD—Cellular Digital Packet Data

GSM is basically circuit switched. A mobile computer with a special modem can place a call using a GSM telephone the same way it would place one on a hardwired telephone. However, using this strategy is not without problems. For one, handoffs between base stations are frequent, sometimes even with stationary users (base stations can shuffle users around for load balancing), and each handoff results in losing ca. 300 msec of data. For another, GSM can suffer from a high error rate. Typing an "a" and having it echoed as an "m" gets tiresome quickly. Finally, wireless calls are expensive, and costs mount quickly because the charge is per minute of connect time, not per byte sent.

One approach to solving these problems is a packet-switched digital datagram service called **CDPD** (**Cellular Digital Packet Data**). It is built on top of AMPS (see Chap. 2) and entirely compatible with AMPS. Basically, any idle 30-kHz channel can be temporarily grabbed for sending data frames at a gross rate of 19.2 kbps. Because CDPD involves quite a bit of overhead, the net data rate is closer to 9600 bps. Still, a connectionless, wireless datagram system for sending, for example, IP packets, using the existing cellular phone system is an interesting proposition for many users, so its use is growing rapidly.

CDPD follows the OSI model closely. The physical layer deals with the details of modulation and radio transmission, which do not concern us here. Data link, network, and transport protocols also exist but are not of special interest to us either. Instead, we will give a general description of the system and then describe the medium access protocol. For more information about the full CDPD system, see (Quick and Balachandran, 1993).

A CDPD system consists of three kinds of stations: mobile hosts, base stations, and base interface stations (in CDPD jargon: mobile end systems, mobile data base systems, and mobile data intermediate systems, respectively). These stations interact with stationary hosts and standard routers, of the kind found in any WAN. The mobile hosts are the users' portable computers. The base stations are the transmitters that talk to the mobile hosts. The base interface stations are special nodes that interface all the base stations in a CDPD provider's area to a standard (fixed) router for further transmission through the Internet or other WAN. This arrangement is shown in Fig. 4-15.

Three kinds of interfaces are defined in CDPD. The **E-interface** (external to the CDPD provider) connects a CDPD area to a fixed network. This interface must be well defined to allow CDPD to connect to a variety of networks. The **I-interface** (internal to the CDPD provider) connects two CDPD areas together. It must be standardized to allow users to roam between areas. The third one is the **A-interface**, (air interface) between the base station and mobile hosts. This is the most interesting one, so we will now examine it more closely.

Data over the air interface are sent using compression, encryption, and error correction. Units of 274 compressed, encrypted data bits are wrapped in 378-bit

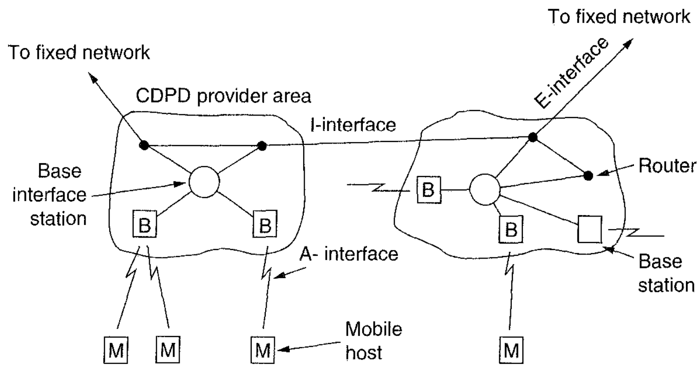

Fig. 4-15. An example CDPD system.

blocks using a Reed-Solomon error correcting code. To each RS block is added seven 6-bit flag words, to form a total of 420-bit blocks. Each 420-bit block is divided up into seven 60-bit microblocks, which are sent consecutively. Each microblock has its own 6-bit flag word, used for indicating channel status. These microblocks go over a 19.2-kbps downlink channel (from the base) or over a second 19.2-kbps uplink channel (to the base), in full-duplex mode. In effect, both the downlink and uplink channel are slotted in time, as a sequence of 60-bit microblocks. Each microblock lasts for 3.125 msec.

Each CDPD cell has only one downlink/uplink pair available for data. The downlink channel is easy to manage since there is only one sender per cell: the base station. All frames sent on it are broadcast, with each mobile host selecting out those destined for it or for everyone.

The tricky part is the uplink channel, for which all mobile hosts wishing to send must contend. When a mobile host has a frame to send, it watches the downlink channel for a flag bit telling whether the current uplink slot is busy or idle. If it is busy, instead of just waiting for the next time slot, it skips a random number of slots and tries again. If it again sees that the uplink channel is busy, it waits a longer random time, and repeats the procedure. The statistically average waiting time doubles with each unsuccessful attempt. When it finally finds the channel supposedly idle, it begins transmitting its microblock.

The point of this algorithm, called **DSMA** (**Digital Sense Multiple Access**), is to prevent all the mobile hosts from jumping on the uplink channel as soon it goes idle. It somewhat resembles the slotted p-persistent CSMA protocol we mentioned earlier, since it, too, uses discrete time slots on both channels.

The trouble is, despite DSMA, a collision with another mobile host is still possible, since two or more of them may pick the same time slot to starting sending. To allow mobile hosts to discover whether or not they have suffered a collision, a flag bit in each microblock tells whether a previous microblock on the uplink channel was received correctly. Unfortunately, the base station cannot make the determination instantly after a microblock terminates, so the correct/incorrect reception of microblock n is delayed until microblock $n + 2$.

Since it cannot tell if its transmission was successful, if a sender has more microblocks to send, it just goes ahead, without having to reacquire the channel. If in the *following* time slot it sees that its *previous* transmission failed, it stops. Otherwise it continues transmitting, up to a certain maximum number of Reed-Solomon blocks, or until the base station sets a flag bit on the downlink channel to indicate that it has heard enough from this particular sender for the moment.

An additional property of CDPD is that data users are second-class citizens. When a new voice call is about to be assigned to a channel currently in use for CDPD, the base station sends a special signal on the downlink, closing down the channel. If the base station already knows the number of the new CDPD channel, it announces it. Otherwise, mobile hosts have to hunt around among a designated set of potential CDPD channels to find it. In this way, CDPD can suck up any idle capacity in a cell, without interfering with the big cash cow, voice.

It should be clear from this description that CDPD was added to the voice system after the latter was already operational, and that its design was subject to the constraint that no changes could be made to the existing voice system. Consequently, when channel selection for voice calls occurs, the algorithm is not aware of the existence of CDPD. This is the reason that the CDPD channel is sometimes suddenly preempted. However, nothing in the design prevents having dedicated CDPD channels. As CDPD grows in popularity, providers are likely to reserve channels exclusively for it.

CDMA—Code Division Multiple Access

GSM might be described as a brute force solution to channel allocation. It uses a combination of practically every known technique (ALOHA, TDM, FDM) intertwined in complex ways. CDPD for single-frame transmissions is fundamentally nonpersistent CSMA. Now we will examine yet another method for allocating a wireless channel, **CDMA** (**Code Division Multiple Access**).

CDMA is completely different from all the other allocation techniques we have studied so far. Some of these have been based on dividing the channel into frequency bands and assigning those statically (FDM) or on demand (wavelength division multiplexing), with the owner using the band indefinitely. Others allocate the channel in bursts, giving stations the entire channel statically (TDM with fixed time slots) or dynamically (ALOHA). CDMA allows each station to transmit over the entire frequency spectrum all the time. Multiple simultaneous transmissions are separated using coding theory. CDMA also relaxes the assumption that colliding frames are totally garbled. Instead, it assumes that multiple signals add linearly.

Before getting into the algorithm, let us consider the cocktail party theory of channel access. In a large room, many pairs of people are conversing. TDM is when all the people are in the middle of the room, but they take turns speaking, first one then another. FDM is when the people group into widely separated clumps, each clump holding its own conversation at the same time as, but still independent of, the others. CDMA is when they are all in the middle of the room talking at once, but with each pair in a different language. The French-speaking couple just hones in on the French, rejecting everything else as noise. Thus the key to CDMA is to be able to extract the desired signal while rejecting everything else as random noise.

In CDMA each bit time is subdivided into m short intervals called **chips**. Typically there are 64 or 128 chips per bit, but in the example given below we will use 8 chips/bit for simplicity.

Each station is assigned a unique m-bit code or **chip sequence**. To transmit a 1 bit, a station sends its chip sequence. To transmit a 0 bit, it sends the one's complement of its chip sequence. No other patterns are permitted. Thus for $m = 8$, if station A is assigned the chip sequence 00011011, it sends a 1 bit by sending 00011011 and a 0 bit by sending 11100100.

Increasing the amount of information to be sent from b bits/sec to mb chips/sec can only be done if the bandwidth available is increased by a factor of m, making CDMA a form of spread spectrum communication (assuming no changes in the modulation or encoding techniques). If we have a 1-MHz band available for 100 stations, with FDM each one would have 10 kHz and could send at 10 kbps (assuming 1 bit per Hz). With CDMA, each station uses the full 1 MHz, so the chip rate is 1 megachip per second. With fewer than 100 chips per bit, the effective bandwidth per station is higher for CDMA than FDM, and the channel allocation problem is also solved, as we will see shortly.

For pedagogical purposes, it is more convenient to use a bipolar notation, with binary 0 being -1 and binary 1 being $+1$. We will show chip sequences in parentheses, so a 1 bit for station A now becomes $(-1\ -1\ -1\ +1\ +1\ -1\ +1\ +1)$. In Fig. 4-16(a) we show the binary chip sequences assigned to four example stations. In Fig. 4-16(b) we show them in our bipolar notation.

Each station has its own unique chip sequence. Let us use the symbol $\mathbf{S}$ to indicate the m-chip vector for station S, and $\overline{\mathbf{S}}$ for its negation. All chip sequences are pairwise **orthogonal**, by which we mean that the normalized inner product of any two distinct chip sequences, $\mathbf{S}$ and $\mathbf{T}$ (written as $\mathbf{S} \bullet \mathbf{T}$) is 0. In mathematical terms,

$$\mathbf{S} \bullet \mathbf{T} \equiv \frac{1}{m} \sum_{i=1}^{m} S_i T_i = 0 \tag{4-5}$$

In plain English, as many pairs are the same as are different. This orthogonality property will prove crucial later on. Note that if $\mathbf{S} \bullet \mathbf{T} = 0$ then $\mathbf{S} \bullet \overline{\mathbf{T}}$ is also 0. The normalized inner product of any chip sequence with itself is 1:

$$\mathbf{S} \bullet \mathbf{S} = \frac{1}{m} \sum_{i=1}^{m} S_i S_i = \frac{1}{m} \sum_{i=1}^{m} S_i^2 = \frac{1}{m} \sum_{i=1}^{m} (\pm 1)^2 = 1$$

(a)	(b)
A: 0 0 0 1 1 0 1 1	A: (−1 −1 −1 +1 +1 −1 +1 +1)
B: 0 0 1 0 1 1 1 0	B: (−1 −1 +1 −1 +1 +1 +1 −1)
C: 0 1 0 1 1 1 0 0	C: (−1 +1 −1 +1 +1 +1 −1 −1)
D: 0 1 0 0 0 0 1 0	D: (−1 +1 −1 −1 −1 −1 +1 −1)

Six examples:

– – 1 –	**C**	S_1 = (−1 +1 −1 +1 +1 +1 −1 −1)
– 1 1 –	**B + C**	S_2 = (−2 0 0 0 +2 +2 0 −2)
1 0 – –	**A** + $\bar{\textbf{B}}$	S_3 = (0 0 −2 +2 0 −2 0 +2)
1 0 1 –	**A** + $\bar{\textbf{B}}$ + **C**	S_4 = (−1 +1 −3 +3 −1 −1 −1 +1)
1 1 1 1	**A + B + C + D**	S_5 = (−4 0 −2 0 +2 0 +2 −2)
1 1 0 1	**A + B** + $\bar{\textbf{C}}$ + **D**	S_6 = (−2 −2 0 −2 0 −2 +4 0)

(c)

$S_1 \bullet C$ = (1 +1 +1 +1 +1 +1 +1 +1)/8 = 1
$S_2 \bullet C$ = (2 +0 +0 +0 +2 +2 +0 +2)/8 = 1
$S_3 \bullet C$ = (0 +0 +2 +2 +0 −2 +0 −2)/8 = 0
$S_4 \bullet C$ = (1 +1 +3 +3 +1 −1 +1 −1)/8 = 1
$S_5 \bullet C$ = (4 +0 +2 +0 +2 +0 −2 +2)/8 = 1
$S_6 \bullet C$ = (2 −2 +0 −2 +0 −2 −4 +0)/8 = −1

(d)

Fig. 4-16. (a) Binary chip sequences for four stations. (b) Bipolar chip sequences. (c) Six examples of transmissions. (d) Recovery of station C's signal.

This follows because each of the m terms in the inner product is 1, so the sum is m. Also note that $\mathbf{S} \bullet \bar{\mathbf{S}} = -1$.

During each bit time, a station can transmit a 1 by sending its chip sequence, it can transmit a 0 by sending the negative of its chip sequence, or it can be silent and transmit nothing. For the moment, we assume that all stations are synchronized in time, so all chip sequences begin at the same instant.

When two or more stations transmit simultaneously, their bipolar signals add linearly. For example, if in one chip period three stations output +1 and one station outputs −1, the result is +2. One can think of this as adding voltages: three stations outputting +1 volts and 1 station outputting −1 volts gives 2 volts.

In Fig. 4-16(c) we see six examples of one or more stations transmitting at the same time. In the first example, C transmits a 1 bit, so we just get C's chip sequence. In the second example, both B and C transmit 1 bits, so we get the sum of their bipolar chip sequences, namely:

$$(-1\ -1\ +1\ -1\ +1\ +1\ +1\ -1) + (-1\ +1\ -1\ +1\ +1\ +1\ -1\ -1) = (-2\ \ 0\ \ 0\ \ 0\ +2\ +2\ \ 0\ -2)$$

In the third example, station A sends a 1 and station B sends a 0. The others are silent. In the fourth example, A and C send a 1 bit while B sends a 0 bit. In the fifth example, all four stations send a 1 bit. Finally, in the last example, A, B, and

D send a 1 bit, while C sends a 0 bit. Note that each of the six sequences S_1 through S_6 given in Fig. 4-16(c) represents only one bit time.

To recover the bit stream of an individual station, the receiver must know that station's chip sequences in advance. It does the recovery by computing the normalized inner product of the received chip sequence (the linear sum of all the stations that transmitted) and the chip sequence of the station whose bit stream it is trying to recover. If the received chip sequence is **S** and the receiver is trying to listen to a station whose chip sequence is **C**, it just computes the normalized inner product, $\mathbf{S} \bullet \mathbf{C}$.

To see why this works, imagine that two stations, A and C, both transmit a 1 bit at the same time that B transmits a 0 bit. The receiver sees the sum: $\mathbf{S} = \mathbf{A} + \overline{\mathbf{B}} + \mathbf{C}$ and computes

$$\mathbf{S} \bullet \mathbf{C} = (\mathbf{A} + \overline{\mathbf{B}} + \mathbf{C}) \bullet \mathbf{C} = \mathbf{A} \bullet \mathbf{C} + \overline{\mathbf{B}} \bullet \mathbf{C} + \mathbf{C} \bullet \mathbf{C} = 0 + 0 + 1 = 1$$

The first two terms vanish because all pairs of chip sequences have been carefully chosen to be orthogonal, as shown in Eq. (4-5). Now it should be clear why this property must be imposed on the chip sequences.

An alternative way of thinking about this situation is to imagine that the three chip sequences all came in separately, rather than summed. Then the receiver would compute the inner product with each one separately and add the results. Due to the orthogonality property, all the inner products except $\mathbf{C} \bullet \mathbf{C}$ would be 0. Adding them and then doing the inner product is in fact the same as doing the inner products and then adding those.

To make the decoding process more concrete, let us consider the six examples of Fig. 4-16(d) again. Suppose that the receiver is interested in extracting the bit sent by station C from each of the six sums S_1 through S_6. It calculates the bit by summing the pairwise products of the received **S** and the **C** vector of Fig. 4-16(b), and then taking 1/8 of the result (since $m = 8$ here). As shown, each time the correct bit is decoded. It is just like speaking French.

In an ideal, noiseless CDMA system, the capacity (i.e., number of stations) can be made arbitrarily large, just as the capacity of a noiseless Nyquist channel can be made arbitrarily large by using more and more bits per sample. In practice, physical limitations reduce the capacity considerably. First, we have assumed that all the chips are synchronized in time. In reality, doing so is impossible. What can be done is that the sender and receiver synchronize by having the sender transmit a long enough known chip sequence that the receiver can lock onto. All the other (unsynchronized) transmissions are then seen as random noise. If there are not too many of them, however, the basic decoding algorithm still works fairly well. A large body of theory exists relating the superposition of chip sequences to noise level (Pickholtz et al., 1982). As one might expect, the longer the chip sequence, the higher the probability of detecting it correctly in the presence of noise. For extra security, the bit sequence can use an error correcting code. Chip sequences never use error correcting codes.

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An implicit assumption in the above discussion is that the power levels of all stations are the same as perceived by the receiver. CDMA is typically used for wireless systems with a fixed base station and many mobile stations at varying distances from it. The power levels received at the base station depend on how far away the transmitters are. A good heuristic here is for each mobile station to transmit to the base station at the inverse of the power level it receives from the base station, so a mobile station receiving a weak signal from the base will use more power than one getting a strong signal. The base station can also give explicit commands to the mobile stations to increase or decrease their transmission power.

We have also assumed that the receiver knows who the sender is. In principle, given enough computing capacity, the receiver can listen to all the senders at once by running the decoding algorithm for each of them in parallel. In real life, suffice it to say that this is easier said than done. CDMA also has many other complicating factors that have been glossed over in this brief introduction. Nevertheless, CDMA is a clever scheme that is being rapidly introduced for wireless mobile communication.

Readers with a solid electrical engineering background who want to gain a deeper understanding of CDMA should read (Viterbi, 1995). An alternative spreading scheme, in which the spreading is over time rather than frequency, is described in (Crespo et al., 1995).

4.3. IEEE STANDARD 802 FOR LANS AND MANS

We have now finished our general discussion of abstract channel allocation protocols, so it is time to see how these principles apply to real systems, in particular, LANs. As discussed in Sec. 1.7.2, IEEE has produced several standards for LANs. These standards, collectively known as **IEEE 802**, include CSMA/CD, token bus, and token ring. The various standards differ at the physical layer and MAC sublayer but are compatible at the data link layer. The IEEE 802 standards have been adopted by ANSI as American National Standards, by NIST as government standards, and by ISO as international standards (known as ISO 8802). They are surprisingly readable (as standards go).

The standards are divided into parts, each published as a separate book. The 802.1 standard gives an introduction to the set of standards and defines the interface primitives. The 802.2 standard describes the upper part of the data link layer, which uses the **LLC** (**Logical Link Control**) protocol. Parts 802.3 through 802.5 describe the three LAN standards, the CSMA/CD, token bus, and token ring standards, respectively. Each standard covers the physical layer and MAC sublayer protocol. The next three sections cover these three systems. Additional information can be found in (Stallings, 1993b).

4.3.1. IEEE Standard 802.3 and Ethernet

The IEEE 802.3 standard is for a 1-persistent CSMA/CD LAN. To review the idea, when a station wants to transmit, it listens to the cable. If the cable is busy, the station waits until it goes idle; otherwise it transmits immediately. If two or more stations simultaneously begin transmitting on an idle cable, they will collide. All colliding stations then terminate their transmission, wait a random time, and repeat the whole process all over again.

The 802.3 standard has an interesting history. The real beginning was the ALOHA system constructed to allow radio communication between machines scattered over the Hawaiian Islands. Later, carrier sensing was added, and Xerox PARC built a 2.94-Mbps CSMA/CD system to connect over 100 personal workstations on a 1-km cable (Metcalfe and Boggs, 1976). This system was called **Ethernet** after the *luminiferous ether*, through which electromagnetic radiation was once thought to propagate. (When the Nineteenth Century British physicist James Clerk Maxwell discovered that electromagnetic radiation could be described by a wave equation, scientists assumed that space must be filled with some ethereal medium in which the radiation was propagating. Only after the famous Michelson-Morley experiment in 1887, did physicists discover that electromagnetic radiation could propagate in a vacuum.)

The Xerox Ethernet was so successful that Xerox, DEC, and Intel drew up a standard for a 10-Mbps Ethernet. This standard formed the basis for 802.3. The published 802.3 standard differs from the Ethernet specification in that it describes a whole family of 1-persistent CSMA/CD systems, running at speeds from 1 to 10-Mbps on various media. Also, the one header field differs between the two (the 802.3 length field is used for packet type in Ethernet). The initial standard also gives the parameters for a 10 Mbps baseband system using 50-ohm coaxial cable. Parameter sets for other media and speeds came later.

Many people (incorrectly) use the name "Ethernet" in a generic sense to refer to all CSMA/CD protocols, even though it really refers to a specific product that almost implements 802.3. We will use the terms "802.3" and "CSMA/CD" except when specifically referring to the Ethernet product in the next few paragraphs.

802.3 Cabling

Since the name "Ethernet" refers to the cable (the ether), let us start our discussion there. Four types of cabling are commonly used, as shown in Fig. 4-17. Historically, **10Base5** cabling, popularly called **thick Ethernet,** came first. It resembles a yellow garden hose, with markings every 2.5 meters to show where the taps go. (The 802.3 standard does not actually *require* the cable to be yellow, but it does *suggest* it.) Connections to it are generally made using **vampire taps**, in which a pin is carefully forced halfway into the coaxial cable's core. The notation 10Base5 means that it operates at 10 Mbps, uses baseband signaling, and can support segments of up to 500 meters.

Name	Cable	Max. segment	Nodes/seg.	Advantages
10Base5	Thick coax	500 m	100	Good for backbones
10Base2	Thin coax	200 m	30	Cheapest system
10Base-T	Twisted pair	100 m	1024	Easy maintenance
10Base-F	Fiber optics	2000 m	1024	Best between buildings

Fig. 4-17. The most common kinds of baseband 802.3 LANs.

Historically, the second cable type was **10Base2** or **thin Ethernet,** which, in contrast to the garden-hose-like thick Ethernet, bends easily. Connections to it are made using industry standard BNC connectors to form T junctions, rather than using vampire taps. These are easier to use and more reliable. Thin Ethernet is much cheaper and easier to install, but it can run for only 200 meters and can handle only 30 machines per cable segment.

Detecting cable breaks, bad taps, or loose connectors can be a major problem with both media. For this reason, techniques have been developed to track them down. Basically, a pulse of known shape is injected into the cable. If the pulse hits an obstacle or the end of the cable, an echo will be generated and sent back. By carefully timing the interval between sending the pulse and receiving the echo, it is possible to localize the origin of the echo. This technique is called **time domain reflectometry**.

The problems associated with finding cable breaks have driven systems toward a different kind of wiring pattern, in which all stations have a cable running to a central **hub**. Usually, these wires are telephone company twisted pairs, since most office buildings are already wired this way, and there are normally plenty of spare pairs available. This scheme is called **10Base-T**.

These three wiring schemes are illustrated in Fig. 4-18. For 10Base5, a **transceiver** is clamped securely around the cable so that its tap makes contact with the inner core. The transceiver contains the electronics that handle carrier detection and collision detection. When a collision is detected, the transceiver also puts a special invalid signal on the cable to ensure that all other transceivers also realize that a collision has occurred.

With 10Base5, a **transceiver cable** connects the transceiver to an interface board in the computer. The transceiver cable may be up to 50 meters long and contains five individually shielded twisted pairs. Two of the pairs are for data in and data out, respectively. Two more are for control signals in and out. The fifth pair, which is not always used, allows the computer to power the transceiver electronics. Some transceivers allow up to eight nearby computers to be attached to them, to reduce the number of transceivers needed.

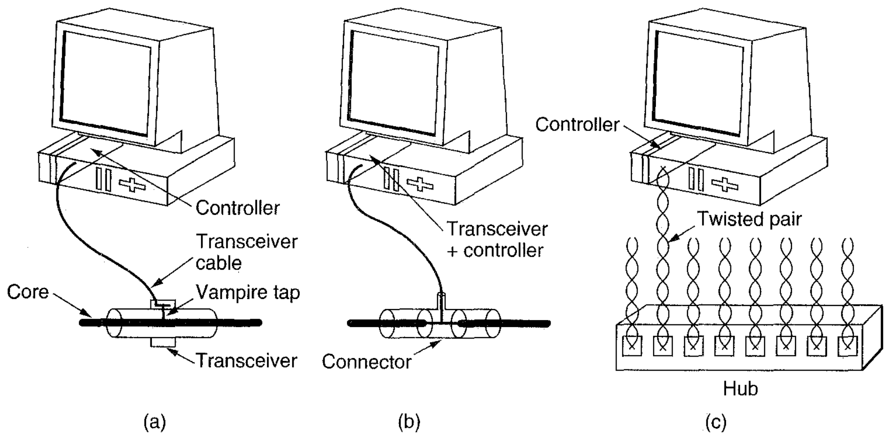

Fig. 4-18. Three kinds of 802.3 cabling. (a) 10Base5. (b) 10Base2. (c) 10Base-T.

The transceiver cable terminates on an interface board inside the computer. The interface board contains a controller chip that transmits frames to, and receives frames from, the transceiver. The controller is responsible for assembling the data into the proper frame format, as well as computing checksums on outgoing frames and verifying them on incoming frames. Some controller chips also manage a pool of buffers for incoming frames, a queue of buffers to be transmitted, DMA transfers with the host computers, and other aspects of network management.

With 10Base2, the connection to the cable is just a passive BNC T-junction connector. The transceiver electronics are on the controller board, and each station always has its own transceiver.

With 10Base-T, there is no cable at all, just the hub (a box full of electronics). Adding or removing a station is simpler in this configuration, and cable breaks can be detected easily. The disadvantage of 10Base-T is that the maximum cable run from the hub is only 100 meters, maybe 150 meters if high-quality (category 5) twisted pairs are used. Also, a large hub costs thousands of dollars. Still, 10Base-T is becoming steadily more popular due to the ease of maintenance that it offers. A faster version of 10Base-T (100Base-T) will be discussed later in this chapter.

A fourth cabling option for 802.3 is **10Base-F**, which uses fiber optics. This alternative is expensive due to the cost of the connectors and terminators, but it has excellent noise immunity and is the method of choice when running between buildings or widely separated hubs.

Figure 4-19 shows different ways of wiring up a building. In Fig. 4-19(a), a single cable is snaked from room to room, with each station tapping onto it at the nearest point. In Fig. 4-19(b), a vertical spine runs from the basement to the roof, with horizontal cables on each floor connected to it by special amplifiers (repeaters). In some buildings the horizontal cables are thin, and the backbone is thick. The most general topology is the tree, as in Fig. 4-19(c), because a network with two paths between some pairs of stations would suffer from interference between the two signals.

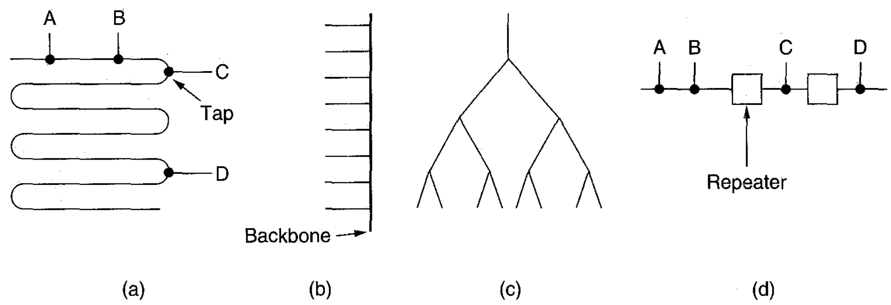

Fig. 4-19. Cable topologies. (a) Linear. (b) Spine. (c) Tree. (d) Segmented.

Each version of 802.3 has a maximum cable length per segment. To allow larger networks, multiple cables can be connected by **repeaters**, as shown in Fig. 4-19(d). A repeater is a physical layer device. It receives, amplifies, and retransmits signals in both directions. As far as the software is concerned, a series of cable segments connected by repeaters is no different than a single cable (except for some delay introduced by the repeaters). A system may contain multiple cable segments and multiple repeaters, but no two transceivers may be more than 2.5 km apart and no path between any two transceivers may traverse more than four repeaters.

Manchester Encoding

None of the versions of 802.3 use straight binary encoding with 0 volts for a 0 bit and 5 volts for a 1 bit because it leads to ambiguities. If one station sends the bit string 0001000, others might falsely interpret it as 10000000 or 01000000 because they cannot tell the difference between an idle sender (0 volts) and a 0 bit (0 volts).

What is needed is a way for receivers to unambiguously determine the start, end, or middle of each bit without reference to an external clock. Two such approaches are called **Manchester encoding** and **differential Manchester encoding**. With Manchester encoding, each bit period is divided into two equal intervals. A binary 1 bit is sent by having the voltage set high during the first interval and low in the second one. A binary 0 is just the reverse: first low and then high. This scheme ensures that every bit period has a transition in the middle, making it easy for the receiver to synchronize with the sender. A disadvantage of Manchester encoding is that it requires twice as much bandwidth as straight binary encoding, because the pulses are half the width. Manchester encoding is shown in Fig. 4-20(b).

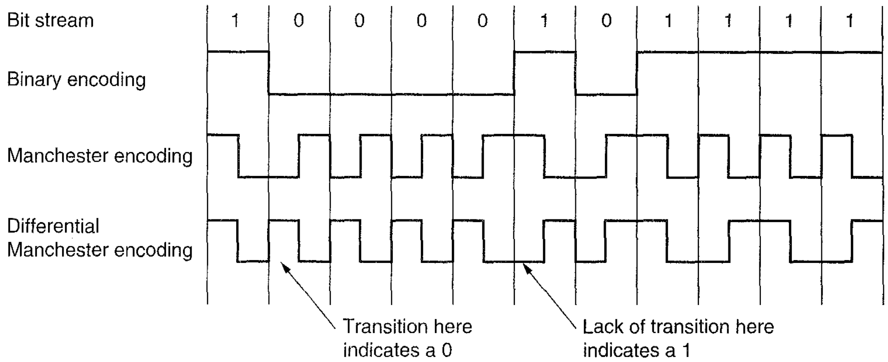

Fig. 4-20. (a) Binary encoding. (b) Manchester encoding. (c) Differential Manchester encoding.

Differential Manchester encoding, shown in Fig. 4-20(c), is a variation of basic Manchester encoding. In it, a 1 bit is indicated by the absence of a transition at the start of the interval. A 0 bit is indicated by the presence of a transition at the start of the interval. In both cases, there is a transition in the middle as well. The differential scheme requires more complex equipment but offers better noise immunity. All 802.3 baseband systems use Manchester encoding due to its simplicity. The high signal is +0.85 volts and the low signal is −0.85 volts, giving a DC value of 0 volts.

The 802.3 MAC Sublayer Protocol

The 802.3 (IEEE, 1985a) frame structure is shown in Fig. 4-21. Each frame starts with a *Preamble* of 7 bytes, each containing the bit pattern 10101010. The Manchester encoding of this pattern produces a 10-MHz square wave for 5.6 μsec to allow the receiver's clock to synchronize with the sender's. Next comes a *Start of frame* byte containing 10101011 to denote the start of the frame itself.

The frame contains two addresses, one for the destination and one for the source. The standard allows 2-byte and 6-byte addresses, but the parameters defined for the 10-Mbps baseband standard use only the 6-byte addresses. The high-order bit of the destination address is a 0 for ordinary addresses and 1 for group addresses. Group addresses allow multiple stations to listen to a single address. When a frame is sent to a group address, all the stations in the group receive it. Sending to a group of stations is called **multicast**. The address consisting of all 1 bits is reserved for **broadcast**. A frame containing all 1s in the destination field is delivered to all stations on the network.

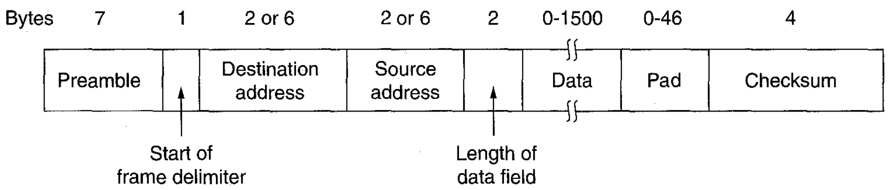

Fig. 4-21. The 802.3 frame format.

Another interesting feature of the addressing is the use of bit 46 (adjacent to the high-order bit) to distinguish local from global addresses. Local addresses are assigned by each network administrator and have no significance outside the local network. Global addresses, in contrast, are assigned by IEEE to ensure that no two stations anywhere in the world have the same global address. With $48 - 2 = 46$ bits available, there are about 7×10^{13} global addresses. The idea is that any station can uniquely address any other station by just giving the right 48-bit number. It is up to the network layer to figure out how to locate the destination.

The *Length* field tells how many bytes are present in the data field, from a minimum of 0 to a maximum of 1500. While a data field of 0 bytes is legal, it causes a problem. When a transceiver detects a collision, it truncates the current frame, which means that stray bits and pieces of frames appear on the cable all the time. To make it easier to distinguish valid frames from garbage, 802.3 states that valid frames must be at least 64 bytes long, from destination address to checksum. If the data portion of a frame is less than 46 bytes, the pad field is used to fill out the frame to the minimum size.

Another (and more important) reason for having a minimum length frame is to prevent a station from completing the transmission of a short frame before the first bit has even reached the far end of the cable, where it may collide with another frame. This problem is illustrated in Fig. 4-22. At time 0, station *A*, at one end of the network, sends off a frame. Let us call the propagation time for this frame to reach the other end τ. Just before the frame gets to the other end (i.e., at time $\tau - \varepsilon$) the most distant station, *B*, starts transmitting. When *B* detects that it is receiving more power than it is putting out, it knows that a collision has occurred, so it aborts its transmission and generates a 48-bit noise burst to warn all other stations. At about time 2τ, the sender sees the noise burst and aborts its transmission, too. It then waits a random time before trying again.

If a station tries to transmit a very short frame, it is conceivable that a collision occurs, but the transmission completes before the noise burst gets back at 2τ. The sender will then incorrectly conclude that the frame was successfully sent. To prevent this situation from occurring, all frames must take more than 2τ to send. For a 10-Mbps LAN with a maximum length of 2500 meters and four

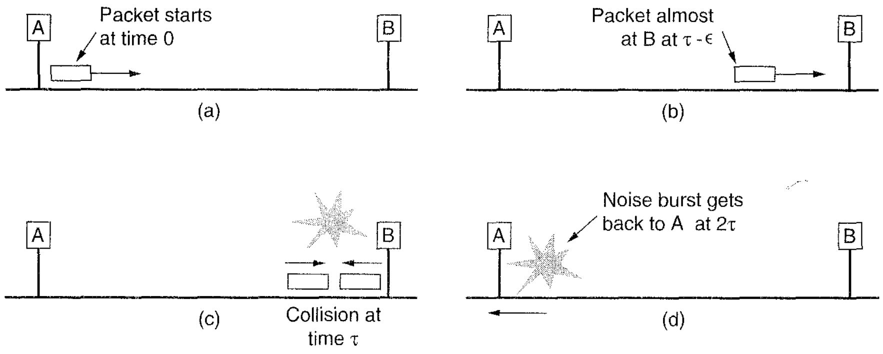

Fig. 4-22. Collision detection can take as long as 2τ.

repeaters (from the 802.3 specification), the minimum allowed frame must take 51.2 μsec. This time corresponds to 64 bytes. Frames with fewer bytes are padded out to 64 bytes.

As the network speed goes up, the minimum frame length must go up or the maximum cable length must come down, proportionally. For a 2500-meter LAN operating at 1 Gbps, the minimum frame size would have to be 6400 bytes. Alternatively, the minimum frame size could be 640 bytes and the maximum distance between any two stations 250 meters. These restrictions are becoming increasingly painful as we move toward gigabit networks.

The final 802.3 field is the *Checksum*. It is effectively a 32-bit hash code of the data. If some data bits are erroneously received (due to noise on the cable), the checksum will almost certainly be wrong, and the error will be detected. The checksum algorithm is a cyclic redundancy check of the kind discussed in Chap. 3.

The Binary Exponential Backoff Algorithm

Let us now see how randomization is done when a collision occurs. The model is that of Fig. 4-5. After a collision, time is divided up into discrete slots whose length is equal to the worst case round-trip propagation time on the ether (2τ). To accommodate the longest path allowed by 802.3 (2.5 km and four repeaters), the slot time has been set to 512 bit times, or 51.2 μsec.

After the first collision, each station waits either 0 or 1 slot times before trying again. If two stations collide and each one picks the same random number, they will collide again. After the second collision, each one picks either 0, 1, 2, or 3 at random and waits that number of slot times. If a third collision occurs (the probability of this happening is 0.25), then the next time the number of slots to wait is chosen at random from the interval 0 to $2^3 - 1$.

In general, after i collisions, a random number between 0 and $2^i - 1$ is chosen, and that number of slots is skipped. However, after ten collisions have been reached, the randomization interval is frozen at a maximum of 1023 slots. After 16 collisions, the controller throws in the towel and reports failure back to the computer. Further recovery is up to higher layers.

This algorithm, called **binary exponential backoff**, was chosen to dynamically adapt to the number of stations trying to send. If the randomization interval for all collisions was 1023, the chance of two stations colliding for a second time would be negligible, but the average wait after a collision would be hundreds of slot times, introducing significant delay. On the other hand, if each station always delayed for either zero or one slots, then if 100 stations ever tried to send at once, they would collide over and over until 99 of them picked 0 and the remaining station picked 1, or vice versa. This might take years. By having the randomization interval grow exponentially as more and more consecutive collisions occur, the algorithm ensures a low delay when only a few stations collide but also ensures that the collision is resolved in a reasonable interval when many stations collide.

As described so far, CSMA/CD provides no acknowledgements. Since the mere absence of collisions does not guarantee that bits were not garbled by noise spikes on the cable, for reliable communication the destination must verify the checksum, and if correct, send back an acknowledgement frame to the source. Normally, this acknowledgement would be just another frame as far as the protocol is concerned and would have to fight for channel time just like a data frame. However, a simple modification to the contention algorithm would allow speedy confirmation of frame receipt (Tokoro and Tamaru, 1977). All that would be needed is to reserve the first contention slot following successful transmission for the destination station.

802.3 Performance

Now let us briefly examine the performance of 802.3 under conditions of heavy and constant load, that is, k stations always ready to transmit. A rigorous analysis of the binary exponential backoff algorithm is complicated. Instead we will follow Metcalfe and Boggs (1976) and assume a constant retransmission probability in each slot. If each station transmits during a contention slot with probability p, the probability A that some station acquires the channel in that slot is

$$A = kp(1-p)^{k-1} \tag{4-6}$$

A is maximized when $p = 1/k$, with $A \to 1/e$ as $k \to \infty$. The probability that the contention interval has exactly j slots in it is $A(1-A)^{j-1}$, so the mean number of slots per contention is given by

$$\sum_{j=0}^{\infty} jA(1-A)^{j-1} = \frac{1}{A}$$

Since each slot has a duration 2τ, the mean contention interval, w, is $2\tau/A$.

Assuming optimal p, the mean number of contention slots is never more than e, so w is at most $2\tau e \approx 5.4\tau$.

If the mean frame takes P sec to transmit, when many stations have frames to send,

$$\text{Channel efficiency} = \frac{P}{P + 2\tau/A} \tag{4-7}$$

Here we see where the maximum cable distance between any two stations enters into the performance figures, giving rise to topologies other than that of Fig. 4-19(a). The longer the cable, the longer the contention interval. By allowing no more than 2.5 km of cable and four repeaters between any two transceivers, the round-trip time can be bounded to 51.2 μsec, which at 10 Mbps corresponds to 512 bits or 64 bytes, the minimum frame size.

It is instructive to formulate Eq. (4-7) in terms of the frame length, F, the network bandwidth, B, the cable length, L, and the speed of signal propagation, c, for the optimal case of e contention slots per frame. With $P = F/B$, Eq. (4-7) becomes

$$\text{Channel efficiency} = \frac{1}{1 + 2BLe/cF} \tag{4-8}$$

When the second term in the denominator is large, network efficiency will be low. More specifically, increasing network bandwidth or distance (the BL product) reduces efficiency for a given frame size. Unfortunately, much research on network hardware is aimed precisely at increasing this product. People want high bandwidth over long distances (fiber optic MANs, for example), which suggests that 802.3 may not be the best system for these applications.

In Fig. 4-23, the channel efficiency is plotted versus number of ready stations for $2\tau = 51.2$ μsec and a data rate of 10 Mbps using Eq. (4-8). With a 64-byte slot time, it is not surprising that 64-byte frames are not efficient. On the other hand, with 1024-byte frames and an asymptotic value of e 64-byte slots per contention interval, the contention period is 174 bytes long and the efficiency is 0.85.

To determine the mean number of stations ready to transmit under conditions of high load, we can use the following (crude) observation. Each frame ties up the channel for one contention period and one frame transmission time, for a total of $P + w$ sec. The number of frames per second is therefore $1/(P + w)$. If each station generates frames at a mean rate of λ frames/sec, when the system is in state k the total input rate of all unblocked stations combined is $k\lambda$ frames/sec. Since in equilibrium the input and output rates must be identical, we can equate these two expressions and solve for k. (Notice that w is a function of k.) A more sophisticated analysis is given in (Bertsekas and Gallager, 1992).

It is probably worth mentioning that there has been a large amount of theoretical performance analysis of 802.3 (and other networks). Virtually all of this work has assumed that traffic is Poisson. As researchers have begun looking at real

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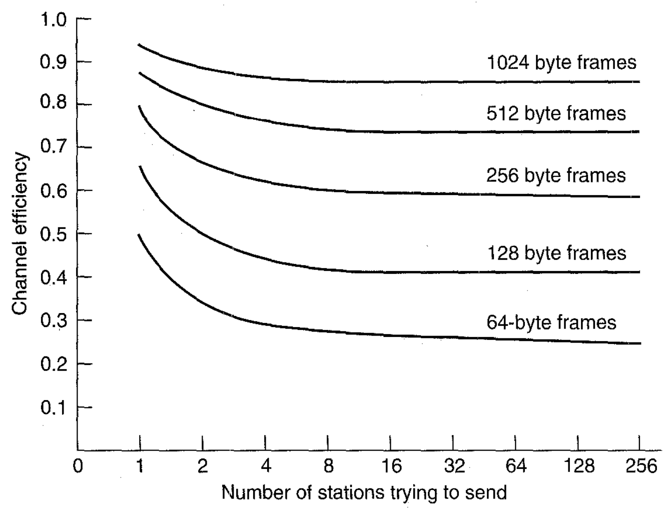

Fig. 4-23. Efficiency of 802.3 at 10 Mbps with 512-bit slot times.

data, it now appears that network traffic is rarely Poisson, but self-similar (Paxson and Floyd, 1994; and Willinger et al., 1995). What this means is that averaging over long periods of time does not smooth out the traffic. The average number of packets in each minute of an hour has as much variance as the average number of packets in each second of a minute. The consequence of this discovery is that most models of network traffic do not apply to the real world and should be taken with a grain (or better yet, a metric ton) of salt.

Switched 802.3 LANs

As more and more stations are added to an 802.3 LAN, the traffic will go up. Eventually, the LAN will saturate. One way out is to go to a higher speed, say from 10 Mbps to 100 Mbps. This solution requires throwing out all the 10 Mbps adaptor cards and buying new ones, which is expensive. If the 802.3 chips are on the computers' main circuit boards, it may not even be possible to replace them.

Fortunately, a different, less drastic solution is possible: a switched 802.3 LAN, as shown in Fig. 4-24. The heart of this system is a switch containing a high-speed backplane and room for typically 4 to 32 plug-in line cards, each containing one to eight connectors. Most often, each connector has a 10Base-T twisted pair connection to a single host computer.

When a station wants to transmit an 802.3 frame, it outputs a standard frame to the switch. The plug-in card getting the frame checks to see if it is destined for one of the other stations connected to the same card. If so, the frame is copied

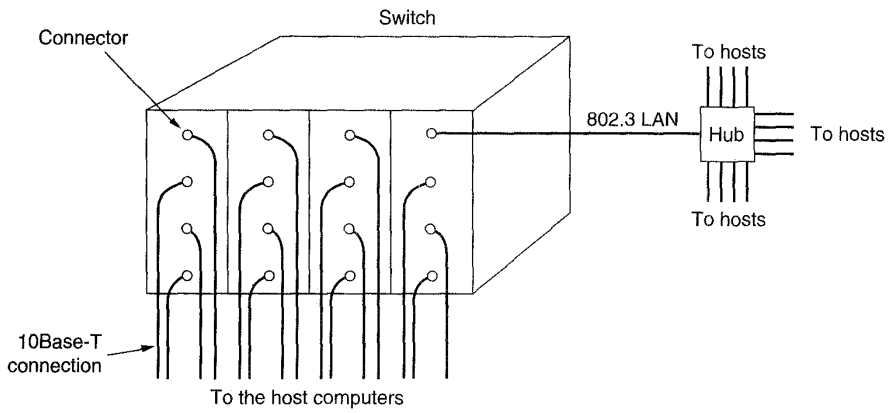

Fig. 4-24. A switched 802.3 LAN.

there. If not, the frame is sent over the high-speed backplane to the destination station's card. The backplane typically runs at over 1 Gbps using a proprietary protocol.

What happens if two machines attached to the same plug-in card transmit frames at the same time? It depends on how the card has been constructed. One possibility is for all the ports on the card to be wired together to form a local on-card LAN. Collisions on this on-card LAN will be detected and handled the same as any other collisions on a CSMA/CD network—with retransmissions using the binary backoff algorithm. With this kind of plug-in card, only one transmission per card is possible at any instant, but all the cards can be transmitting in parallel. With this design, each card forms its own **collision domain**, independent of the others.

With the other kind of plug-in card, each input port is buffered, so incoming frames are stored in the card's on-board RAM as they arrive. This design allows all input ports to receive (and transmit) frames at the same time, for parallel, full-duplex operation. Once a frame has been completely received, the card can then check to see if the frame is destined for another port on the same card, or for a distant port. In the former case it can be transmitted directly to the destination. In the latter case, it must be transmitted over the backplane to the proper card. With this design, each port is a separate collision domain, so collisions do not occur. The total system throughput can often be increased by an order of magnitude over 10Base-5, which has a single collision domain for the entire system.

Since the switch just expects standard 802.3 frames on each input port, it is possible use some of the ports as concentrators. In Fig. 4-24, the port in the upper right-hand corner is connected not to a single station, but to a 12-port hub. As frames arrive at the hub, they contend for the 802.3 LAN in the usual way, including collisions and binary backoff. Successful frames make it to the switch, and are treated there like any other incoming frames: they are switched to the correct output line over the high-speed backplane. If all the input ports are connected to hubs, rather than to individual stations, the switch just becomes an 802.3 to 802.3 bridge. We will study bridges later in this chapter.

4.3.2. IEEE Standard 802.4: Token Bus

Although 802.3 is widely used in offices, during the development of the 802 standard, people from General Motors and other companies interested in factory automation had serious reservations about it. For one thing, due to the probabilistic MAC protocol, with a little bad luck a station might have to wait arbitrarily long to send a frame (i.e., the worst case is unbounded). For another, 802.3 frames do not have priorities, making them unsuited for real-time systems in which important frames should not be held up waiting for unimportant frames.

A simple system with a known worst case is a ring in which the stations take turns sending frames. If there are n stations and it takes T sec to send a frame, no frame will ever have to wait more than nT sec to be sent. The factory automation people in the 802 committee liked the conceptual idea of a ring but did not like the physical implementation because a break in the ring cable would bring the whole network down. Furthermore, they noted that a ring is a poor fit to the linear topology of most assembly lines. As a result, a new standard was developed, having the robustness of the 802.3 broadcast cable, but the known worst-case behavior of a ring.

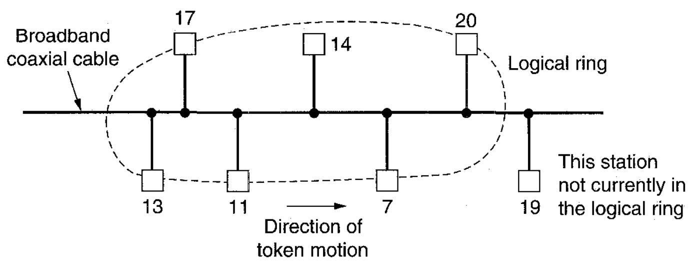

Fig. 4-25. A token bus.

This standard, 802.4 (Dirvin and Miller, 1986; and IEEE, 1985b), describes a LAN called a **token bus**. Physically, the token bus is a linear or tree-shaped cable onto which the stations are attached. Logically, the stations are organized into a ring (see Fig. 4-25), with each station knowing the address of the station to its "left" and "right." When the logical ring is initialized, the highest numbered station may send the first frame. After it is done, it passes permission to its immediate neighbor by sending the neighbor a special control frame called a **token**. The

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token propagates around the logical ring, with only the token holder being permitted to transmit frames. Since only one station at a time holds the token, collisions do not occur.

An important point to realize is that the physical order in which the stations are connected to the cable is not important. Since the cable is inherently a broadcast medium, each station receives each frame, discarding those not addressed to it. When a station passes the token, it sends a token frame specifically addressed to its logical neighbor in the ring, irrespective of where that station is physically located on the cable. It is also worth noting that when stations are first powered on, they will not be in the ring (e.g., stations 14 and 19 in Fig. 4-25), so the MAC protocol has provisions for adding stations to, and deleting stations from, the ring.

The 802.4 MAC protocol is very complex, with each station having to maintain ten different timers and more than two dozen internal state variables. The 802.4 standard is much longer than 802.3, filling more than 200 pages. The two standards are also quite different in style, with 802.3 giving the protocols as Pascal procedures, whereas 802.4 gives them as finite state machines, with the actions written in Ada®.

For the physical layer, the token bus uses the 75-ohm broadband coaxial cable used for cable television. Both single- and dual-cable systems are allowed, with or without head-ends. Three different analog modulation schemes are permitted: phase continuous frequency shift keying, phase coherent frequency shift keying, and multilevel duobinary amplitude modulated phase shift keying. Speeds of 1, 5, and 10 Mbps are possible. Furthermore, the modulation schemes not only provide ways to represent 0, 1, and idle on the cable, but also three other symbols used for network control. All in all, the physical layer is totally incompatible with 802.3, and a lot more complicated.

The Token Bus MAC Sublayer Protocol

When the ring is initialized, stations are inserted into it in order of station address, from highest to lowest. Token passing is also done from high to low addresses. Each time a station acquires the token, it can transmit frames for a certain amount of time; then it must pass the token on. If the frames are short enough, several consecutive frames may be sent. If a station has no data, it passes the token immediately upon receiving it.

The token bus defines four priority classes, 0, 2, 4, and 6 for traffic, with 0 the lowest and 6 the highest. It is easiest to think of each station internally being divided into four substations, one at each priority level. As input comes in to the MAC sublayer from above, the data are checked for priority and routed to one of the four substations. Thus each substation maintains its own queue of frames to be transmitted.

When the token comes into the station over the cable, it is passed internally to the priority 6 substation, which may begin transmitting frames, if it has any.

When it is done (or when its timer expires), the token is passed internally to the priority 4 substation, which may then transmit frames until its timer expires, at which point the token is passed internally to the priority 2 substation. This process is repeated until either the priority 0 substation has sent all its frames or its timer has expired. Either way, at this point the token is sent to the next station in the ring.

Without getting into all the details of how the various timers are managed, it should be clear that by setting the timers properly, we can ensure that a guaranteed fraction of the total token-holding time can be allocated to priority 6 traffic. The lower priorities will have to live with what is left over. If the higher priority substations do not need all of their allocated time, the lower priority substations can have the unused portion, so it is not wasted.

This priority scheme, which guarantees priority 6 traffic a known fraction of the network bandwidth, can be used to implement real-time traffic. For example, suppose the parameters of a 50-station network running at 10 Mbps have been adjusted to give priority 6 traffic 1/3 of the bandwidth. Then each station has a guaranteed 67 kbps for priority 6 traffic. This bandwidth could be used to synchronize robots on an assembly line or carry one digital voice channel per station, with a little left over for control information.

The token bus frame format is shown in Fig. 4-26. It is unfortunately different from the 802.3 frame format. The preamble is used to synchronize the receiver's clock, as in 802.3, except that here it may be as short as 1 byte. The *Starting delimiter* and *Ending delimiter* fields are used to mark the frame boundaries. Both of these fields contain analog encoding of symbols other than 0s and 1s, so that they cannot occur accidentally in the user data. As a result, no length field is needed.

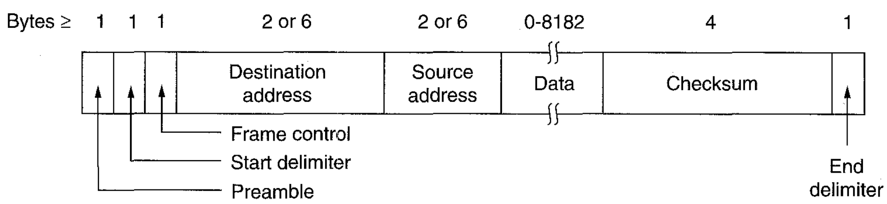

Fig. 4-26. The 802.4 frame format.

The *Frame control* field is used to distinguish data frames from control frames. For data frames, it carries the frame's priority. It can also carry an indicator requiring the destination station to acknowledge correct or incorrect receipt of the frame. Without this indicator, the destination would not be allowed to send anything because it does not have the token. This indicator turns the token bus into something resembling the acknowledgement scheme of Tokoro and Tamaru.

For control frames, the *Frame control* field is used to specify the frame type.

The allowed types include token passing and various ring maintenance frames, including the mechanism for letting new stations enter the ring, the mechanism for allowing stations to leave the ring, and so on. Note that the 802.3 protocol does not have any control frames. All the MAC layer does there is provide a way to get frames onto the cable; it does not care what is in them.

The *Destination address* and *Source address* fields are the same as in 802.3 (yes, the two groups did talk to each other; no, they did not agree on very much). As in 802.3, a given network must use all 2-byte addresses or all 6-byte addresses, not a mixture on the same cable. The initial 802.4 standard allows either size. The individual and group addressing and the local and global address assignments are identical to 802.3.

The *Data* field may be up to 8182 bytes long when 2-byte addresses are used, and up to 8174 bytes long when 6-byte addresses are used. This is more than five times as long as the maximum 802.3 frame, which was made short to prevent one station from hogging the channel too long. With the token bus, the timers can be used as an antihogging measure, but it is nice to be able to send long frames when real-time traffic is not an issue. The *Checksum* is used to detect transmission errors. It uses the same algorithm and polynomial as 802.3.

The token bus control frames are shown in Fig. 4-27. They will be discussed below. The only one we have seen so far is the *token* frame, used to pass the token from station to station. Most of the rest relate to adding and deleting stations from the logical ring.

Frame control field	Name	Meaning
00000000	Claim_token	Claim token during ring initialization
00000001	Solicit_successor_1	Allow stations to enter the ring
00000010	Solicit_successor_2	Allow stations to enter the ring
00000011	Who_follows	Recover from lost token
00000100	Resolve_contention	Used when multiple stations want to enter
00001000	Token	Pass the token
00001100	Set_successor	Allow station to leave the ring

Fig. 4-27. The token bus control frames.

Logical Ring Maintenance

From time to time, stations are powered on and want to join the ring. Other are turned off and want to leave. The MAC sublayer protocol provides a detailed specification of exactly how this is done while maintaining the known worst case bound on token rotation. Below we will just briefly sketch the mechanisms used.

Once the ring has been established, each station's interface maintains the addresses of the predecessor and successor stations internally. Periodically, the token holder sends one of the SOLICIT_SUCCESSOR frames shown in Fig. 4-27 to solicit bids from stations that wish to join the ring. The frame gives the sender's address and the successor's address. Stations inside that range may bid to enter (to keep the ring sorted in descending order of station address).

If no station bids to enter within a slot time (2τ, as in 802.3), the **response window** is closed and the token holder continues with its normal business. If exactly one station bids to enter, it is inserted into the ring and becomes the token holder's successor.

If two or more stations bid to enter, their frames will collide and be garbled, as in 802.3. The token holder then runs an arbitration algorithm, starting with the broadcast of a RESOLVE_CONTENTION frame. The algorithm is a variation of binary countdown, using two bits at a time.

Furthermore, all station interfaces maintain two random bits inside. These bits are used to delay all bids by 0, 1, 2, or 3 slot times, to further reduce contention. In other words, two stations only collide on a bid if the current two address bits being used are the same and they happen to have the same two random bits. To prevent stations that must wait 3 slot times from being at a permanent disadvantage, the random bits are regenerated every time they are used or periodically every 50 msec.

The solicitation of new stations may not interfere with the guaranteed worst case for token rotation. Each station has a timer that is reset whenever it acquires the token. When the token comes in, the old value of this timer (i.e., the previous token rotation time) is inspected just before the timer is reset. If it exceeds a certain threshold value, there has been too much traffic recently, so no bids may be solicited this time around. In any event, only one station may enter at each solicitation, to put a bound on how much time can be consumed in ring maintenance. No guarantee is provided for how long a station may have to wait to join the ring when traffic is heavy, but in practice it should not be more than a few seconds. This uncertainty is unfortunate, making 802.4 less suitable for real-time systems than its supporters often claim.

Leaving the ring is easy. A station, X, with successor S, and predecessor P, leaves the ring, by sending P a SET_SUCCESSOR frame telling it that henceforth its successor is S instead of X. Then X just stops transmitting.

Ring initialization is a special case of adding new stations. Consider an idle system with all stations powered off. When the first station comes on-line, it notices that there is no traffic for a certain period. Then it sends a CLAIM_TOKEN frame. Not hearing any competitors contending for the token, it creates a token and sets up a ring containing only itself. Periodically, it solicits bids for new stations to join. As new stations are powered on, they will respond to these bids and join the ring using the contention algorithm described above. Eventually, every station that wants to join the ring will be able to do so. If the first two stations are powered on simultaneously, the protocol deals with this by letting them bid for the token using the standard modified binary countdown algorithm and the two random bits.

Due to transmission errors or hardware failures, problems can arise with the logical ring or the token. For example, if a station tries to pass the token to a station that has gone down, what happens? The solution is straightforward. After passing the token, a station listens to see if its successor either transmits a frame or passes the token. If it does neither, the token is passed a second time.

If that also fails, the station transmits a WHO_FOLLOWS frame specifying the address of its successor. When the failed station's successor sees a WHO_FOLLOWS frame naming its predecessor, it responds by sending a SET_SUCCESSOR frame to the station whose successor failed, naming itself as the new successor. In this way, the failed station is removed from the ring.

Now suppose that a station fails to pass the token to its successor and also fails to locate the successor's successor, which may also be down. It adopts a new strategy by sending a SOLICIT_SUCCESSOR_2 frame to see if *anyone* else is still alive. Once again the standard contention protocol is run, with all stations that want to be in the ring now bidding for a place. Eventually, the ring is reestablished.

Another kind of problem occurs if the token holder goes down and takes the token with it. This problem is solved using the ring initialization algorithm. Each station has a timer that is reset whenever a frame appears on the network. When this timer hits a threshold value, the station issues a CLAIM_TOKEN frame, and the modified binary countdown algorithm with random bits determines who gets the token.

Still another problem is multiple tokens. If a station holding the token notices a transmission from another station, it discards its token. If there were two, there would now be one. If there were more than two, this process would be repeated sooner or later until all but one were discarded. If, by accident, all the tokens are discarded, then the lack of activity will cause one or more stations to try to claim the token.

4.3.3. IEEE Standard 802.5: Token Ring

Ring networks have been around for many years (Pierce, 1972) and have long been used for both local and wide area networks. Among their many attractive features is the fact that a ring is not really a broadcast medium, but a collection of individual point-to-point links that happen to form a circle. Point-to-point links involve a well-understood and field-proven technology and can run on twisted pair, coaxial cable, or fiber optics. Ring engineering is also almost entirely digital, whereas 802.3, for example, has a substantial analog component for collision detection. A ring is also fair and has a known upper bound on channel access.

For these reasons, IBM chose the ring as its LAN and IEEE has included the **token ring** standard as 802.5 (IEEE, 1985c; Latif et al., 1992).

A major issue in the design and analysis of any ring network is the "physical length" of a bit. If the data rate of the ring is R Mbps, a bit is emitted every $1/R$ μsec. With a typical signal propagation speed of about 200 m/μsec, each bit occupies $200/R$ meters on the ring. This means, for example, that a 1-Mbps ring whose circumference is 1000 meters can contain only 5 bits on it at once. The implications of the number of bits on the ring will become clearer later.

As mentioned above, a ring really consists of a collection of ring interfaces connected by point-to-point lines. Each bit arriving at an interface is copied into a 1-bit buffer and then copied out onto the ring again. While in the buffer, the bit can be inspected and possibly modified before being written out. This copying step introduces a 1-bit delay at each interface. A ring and its interfaces are shown in Fig. 4-28.

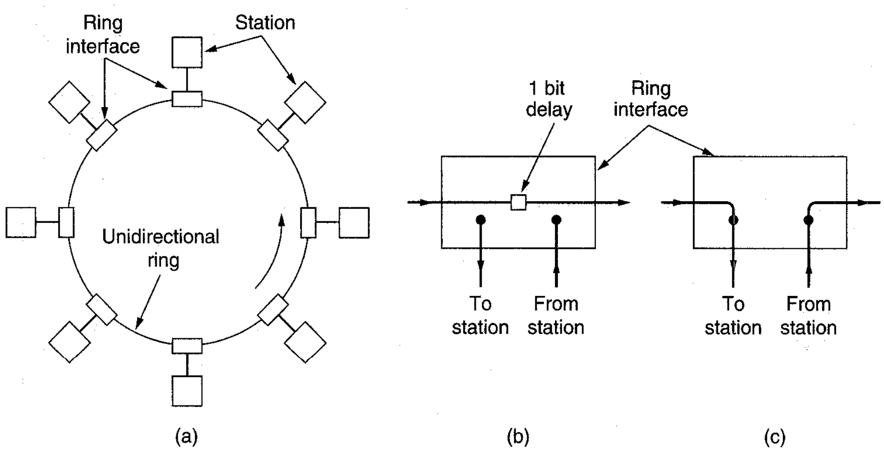

Fig. 4-28. (a) A ring network. (b) Listen mode. (c) Transmit mode.

In a token ring a special bit pattern, called the **token**, circulates around the ring whenever all stations are idle. When a station wants to transmit a frame, it is required to seize the token and remove it from the ring before transmitting. This action is done by inverting a single bit in the 3-byte token, which instantly changes it into the first 3 bytes of a normal data frame. Because there is only one token, only one station can transmit at a given instant, thus solving the channel access problem the same way the token bus solves it.

An implication of the token ring design is that the ring itself must have a sufficient delay to contain a complete token to circulate when all stations are idle. The delay has two components: the 1-bit delay introduced by each station, and the signal propagation delay. In almost all rings, the designers must assume that stations may be powered down at various times, especially at night. If the interfaces are powered from the ring, shutting down the station has no effect on the interface, but if the interfaces are powered externally, they must be designed to connect the input to the output when power goes down, thus removing the 1-bit delay. The point here is that on a short ring an artificial delay may have to be inserted into the ring at night to ensure that a token can be contained on it.

Ring interfaces have two operating modes, listen and transmit. In listen mode, the input bits are simply copied to output, with a delay of 1 bit time, as shown in Fig. 4-28(b). In transmit mode, which is entered only after the token has been seized, the interface breaks the connection between input and output, entering its own data onto the ring. To be able to switch from listen to transmit mode in 1 bit time, the interface usually needs to buffer one or more frames itself rather than having to fetch them from the station on such short notice.

As bits that have propagated around the ring come back, they are removed from the ring by the sender. The sending station can either save them, to compare with the original data to monitor ring reliability, or discard them. Because the entire frame never appears on the ring at one instant, this ring architecture puts no limit on the size of the frames. After a station has finished transmitting the last bit of its last frame, it must regenerate the token. When the last bit of the frame has gone around and come back, it must be removed, and the interface must switch back into listen mode immediately, to avoid removing the token that might follow if no other station has removed it.

It is straightforward to handle acknowledgements on a token ring. The frame format need only include a 1-bit field for acknowledgements, initially zero. When the destination station has received a frame, it sets the bit. Of course, if the acknowledgement means that the checksum has been verified, the bit must follow the checksum, and the ring interface must be able to verify the checksum as soon as its last bit has arrived. When a frame is broadcast to multiple stations, a more complicated acknowledgement mechanism must be used (if any is used at all).

When traffic is light, the token will spend most of its time idly circulating around the ring. Occasionally a station will seize it, transmit a frame, and then output a new token. However, when the traffic is heavy, so that there is a queue at each station, as soon as a station finishes its transmission and regenerates the token, the next station downstream will see and remove the token. In this manner the permission to send rotates smoothly around the ring, in round-robin fashion. The network efficiency can begin to approach 100 percent under conditions of heavy load.

Now let us turn from token rings in general to the 802.5 standard in particular. At the physical layer, 802.5 calls for shielded twisted pairs running at 1 or 4 Mbps, although IBM later introduced a 16-Mbps version. Signals are encoded using differential Manchester encoding [see Fig. 4-20(c)] with high and low being positive and negative signals of absolute magnitude 3.0 to 4.5 volts. Normally, differential Manchester encoding uses high-low or low-high for each bit, but

802.5 also uses high-high and low-low in certain control bytes (e.g., to mark the start and end of a frame). These nondata signals always occur in consecutive pairs so as not to introduce a DC component into the ring voltage.

One problem with a ring network is that if the cable breaks somewhere, the ring dies. This problem can be solved very elegantly by the use of a **wire center**, as shown in Fig. 4-29. While logically still a ring, physically each station is connected to the wire center by a cable containing (at least) two twisted pairs, one for data to the station and one for data from the station.

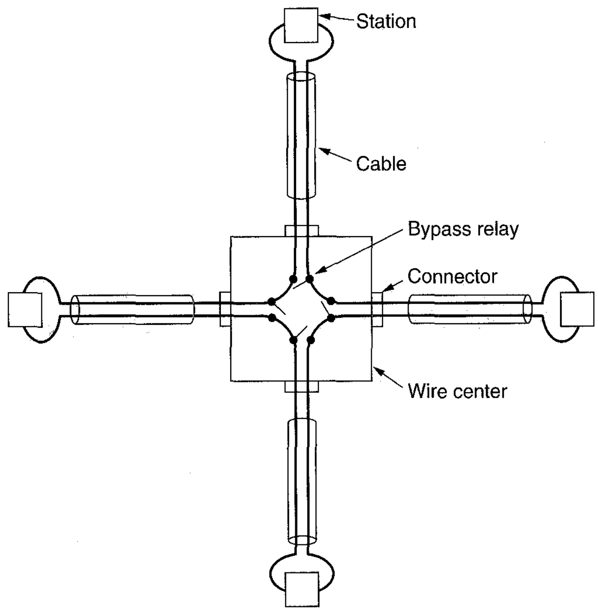

Fig. 4-29. Four stations connected via a wire center.

Inside the wire center are bypass relays that are energized by current from the stations. If the ring breaks or a station goes down, loss of the drive current will release the relay and bypass the station. The relays can also be operated by software to permit diagnostic programs to remove stations one at a time to find faulty stations and ring segments. The ring can then continue operation with the bad segment bypassed. Although the 802.5 standard does not formally require this kind of ring, often called a **star-shaped ring** (Saltzer et al., 1983), most 802.5 LANs, in fact, do use wire centers to improve their reliability and maintainability.

When a network consists of many clusters of stations far apart, a topology with multiple wire centers can be used. Just imagine that the cable to one of the stations in Fig. 4-29 were replaced by a cable to a distant wire center. Although logically all the stations are on the same ring, the wiring requirements are greatly reduced. An 802.5 ring using a wire center has a similar topology to an 802.3 10Base-T hub-based network, but the formats and protocols are different.

The Token Ring MAC Sublayer Protocol

The basic operation of the MAC protocol is straightforward. When there is no traffic on the ring, a 3-byte token circulates endlessly, waiting for a station to seize it by setting a specific 0 bit to a 1 bit, thus converting the token into the start-of-frame sequence. The station then outputs the rest of a normal data frame, as shown in Fig. 4-30.

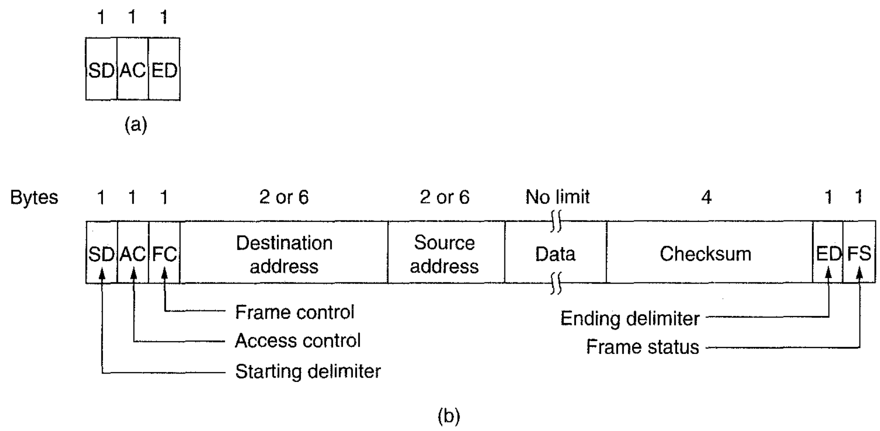

Fig. 4-30. (a) Token format. (b) Data frame format.

Under normal conditions, the first bit of the frame will go around the ring and return to the sender before the full frame has been transmitted. Only a very long ring will be able to hold even a short frame. Consequently, the transmitting station must drain the ring while it continues to transmit. As shown in Fig. 4-28(c), this means that the bits that have completed the trip around the ring come back to the sender and are removed.

A station may hold the token for the **token-holding time**, which is 10 msec unless an installation sets a different value. If there is enough time left after the first frame has been transmitted to send more frames, these may be sent as well. After all pending frames have been transmitted or the transmission of another frame would exceed the token-holding time, the station regenerates the 3-byte token frame and puts it out onto the ring.

The *Starting delimiter* and *Ending delimiter* fields of Fig. 4-30(b) mark the beginning and ending of the frame. Each contains invalid differential Manchester patterns (HH and LL) to distinguish them from data bytes. The *Access control* byte contains the token bit, and also the *Monitor bit*, *Priority bits*, and *Reservation bits* (described below). The *Frame control* byte distinguishes data frames from various possible control frames.

Next come the *Destination address* and *Source address* fields, which are the same as in 802.3 and 802.4. These are followed by the data, which may be as long as necessary, provided that the frame can still be transmitted within the token-holding time. The *Checksum* field, like the destination and source addresses, is also the same as 802.3 and 802.4.

An interesting byte not present in the other two protocols is the *Frame status* byte. It contains the *A* and *C* bits. When a frame arrives at the interface of a station with the destination address, the interface turns on the *A* bit as it passes through. If the interface copies the frame to the station, it also turns on the *C* bit. A station might fail to copy a frame due to lack of buffer space or other reasons.

When the sending station drains the frame from the ring, it examines the *A* and *C* bits. Three combinations are possible:

1. $A = 0$ and $C = 0$: destination not present or not powered up.
2. $A = 1$ and $C = 0$: destination present but frame not accepted.
3. $A = 1$ and $C = 1$: destination present and frame copied.

This arrangement provides an automatic acknowledgement for each frame. If a frame is rejected but the station is present, the sender has the option of trying again in a little while. The *A* and *C* bits are present twice in the *Frame status* to increase reliability inasmuch as they are not covered by the checksum.

The *Ending delimiter* contains an *E* bit which is set if any interface detects an error (e.g., a non-Manchester pattern where that is not permitted). It also contains a bit that can be used to mark the last frame in a logical sequence, sort of like an end-of-file bit.

The 802.5 protocol has an elaborate scheme for handling multiple priority frames. The 3-byte token frame contains a field in the middle byte giving the priority of the token. When a station wants to transmit a priority n frame, it must wait until it can capture a token whose priority is less than or equal to n. Furthermore, when a data frame goes by, a station can try to reserve the next token by writing the priority of the frame it wants to send into the frame's *Reservation bits*. However, if a higher priority has already been reserved there, the station may not make a reservation. When the current frame is finished, the next token is generated at the priority that has been reserved.

A little thought will show that this mechanism acts like a ratchet, always jacking the reservation priority higher and higher. To eliminate this problem, the protocol contains some complex rules. The essence of the idea is that the station raising the priority is responsible for lowering the priority again when it is done.

Notice that this priority scheme is substantially different from the token bus scheme, in which each station always gets its fair share of the bandwidth, no matter what other stations are doing. In the token ring, a station with only low priority frames may starve to death waiting for a low priority token to appear. Clearly, the two committees had different taste when trading off good service for high priority traffic versus fairness to all stations.

Ring Maintenance

The token bus protocol goes to considerable lengths to do ring maintenance in a fully decentralized way. The token ring protocol handles maintenance quite differently. Each token ring has a **monitor station** that oversees the ring. If the monitor goes down, a contention protocol ensures that another station is quickly elected as monitor. (Every station has the capability of becoming the monitor.) While the monitor is functioning properly, it alone is responsible for seeing that the ring operates correctly.

When the ring comes up or any station notices that there is no monitor, it can transmit a CLAIM TOKEN control frame. If this frame circumnavigates the ring before any other CLAIM TOKEN frames are sent, the sender becomes the new monitor (each station has monitor capability built in). The token ring control frames are shown in Fig. 4-31.

Control field	Name	Meaning
00000000	Duplicate address test	Test if two stations have the same address
00000010	Beacon	Used to locate breaks in the ring
00000011	Claim token	Attempt to become monitor
00000100	Purge	Reinitialize the ring
00000101	Active monitor present	Issued periodically by the monitor
00000110	Standby monitor present	Announces the presence of potential monitors

Fig. 4-31. Token ring control frames.

Among the monitor's responsibilities are seeing that the token is not lost, taking action when the ring breaks, cleaning the ring up when garbled frames appear, and watching out for orphan frames. An orphan frame occurs when a station transmits a short frame in its entirety onto a long ring and then crashes or is powered down before the frame can be drained. If nothing is done, the frame will circulate forever.

To check for lost tokens, the monitor has a timer that is set to the longest possible tokenless interval: each station transmitting for the full token-holding time. If this timer goes off, the monitor drains the ring and issues a new token.

When a garbled frame appears, the monitor can detect it by its invalid format or checksum, open the ring to drain it, and issue a new token when the ring has been cleaned up. Finally, the monitor detects orphan frames by setting the *monitor* bit in the *Access control* byte whenever it passes through. If an incoming frame has this bit set, something is wrong since the same frame has passed the monitor twice without having been drained, so the monitor drains it.

One last monitor function concerns the length of the ring. The token is 24 bits long, which means that the ring must be big enough to hold 24 bits. If the 1-bit delays in the stations plus the cable length add up to less than 24 bits, the monitor inserts extra delay bits so that a token can circulate.

One maintenance function that cannot be handled by the monitor is locating breaks in the ring. When a station notices that either of its neighbors appears to be dead, it transmits a BEACON frame giving the address of the presumably dead station. When the beacon has propagated around as far as it can, it is then possible to see how many stations are down and delete them from the ring using the bypass relays in the wire center, all without human intervention.

It is instructive to compare the approaches taken to controlling the token bus and the token ring. The 802.4 committee was scared to death of having any centralized component that could fail in some unexpected way and take the system down with it. Therefore they designed a system in which the current token holder had special powers (e.g., soliciting bids to join the ring), but no station was otherwise different from the others (e.g., currently assigned administrative responsibility for maintenance).

The 802.5 committee, on the other hand, felt that having a centralized monitor made handling lost tokens, orphan frames and so on much easier. Furthermore, in a normal system, stations hardly ever crash, so occasionally having to put up with contention for a new monitor is not a great hardship. The price paid is that if the monitor ever really goes berserk but continues to issue ACTIVE MONITOR PRESENT control frames periodically, no station will ever challenge it. Monitors cannot be impeached.

This difference in approach comes from the different application areas the two committees had in mind. The 802.4 committee was thinking in terms of factories with large masses of metal moving around under computer control. Network failures could result in severe damage and had to be prevented at all costs. The 802.5 committee was interested in office automation, where a failure once in a rare while could be tolerated as the price for a simpler system. Whether 802.4 is, in fact, more reliable than 802.5 is a matter of some controversy.

4.3.4. Comparison of 802.3, 802.4, and 802.5

With three different and incompatible LANs available, each with different properties, many organizations are faced with the question: Which one should we install? In this section we will look at all three of the 802 LAN standards, pointing out their strengths and weaknesses, comparing and contrasting them.

To start with, it is worth noting that the three LAN standards use roughly similar technology and get roughly similar performance. While computer scientists and engineers can discuss the merits of coax versus twisted pair for hours on end if given half a chance, the people in the marketing, personnel, or accounting departments probably do not really care that much one way or the other.

Let us start with the advantages of 802.3. It is far and away the most widely used type at present, with a huge installed base and considerable operational experience. The protocol is simple. Stations can be installed on the fly, without taking the network down. A passive cable is used and modems are not required. Furthermore, the delay at low load is practically zero (stations do not have to wait for a token; they just transmit immediately).

On the other hand, 802.3 has a substantial analog component. Each station has to be able to detect the signal of the weakest other station, even when it itself is transmitting, and all of the collision detect circuitry in the transceiver is analog. Due to the possibility of having frames aborted by collisions, the minimum valid frame is 64 bytes, which represents substantial overhead when the data consist of just a single character from a terminal.

Furthermore, 802.3 is nondeterministic, which is often inappropriate for real-time work [although some real-time work is possible by simulating a token ring in software (Venkatramani and Chiueh, 1995)]. It also has no priorities. The cable length is limited to 2.5 km (at 10 Mbps) because the round-trip cable length determines the slot time, hence the performance. As the speed increases, the efficiency drops because the frame transmission times drop but the contention interval does not (the slot width is 2τ no matter what the data rate is). Alternatively, the cable has to be made shorter. Also, at high load, the presence of collisions becomes a major problem and can seriously affect the throughput.

Now let us consider 802.4, the token bus. It uses highly reliable cable television equipment, which is available off-the-shelf from numerous vendors. It is more deterministic than 802.3, although repeated losses of the token at critical moments can introduce more uncertainty than its supporters like to admit. It can handle short minimum frames.

Token bus also supports priorities and can be configured to provide a guaranteed fraction of the bandwidth to high-priority traffic, such as digitized voice. It also has excellent throughput and efficiency at high load, effectively becoming TDM. Finally, broadband cable can support multiple channels, not only for data, but also for voice and television.

On the down side, broadband systems use a lot of analog engineering and include modems and wideband amplifiers. The protocol is extremely complex and has substantial delay at low load (stations must always wait for the token, even in an otherwise idle system). Finally, it is poorly suited for fiber optic implementations and has a small installed base of users.

Now consider the token ring. It uses point-to-point connections, meaning that the engineering is easy and can be fully digital. Rings can be built using virtually any transmission medium from carrier pigeon to fiber optics. The standard twisted pair is cheap and simple to install. The use of wire centers make the token ring the only LAN that can detect and eliminate cable failures automatically.

Like the token bus, priorities are possible, although the scheme is not as fair. Also like the token bus, short frames are possible, but unlike the token bus, so are arbitrarily large ones, limited only by the token-holding time. Finally, the throughput and efficiency at high load are excellent, like the token bus and unlike 802.3.

The major minus is the presence of a centralized monitor function, which introduces a critical component. Even though a dead monitor can be replaced, a sick one can cause headaches. Furthermore, like all token passing schemes, there is always delay at low load because the sender must wait for the token.

It is also worth pointing out that there have been numerous studies of all three LANs. The principal conclusion we can draw from these studies is that we can draw no conclusions from them. One can always find a set of parameters that makes one of the LANs look better than the others. Under most circumstances, all three perform well, so that factors other than the performance are probably more important when making a choice.

4.3.5. IEEE Standard 802.6: Distributed Queue Dual Bus

None of the 802 LANs we have studied so far are suitable for use as a MAN. Cable length limitations and performance problems when thousands of stations are connected limits them to campus-sized areas. For networks covering an entire city, IEEE defined one MAN, called **DQDB** (**Distributed Queue Dual Bus**), as standard 802.6. In this section we will examine how it works. For additional information, see (Kessler and Train, 1992). A bibliography listing 171 papers about DQDB is given in (Sadiku and Arvind, 1994).

The basic geometry of 802.6 is illustrated in Fig. 1-4. Two parallel, unidirectional buses snake through the city, with stations attached to both buses in parallel. Each bus has a head-end, which generates a steady stream of 53-byte cells. Each cell travels downstream from the head-end. When it reaches the end, it falls off the bus.

Each cell carries a 44-byte payload field, making it compatible with some AAL modes. Each cell also holds two protocol bits, *Busy*, set to indicate that a cell is occupied, and *Request*, which can be set when a station wants to make a request.

To transmit a cell, a station has to know whether the destination is to the left of it or to the right of it. If the destination is to the right, the sender uses bus *A*. Otherwise, it uses bus *B*. Data are inserted onto either bus using a wired-OR circuit, so failure of a station does not take down the network.

Unlike all the other 802 LAN protocols, 802.6 is not greedy. In all the others, if a station gets the chance to send, it will. Here, stations queue up in the order

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they became ready to send and transmit in FIFO order. The interesting part about the protocol is how it achieves FIFO order without having a central queue.

The basic rule is that stations are polite: they defer to stations downstream from them. This politeness is needed to prevent a situation in which the station nearest the head-end simply grabs all the empty cells as they come by and fills them up, starving everyone downstream. For simplicity, we will only examine transmission on bus A, but the same story holds for bus B as well.

To simulate the FIFO queue, each station maintains two counters, RC and CD. RC (*Request Counter*) counts the number of downstream requests pending until the station itself has a frame to send. At that point, RC is copied to CD, RC is reset to 0, and now counts the number of requests made after the station became ready. For example, if $CD = 3$ and $RC = 2$ for station k, the next three empty cells that pass by station k are reserved for downstream stations, then station k may send, then two more cells are reserved for downstream stations. For simplicity, we assume a station can have only one cell ready for transmission at a time.

To send a cell, a station must first make a reservation by setting the *Request* bit in some cell on the reverse bus (i.e., on bus B for a transmission that will later take place on bus A). As this cell propagates down the reverse bus, every station along the way notes it and increments its RC. To illustrate this concept, we will use an example. Initially, all the RC counters are 0, and no cells are queued up, as shown in Fig. 4-32(a). Then station D makes a request, which causes stations, C, B, and A, to increment their RC counters, as shown in Fig. 4-32(b). After that, B makes a request, copying its current RC value into CD, leading to the situation of Fig. 4-32(c).

At this point, the head-end on bus A generates an empty cell. As it passes by B, that station sees that its $CD > 0$, so it may not use the empty cell. (When a station has a cell queued, CD represents its position in the queue, with 0 being the front of the queue.) Instead it decrements CD. When the still-empty cell gets to D, that station sees that $CD = 0$, meaning that no one is ahead of it on the queue, so it ORs its data into the cell and sets the *Busy* bit. After the transmissions are done, we have the situation of Fig. 4-32(d).

When the next empty cell is generated, station B sees that it is now at the head of the queue, and seizes the cell (by setting 1 bit), as illustrated in Fig. 4-32(e). In this way, stations queue up to take turns, without a centralized queue manager.

DQDB systems are now being installed by many carriers throughout entire cities. Typically they run for up to 160 km at speeds of 44.736 Mbps (T3).

4.3.6. IEEE Standard 802.2: Logical Link Control

It is now perhaps time to step back and compare what we have learned in this chapter with what we studied in the previous one. In Chap. 3, we saw how two machines could communicate reliably over an unreliable line by using various

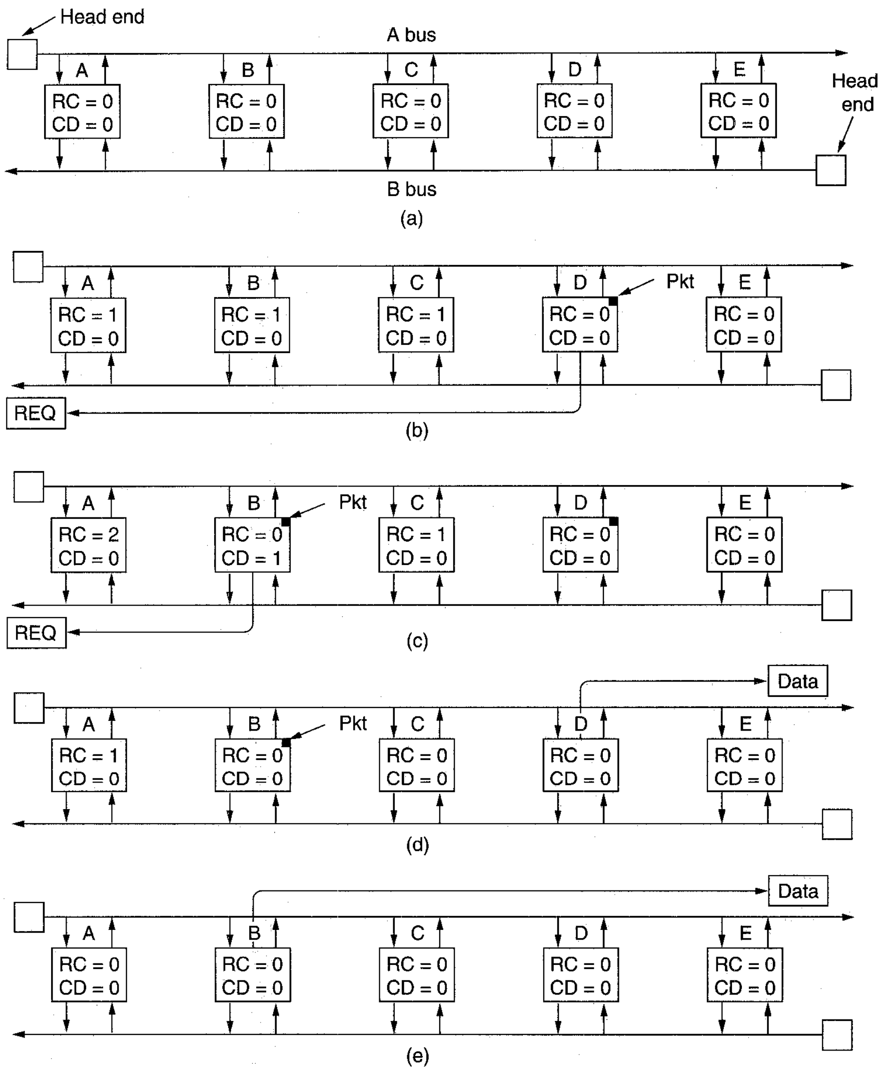

Fig. 4-32. (a) Initially the MAN is idle. (b) After *D* makes a request. (c) After *B* makes a request. (d) After *D* transmits. (e) After *B* transmits.

data link protocols. These protocols provided error control (using acknowledgements) and flow control (using a sliding window).

In contrast, in this chapter, we have not said a word about reliable communication. All that the 802 LANs and MAN offer is a best-efforts datagram service.

Sometimes, this service is adequate. For example, for transporting IP packets, no guarantees are required or even expected. An IP packet can just be inserted into an 802 payload field and sent on its way. If it gets lost, so be it.

Nevertheless, there are also systems in which an error-controlled, flow-controlled data link protocol is desired. IEEE has defined one that can run on top of all the 802 LAN and MAN protocols. In addition, this protocol, called **LLC** (**Logical Link Control**), hides the differences between the various kinds of 802 networks by providing a single format and interface to the network layer. This format, interface, and protocol are all closely based on OSI. LLC forms the upper half of the data link layer, with the MAC sublayer below it, as shown in Fig. 4-33.

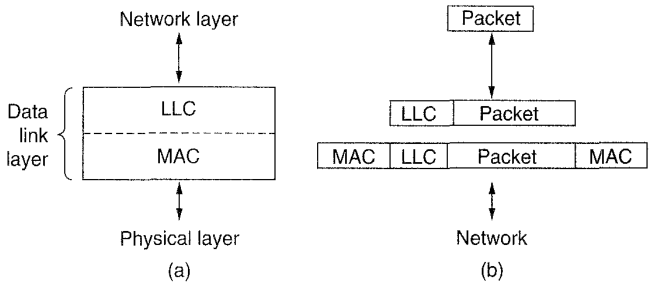

Fig. 4-33. (a) Position of LLC. (b) Protocol formats.

Typical usage of LLC is as follows. The network layer on the sending machine passes a packet to LLC using the LLC access primitives. The LLC sublayer then adds an LLC header, containing sequence and acknowledgement numbers. The resulting structure is then inserted into the payload field of an 802.x frame and transmitted. At the receiver, the reverse process takes place.

LLC provides three service options: unreliable datagram service, acknowledged datagram service, and reliable connection-oriented service. The LLC header is based on the older HDLC protocol. A variety of different formats are used for data and control. For acknowledged datagram or connection-oriented service, the data frames contain a source address, a destination address, a sequence number, an acknowledgement number, and a few miscellaneous bits. For unreliable datagram service, the sequence number and acknowledgement number are omitted.

4.4. BRIDGES

Many organizations have multiple LANs and wish to connect them. LANs can be connected by devices called **bridges**, which operate in the data link layer. This statement means that bridges do not examine the network layer header and can thus copy IP, IPX, and OSI packets equally well. In contrast, a pure IP, IPX, or OSI router can handle only its own native packets.

In the following sections we will look at bridge design, especially for connecting 802.3, 802.4, and 802.5 LANs. For a comprehensive treatment of bridges and related topics, see (Perlman, 1992). Before getting into the technology of bridges, it is worthwhile taking a look at some common situations in which bridges are used. We will mention six reasons why a single organization may end up with multiple LANs. First, many university and corporate departments have their own LANs, primarily to connect their own personal computers, workstations, and servers. Since the goals of the various departments differ, different departments choose different LANs, without regard to what other departments are doing. Sooner or later, there is a need for interaction, so bridges are needed. In this example, multiple LANs came into existence due to the autonomy of their owners.

Second, the organization may be geographically spread over several buildings separated by considerable distances. It may be cheaper to have separate LANs in each building and connect them with bridges and infrared links than to run a single coaxial cable over the entire site.

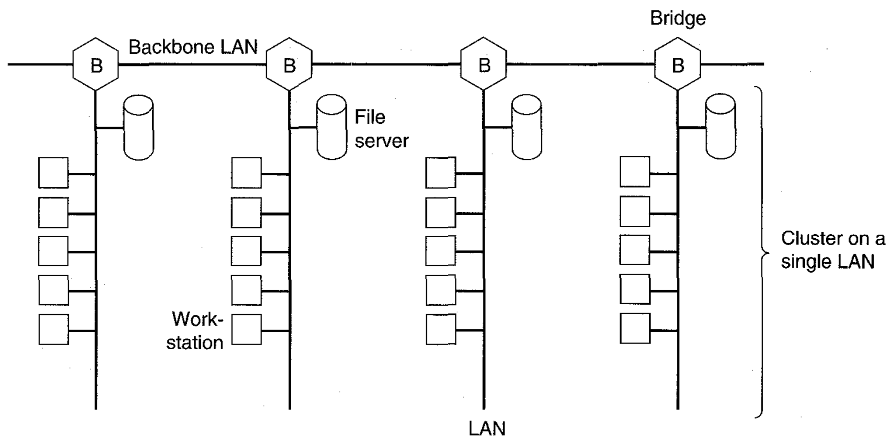

Fig. 4-34. Multiple LANs connected by a backbone to handle a total load higher than the capacity of a single LAN.

Third, it may be necessary to split what is logically a single LAN into separate LANs to accommodate the load. At many universities, for example, thousands of workstations are available for student and faculty computing. Files are normally kept on file server machines, and are downloaded to users' machines upon request. The enormous scale of this system precludes putting all the workstations on a single LAN—the total bandwidth needed is far too high. Instead multiple LANs connected by bridges are used, as shown in Fig. 4-34. Each LAN contains a cluster of workstations with its own file server, so that most traffic is restricted to a single LAN and does not add load to the backbone.

Fourth, in some situations, a single LAN would be adequate in terms of the load, but the physical distance between the most distant machines is too great (e.g., more than 2.5 km for 802.3). Even if laying the cable is easy to do, the network would not work due to the excessively long round-trip delay. The only solution is to partition the LAN and install bridges between the segments. Using bridges, the total physical distance covered can be increased.

Fifth, there is the matter of reliability. On a single LAN, a defective node that keeps outputting a continuous stream of garbage will cripple the LAN. Bridges can be inserted at critical places, like fire doors in a building, to prevent a single node which has gone berserk from bringing down the entire system. Unlike a repeater, which just copies whatever it sees, a bridge can be programmed to exercise some discretion about what it forwards and what it does not forward.

Sixth, and last, bridges can contribute to the organization's security. Most LAN interfaces have a **promiscuous mode**, in which *all* frames are given to the computer, not just those addressed to it. Spies and busybodies love this feature. By inserting bridges at various places and being careful not to forward sensitive traffic, it is possible to isolate parts of the network so that its traffic cannot escape and fall into the wrong hands.

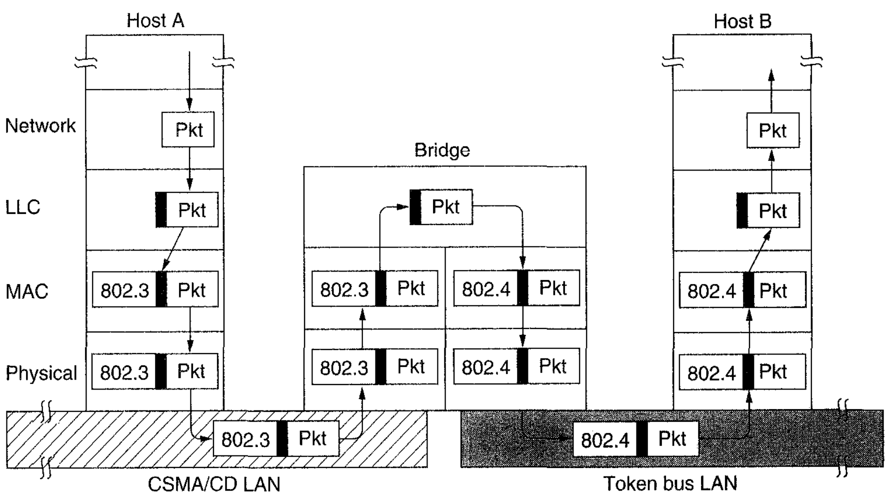

Fig. 4-35. Operation of a LAN bridge from 802.3 to 802.4.

Having seen why bridges are needed, let us now turn to the question of how they work. Figure 4-35 illustrates the operation of a simple two-port bridge. Host *A* has a packet to send. The packet descends into the LLC sublayer and acquires an LLC header. Then it passes into the MAC sublayer and an 802.3 header is prepended to it (also a trailer, not shown in the figure). This unit goes out onto the cable and eventually is passed up to the MAC sublayer in the bridge, where the 802.3 header is stripped off. The bare packet (with LLC header) is then handed off to the LLC sublayer in the bridge. In this example, the packet is destined for an 802.4 subnet connected to the bridge, so it works its way down the 802.4 side of the bridge and off it goes. Note that a bridge connecting *k* different LANs will have *k* different MAC sublayers and *k* different physical layers, one for each type.

4.4.1. Bridges from 802.x to 802.y

You might naively think that a bridge from one 802 LAN to another one would be completely trivial. Such is not the case. In the remainder of this section we will point out some of the difficulties that will be encountered when trying to build a bridge between the various 802 LANs.

Each of the nine combinations of 802.x to 802.y has its own unique set of problems. However, before dealing with these one at a time, let us look at some general problems common to all the bridges. To start with, each of the LANs uses a different frame format (see Fig. 4-36). There is no valid technical reason for this incompatibility. It is just that none of the corporations supporting the three standards (Xerox, GM, and IBM) wanted to change *theirs*. As a result, any copying between different LANs requires reformatting, which takes CPU time, requires a new checksum calculation, and introduces the possibility of undetected errors due to bad bits in the bridge's memory. None of this would have been necessary if the three committees had been able to agree on a single format.

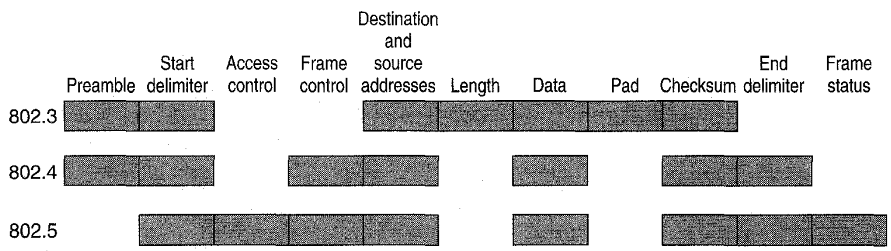

Fig. 4-36. The IEEE 802 frame formats.

A second problem is that interconnected LANs do not necessarily run at the same data rate. When forwarding a long run of back-to-back frames from a fast LAN to a slower one, the bridge will not be able to get rid of the frames as fast as they come in. It will have to buffer them, hoping not to run out of memory. The problem also exists from 802.4 to 802.3 at 10 Mbps to some extent because some of 802.3's bandwidth is lost to collisions. It does not really have 10 Mbps, whereas 802.4 really does (well, almost). Bridges that connect three or more LANs have a similar problem when several LANs are trying to feed the same output LAN at the same time.

A subtle, but important problem related to the bridge-as-bottleneck problem is the value of timers in the higher layers. Suppose that the network layer on an 802.4 LAN is trying to send a very long message as a sequence of frames. After sending the last one it starts a timer to wait for an acknowledgement. If the message has to transit a bridge to a slower 802.5 LAN, there is a danger that the timer will go off before the last frame has been forwarded onto the slower LAN. The network layer will assume the problem is due to a lost frame and just retransmit the entire sequence again. After n failed attempts it may give up and tell the transport layer that the destination is dead.

A third, and potentially most serious problem of all, is that all three 802 LANs have a different maximum frame length. For 802.3 it depends on the parameters of the configuration, but for the standard 10-Mbps system the payload is a maximum of 1500 bytes. For 802.4 it is fixed at 8191 bytes. For 802.5 there is no upper limit, except that a station may not transmit longer than the token-holding time. With the default value of 10 msec, the maximum frame length is 5000 bytes.

An obvious problem arises when a long frame must be forwarded onto a LAN that cannot accept it. Splitting the frame into pieces is out of the question in this layer. All the protocols assume that frames either arrive or they do not. There is no provision for reassembling frames out of smaller units. This is not to say that such protocols could not be devised. They could be and have been. It is just that 802 does not provide this feature. Basically, there is no solution. Frames that are too large to be forwarded must be discarded. So much for transparency.

Now let us briefly consider each of the nine cases of 802.x to 802.y bridges to see what other problems are lurking in the shadows. From 802.3 to 802.3 is easy. The only thing that can go wrong is that the destination LAN is so heavily loaded that frames keep pouring into the bridge, but the bridge cannot get rid of them. If this situation persists long enough, the bridge might run out of buffer space and begin dropping frames. Since this problem is always potentially present when forwarding onto an 802.3 LAN, we will not mention it further. With the other two LANs, each station, including the bridge is guaranteed to acquire the token periodically and cannot be shut out for long intervals.

From 802.4 to 802.3 two problems exist. First, 802.4 frames carry priority bits that 802.3 frames do not have. As a result, if two 802.4 LANs communicate via an 802.3 LAN, the priority will be lost by the intermediate LAN.

The second problem is caused by a specific feature in 802.4: temporary token handoff. It is possible for an 802.4 frame to have a header bit set to 1 to temporarily pass the token to the destination, to let it send an acknowledgement frame. However, if such a frame is forwarded by a bridge, what should the bridge

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do? If it sends an acknowledgement frame itself, it is lying because the frame really has not been delivered yet. In fact, the destination may be dead.

On the other hand, if it does not generate the acknowledgement, the sender will almost assuredly conclude that the destination is dead and report back failure to its superiors. There does not seem to be any way to solve this problem.

From 802.5 to 802.3 we have a similar problem. The 802.5 frame format has *A* and *C* bits in the frame status byte. These bits are set by the destination to tell the sender whether the station addressed saw the frame, and whether it copied it. Here again, the bridge can lie and say the frame has been copied, but if it later turns out that the destination is down, serious problems may arise. In essence, the insertion of a bridge into the network has changed the semantics of the bits. It is hard to imagine a proper solution to this problem.

From 802.3 to 802.4 we have the problem of what to put in the priority bits. A good case can be made for having the bridge retransmit all frames at the highest priority, because they have probably suffered enough delay already.

From 802.4 to 802.4 the only problem is what to do with the temporary token handoff. At least here we have the possibility of the bridge managing to forward the frame fast enough to get the response before the timer runs out. Still it is a gamble. By forwarding the frame at the highest priority, the bridge is telling a little white lie, but it thereby increases the probability of getting the response in time.

From 802.5 to 802.4 we have the same problem with the *A* and *C* bits as before. Also, the definition of the priority bits is different for the two LANs, but beggars can't be choosers. At least the two LANs have the same number of priority bits. All the bridge can do is copy the priority bits across and hope for the best.

From 802.3 to 802.5 the bridge must generate priority bits, but there are no other special problems. From 802.4 to 802.5 there is a potential problem with frames that are too long and the token handoff problem is present again. Finally, from 802.5 to 802.5 the problem is what to do with the *A* and *C* bits again. Figure 4-37 summarizes the various problems we have been discussing.

When the IEEE 802 committee set out to come up with a LAN standard, it was unable to agree on a single standard, so it produced *three* incompatible standards, as we have just seen in some detail. For this failure, it has been roundly criticized. When it was later assigned the job of designing a standard for bridges to interconnect its three incompatible LANs, it resolved to do better. It did. It came up with *two* incompatible bridge designs. So far nobody has asked it to design a gateway standard to connect its two incompatible bridges, but at least the trend is in the right direction.

This section has dealt with the problems encountered in connecting two IEEE 802 LANs via a single bridge. The next two sections deal with the problems of connecting large internetworks containing many LANs and many bridges and the two IEEE approaches to designing these bridges.

Source LAN	Destination LAN: 802.3 (CSMA/CD)	802.4 (Token bus)	802.4 (Token ring)
802.3		1, 4	1, 2, 4, 8
802.4	1, 5, 8, 9, 10	9	1, 2, 3, 8, 9, 10
802.5	1, 2, 5, 6, 7, 10	1, 2, 3, 6, 7	6, 7

Actions:
1. Reformat the frame and compute new checksum
2. Reverse the bit order.
3. Copy the priority, meaningful or not.
4. Generate a ficticious priority.
5. Discard priority.
6. Drain the ring (somehow).
7. Set A and C bits (by lying).
8. Worry about congestion (fast LAN to slow LAN).
9. Worry about token handoff ACK being delayed or impossible.
10. Panic if frame is too long for destination LAN.

Parameters assumed:

802.3:	1500-byte frames,	10 Mbps (minus collisions)
802.4:	8191-byte frames	10 Mbps
802.5:	5000-byte frames	4 Mbps

Fig. 4-37. Problems encountered in building bridges from 802.x to 802.y.

4.4.2. Transparent Bridges

The first 802 bridge is a **transparent bridge** or **spanning tree bridge** (Perlman, 1992). The overriding concern of the people who supported this design was complete transparency. In their view, a site with multiple LANs should be able to go out and buy bridges designed to the IEEE standard, plug the connectors into the bridges, and everything should work perfectly, instantly. There should be no hardware changes required, no software changes required, no setting of address switches, no downloading of routing tables or parameters, nothing. Just plug in the cables and walk away. Furthermore, the operation of the existing LANs should not be affected by the bridges at all. Surprisingly enough, they actually succeeded.

A transparent bridge operates in promiscuous mode, accepting every frame transmitted on all the LANs to which it is attached. As an example, consider the configuration of Fig. 4-38. Bridge B1 is connected to LANs 1 and 2, and bridge B2 is connected to LANs 2, 3, and 4. A frame arriving at bridge B1 on LAN 1 destined for *A* can be discarded immediately, because it is already on the right LAN, but a frame arriving on LAN 1 for *C* or *F* must be forwarded.

When a frame arrives, a bridge must decide whether to discard or forward it,

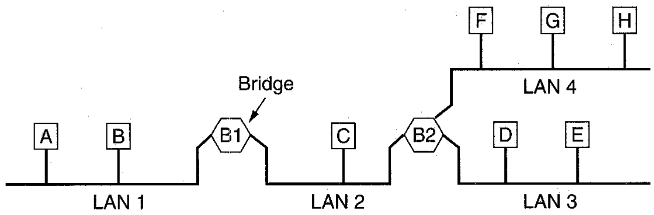

Fig. 4-38. A configuration with four LANs and two bridges.

and if the latter, on which LAN to put the frame. This decision is made by looking up the destination address in a big (hash) table inside the bridge. The table can list each possible destination and tell which output line (LAN) it belongs on. For example, B2's table would list *A* as belonging to LAN 2, since all B2 has to know is which LAN to put frames for *A* on. That, in fact, more forwarding happens later is not of interest to it.

When the bridges are first plugged in, all the hash tables are empty. None of the bridges know where any of the destinations are, so they use the flooding algorithm: every incoming frame for an unknown destination is output on all the LANs to which the bridge is connected except the one it arrived on. As time goes on, the bridges learn where destinations are, as described below. Once a destination is known, frames destined for it are put on only the proper LAN and are not flooded.

The algorithm used by the transparent bridges is **backward learning**. As mentioned above, the bridges operate in promiscuous mode, so they see every frame sent on any of their LANs. By looking at the source address, they can tell which machine is accessible on which LAN. For example, if bridge B1 in Fig. 4-38 sees a frame on LAN 2 coming from *C*, it knows that *C* must be reachable via LAN 2, so it makes an entry in its hash table noting that frames going to *C* should use LAN 2. Any subsequent frame addressed to *C* coming in on LAN 1 will be forwarded, but a frame for *C* coming in on LAN 2 will be discarded.

The topology can change as machines and bridges are powered up and down and moved around. To handle dynamic topologies, whenever a hash table entry is made, the arrival time of the frame is noted in the entry. Whenever a frame whose destination is already in the table arrives, its entry is updated with the current time. Thus the time associated with every entry tells the last time a frame from that machine was seen.

Periodically, a process in the bridge scans the hash table and purges all entries more than a few minutes old. In this way, if a computer is unplugged from its LAN, moved around the building, and replugged in somewhere else, within a few minutes it will be back in normal operation, without any manual intervention. This algorithm also means that if a machine is quiet for a few minutes, any traffic sent to it will have to be flooded, until it next sends a frame itself.

The routing procedure for an incoming frame depends on the LAN it arrives on (the source LAN) and the LAN its destination is on (the destination LAN), as follows:

1. If destination and source LANs are the same, discard the frame.
2. If the destination and source LANs are different, forward the frame.
3. If the destination LAN is unknown, use flooding.

As each frame arrives, this algorithm must be applied. Special purpose VLSI chips exist to do the lookup and update the table entry, all in a few microseconds.

To increase reliability, some sites use two or more bridges in parallel between pairs of LANs, as shown in Fig. 4-39. This arrangement, however, also introduces some additional problems because it creates loops in the topology.

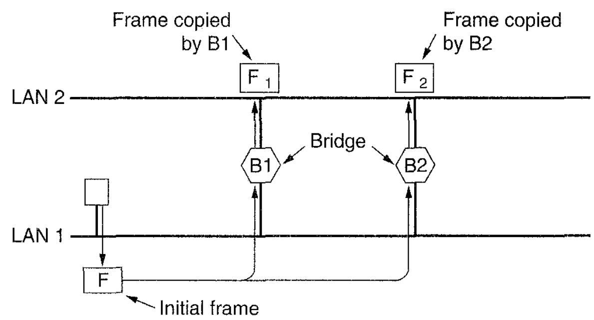

Fig. 4-39. Two parallel transparent bridges.

A simple example of these problems can be seen by observing how a frame, F, with unknown destination is handled in Fig. 4-39. Each bridge, following the normal rules for handling unknown destinations, uses flooding, which in this example, just means copying it to LAN 2. Shortly thereafter, bridge 1 sees F_2, a frame with an unknown destination, which it copies to LAN 1, generating F_3 (not shown). Similarly, bridge 2 copies F_1 to LAN 1 generating F_4 (also not shown). Bridge 1 now forwards F_4 and bridge 2 copies F_3. This cycle goes on forever.

Spanning Tree Bridges

The solution to this difficulty is for the bridges to communicate with each other and overlay the actual topology with a spanning tree that reaches every LAN. In effect, some potential connections between LANs are ignored in the interest of constructing a fictitious loop-free topology. For example, in Fig. 4-40(a) we see nine LANs interconnected by ten bridges. This configuration can be abstracted into a graph with the LANs as the nodes. An arc connects any two LANs that are connected by a bridge. The graph can be reduced to a spanning tree by dropping the arcs shown as dotted lines in Fig. 4-40(b). Using this spanning tree, there is exactly one path from every LAN to every other LAN. Once the bridges have agreed on the spanning tree, all forwarding between LANs follows the spanning tree. Since there is a unique path from each source to each destination, loops are impossible.

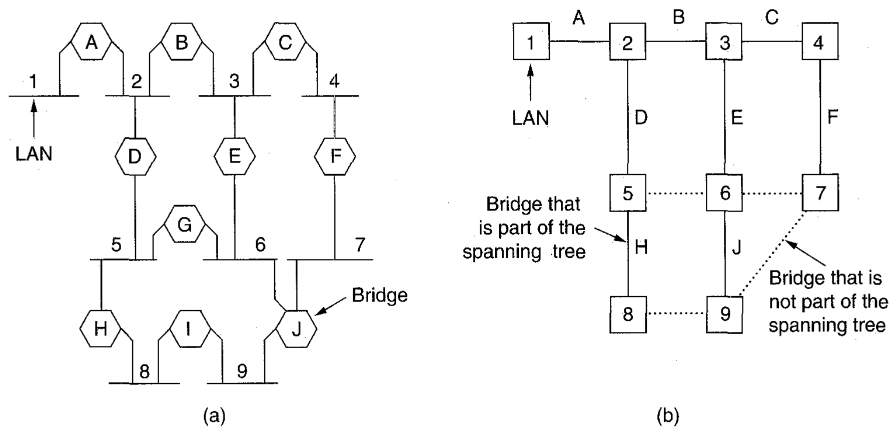

Fig. 4-40. (a) Interconnected LANs. (b) A spanning tree covering the LANs. The dotted lines are not part of the spanning tree.

To build the spanning tree, first the bridges have to choose one bridge to be the root of the tree. They make this choice by having each one broadcast its serial number, installed by the manufacturer, and guaranteed to be unique worldwide. The bridge with the lowest serial number becomes the root. Next, a tree of shortest paths from the root to every bridge and LAN is constructed. This tree is the spanning tree. If a bridge or LAN fails, a new one is computed.

The result of this algorithm is that a unique path is established from every LAN to the root, and thus to every other LAN. Although the tree spans all the LANs, not all the bridges are necessarily present in the tree (to prevent loops). Even after the spanning tree has been established, the algorithm continues to run in order to automatically detect topology changes and update the tree. The distributed algorithm used for constructing the spanning tree was invented by Perlman and is described in detail in (Perlman, 1992).

Bridges can also be used to connect LANs that are widely separated. In this model, each site consists of a collection of LANs and bridges, one of which has a connection to a WAN. Frames for remote LANs travel over the WAN. The basic spanning tree algorithm can be used, preferably with certain optimizations to select a tree that minimizes the amount of WAN traffic.

4.4.3. Source Routing Bridges

Transparent bridges have the advantage of being easy to install. You just plug them in and walk away. On the other hand, they do not make optimal use of the bandwidth, since they only use a subset of the topology (the spanning tree). The relative importance of these two (and other) factors led to a split within the 802 committees (Pitt, 1988). The CSMA/CD and token bus people chose the transparent bridge. The ring people (with encouragement from IBM) preferred a scheme called **source routing**, which we will now describe. For additional details, see (Dixon, 1987).

Reduced to its barest essentials, source routing assumes that the sender of each frame knows whether or not the destination is on its own LAN. When sending a frame to a different LAN, the source machine sets the high-order bit of the source address to 1, to mark it. Furthermore, it includes in the frame header the exact path that the frame will follow.

This path is constructed as follows. Each LAN has a unique 12-bit number, and each bridge has a 4-bit number that uniquely identifies it in the context of its LANs. Thus, two bridges far apart may both have number 3, but two bridges between the same two LANs must have different bridge numbers. A route is then a sequence of bridge, LAN, bridge, LAN, ... numbers. Referring to Fig. 4-38, the route from *A* to *D* would be (L1, B1, L2, B2, L3).

A source routing bridge is only interested in those frames with the high-order bit of the destination set to 1. For each such frame that it sees, it scans the route looking for the number of the LAN on which the frame arrived. If this LAN number is followed by its own bridge number, the bridge forwards the frame onto the LAN whose number follows its bridge number in the route. If the incoming LAN number is followed by the number of some other bridge, it does not forward the frame.

This algorithm lends itself to three possible implementations:

1. Software: the bridge runs in promiscuous mode, copying all frames to its memory to see if they have the high-order destination bit set to 1. If so, the frame is inspected further; otherwise it is not.

2. Hybrid: the bridge's LAN interface inspects the high-order destination bit and only accepts frames with the bit set. This interface is easy to build into hardware and greatly reduces the number of frames the bridge must inspect.

3. Hardware: the bridge's LAN interface not only checks the high-order destination bit, but it also scans the route to see if this bridge must do forwarding. Only frames that must actually be forwarded are given to the bridge. This implementation requires the most complex hardware but wastes no bridge CPU cycles because all irrelevant frames are screened out.

These three implementations vary in their cost and performance. The first one has no additional hardware cost for the interface but may require a very fast CPU to handle all the frames. The last one requires a special VLSI chip but offloads much of the processing from the bridge to the chip, so that a slower CPU can be used, or alternatively, the bridge can handle more LANs.

Implicit in the design of source routing is that every machine in the internetwork knows, or can find, the best path to every other machine. How these routes are discovered is an important part of the source routing algorithm. The basic idea is that if a destination is unknown, the source issues a broadcast frame asking where it is. This **discovery frame** is forwarded by every bridge so that it reaches every LAN on the internetwork. When the reply comes back, the bridges record their identity in it, so that the original sender can see the exact route taken and ultimately choose the best route.

While this algorithm clearly finds the best route (it finds *all* routes), it suffers from a frame explosion. Consider the configuration of Fig. 4-41, with N LANs linearly connected by triple bridges. Each discovery frame sent by station 1 is copied by each of the three bridges on LAN 1, yielding three discovery frames on LAN 2. Each of these is copied by each of the bridges on LAN 2, resulting in nine frames on LAN 3. By the time we reach LAN N, 3^{N-1} frames are circulating. If a dozen sets of bridges are traversed, more than half a million discovery frames will have to be injected into the last LAN, causing severe congestion.

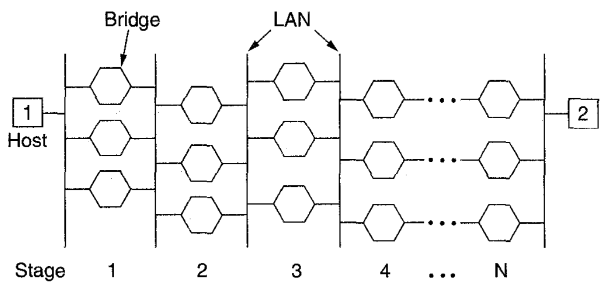

Fig. 4-41. A series of LANs connected by triple bridges.

A somewhat analogous process happens with the transparent bridge, only it is not nearly so severe. When an unknown frame arrives, it is flooded, but only along the spanning tree, so the total volume of frames sent is linear with the size of the network, not exponential.

Once a host has discovered a route to a certain destination, it stores the route in a cache, so that the discovery process will not have to be run next time. While this approach greatly limits the impact of the frame explosion, it does put some administrative burden on all the hosts, and the whole algorithm is definitely not transparent, which was one of the original goals, as we mentioned above.

4.4.4. Comparison of 802 Bridges

The transparent and source routing bridges each have advantages and disadvantages. In this section we will discuss some of the major ones. They are summarized in Fig. 4-42 and covered in more detail in (Soha and Perlman, 1988; and Zhang, 1988). Be warned, however, that every one of the points is highly contested.

Issue	Transparent bridge	Source routing bridge
Orientation	Connectionless	Connection-oriented
Transparency	Fully transparent	Not transparent
Configuration	Automatic	Manual
Routing	Suboptimal	Optimal
Locating	Backward learning	Discovery frames
Failures	Handled by the bridges	Handled by the hosts
Complexity	In the bridges	In the hosts

Fig. 4-42. Comparison of transparent and source routing bridges.

At the heart of the difference between the two bridge types is the distinction between connectionless and connection-oriented networking. The transparent bridges have no concept of a virtual circuit at all and route each frame independently from all the others. The source routing bridges, in contrast, determine a route using discovery frames and then use that route thereafter.

The transparent bridges are completely invisible to the hosts and are fully compatible with all existing 802 products. The source routing bridges are neither transparent nor compatible. To use source routing, hosts must be fully aware of the bridging scheme and must actively participate in it. Splitting an existing LAN into two LANs connected by a source routing bridge requires making changes to the host software.

When using transparent bridges, no network management is needed. The bridges configure themselves to the topology automatically. With source routing bridges, the network manager must manually install the LAN and bridge numbers. Mistakes, such as duplicating a LAN or bridge number, can be very difficult to detect, as they may cause some frames to loop, but not others on different routes. Furthermore, when connecting two previously disjoint internetworks, with transparent bridges there is nothing to do except connect them, whereas with source routing, it may be necessary to manually change many LAN numbers to make them unique in the combined internetwork.

One of the few advantages of source routing is that, in theory, it can use optimal routing, whereas transparent bridging is restricted to the spanning tree.

Furthermore, source routing can also make good use of parallel bridges between two LANs to split the load. Whether actual bridges will be clever enough to make use of these theoretical advantages is questionable.

Locating destinations is done using backward learning in the transparent bridge and using discovery frames in source routing bridges. The disadvantage of backward learning is that the bridges have to wait until a frame from a particular machine happens to come along in order to learn where that machine is. The disadvantage of discovery frames is the exponential explosion in moderate to large internetworks with parallel bridges.

Failure handling is quite different in the two schemes. Transparent bridges learn about bridge and LAN failures and other topology changes quickly and automatically, just from listening to each other's control frames. Hosts do not notice these changes at all.

With source routing, the situation is quite different. When a bridge fails, machines that are routing over it initially notice that their frames are no longer being acknowledged, so they time out and try over and over. Finally, they conclude that something is wrong, but they still do not know if the problem is with the destination itself, or with the current route. Only by sending another discovery frame can they see if the destination is available. Unfortunately, when a major bridge fails, a large number of hosts will have to experience timeouts and send new discovery frames before the problem is resolved, even if an alternative route is available. This greater vulnerability to failures is one of the major weaknesses of all connection-oriented systems.

Finally, we come to complexity and cost, a very controversial topic. If source routing bridges have a VLSI chip that reads in only those frames that must be forwarded, these bridges will experience a lighter frame processing load and deliver a better performance for a given investment in hardware. Without this chip they will do worse because the amount of processing per frame (searching the route in the frame header) is substantially more.

In addition, source routing puts extra complexity in the hosts: they must store routes, send discovery frames, and copy route information into each frame. All of these things require memory and CPU cycles. Since there are typically one to two orders of magnitude more hosts than bridges, it may be better to put the extra cost and complexity into a few bridges, rather than in all the hosts.

4.4.5. Remote Bridges

A common use of bridges is to connect two (or more) distant LANs. For example, a company might have plants in several cities, each with its own LAN. Ideally, all the LANs should be interconnected, so the complete system acts like one large LAN.

This goal can be achieved by putting a bridge on each LAN and connecting the bridges pairwise with point-to-point lines (e.g., lines leased from a telephone company). A simple system, with three LANs, is illustrated in Fig. 4-43. The usual routing algorithms apply here. The simplest way to see this is to regard the three point-to-point lines as hostless LANs. Then we have a normal system of six LANS interconnected by four bridges. Nothing in what we have studied so far says that a LAN must have hosts on it.

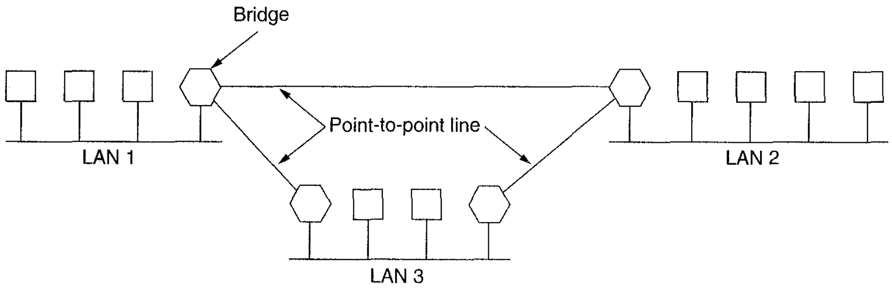

Fig. 4-43. Remote bridges can be used to interconnect distant LANs.

Various protocols can be used on the point-to-point lines. One possibility is to choose some standard point-to-point data link protocol, putting complete MAC frames in the payload field. This strategy works best if all the LANs are identical, and the only problem is getting frames to the right LAN. Another option is to strip off the MAC header and trailer at the source bridge and put what is left in the payload field of the point-to-point protocol. A new MAC header and trailer can then be generated at the destination bridge. A disadvantage of this approach is that the checksum that arrives at the destination host is not the one computed by the source host, so errors caused by bad bits in a bridge's memory may not be detected.

4.5. HIGH-SPEED LANS

The 802 LANs and MAN we have just studied are all based on one copper wire (two copper wires for 802.6). For low speeds and short distances, this will do just fine, but for high speeds and longer distances LANs must be based on fiber optics or highly parallel copper networks. Fiber has high bandwidth, is thin and lightweight, is not affected by electromagnetic interference from heavy machinery (important when cabling runs through elevator shafts), power surges, or lightning, and has excellent security because it is nearly impossible to wiretap without detection. Consequently, fast LANs often use fiber. In the following sections we will look at some local area networks that use fiber optics, as well as one extremely high-speed LAN that uses old fashioned copper wire (but lots of it).

4.5.1. FDDI

FDDI (Fiber Distributed Data Interface) is a high-performance fiber optic token ring LAN running at 100 Mbps over distances up to 200 km with up to 1000 stations connected (Black, 1994; Jain, 1994; Mirchandani and Khanna, 1993; Ross and Hamstra, 1993; Shah and Ramakrishnan, 1994; and Wolter, 1990). It can be used in the same way as any of the 802 LANs, but with its high bandwidth, another common use is as a backbone to connect copper LANs, as shown in Fig. 4-44. FDDI-II is the successor to FDDI, modified to handle synchronous circuit-switched PCM data for voice or ISDN traffic, in addition to ordinary data. We will refer to both of them as just FDDI. This section deals with both the physical layer and the MAC sublayer of FDDI.

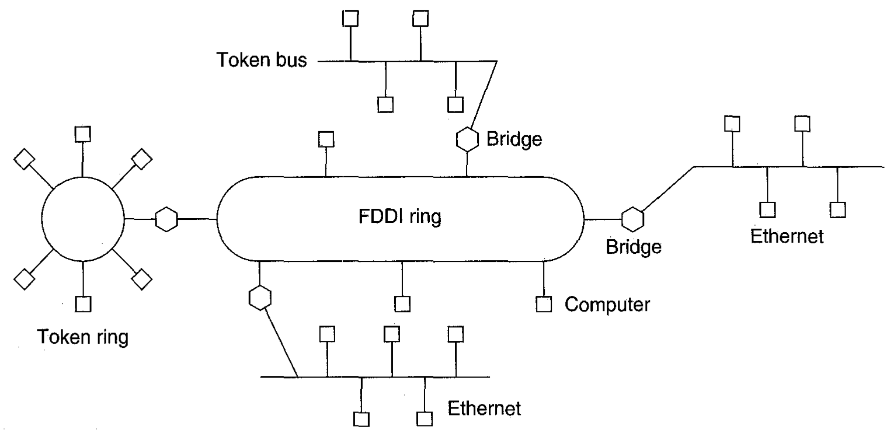

Fig. 4-44. An FDDI ring being used as a backbone to connect LANs and computers.

FDDI uses multimode fibers because the additional expense of single mode fibers is not needed for networks running at only 100 Mbps. It also uses LEDs rather than lasers, not only due to their lower cost, but also because FDDI may sometimes be used to connect directly to user workstations. There is a danger that curious users may occasionally unplug the fiber connector and look directly into it to watch the bits go by at 100 Mbps. With a laser the curious user might end up with a hole in his retina. LEDs are too weak to do any eye damage but are strong enough to transfer data accurately at 100 Mbps. The FDDI design specification calls for no more than 1 error in 2.5×10^{10} bits. Many implementations do much better.

The FDDI cabling consists of two fiber rings, one transmitting clockwise and the other transmitting counterclockwise, as illustrated in Fig. 4-45(a). If either one breaks, the other can be used as a backup. If both break at the same point, for example, due to a fire or other accident in the cable duct, the two rings can be joined into a single ring approximately twice as long, as shown in Fig. 4-45(b). Each station contains relays that can be used to join the two rings or bypass the station in the event of station problems. Wire centers can also be used, as in 802.5.

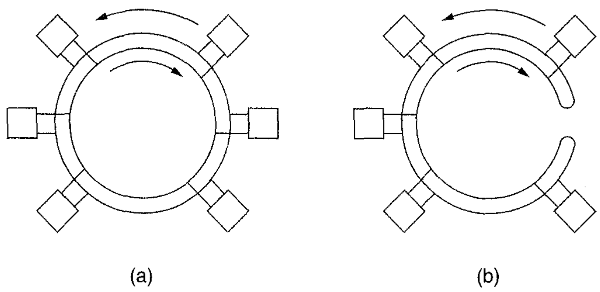

Fig. 4-45. (a) FDDI consists of two counterrotating rings. (b) In the event of failure of both rings at one point, the two rings can be joined together to form a single long ring.

FDDI defines two classes of stations, *A* and *B*. Class *A* stations connect to both rings. The cheaper class *B* stations only connect to one of the rings. Depending on how important fault tolerance is, an installation can choose class *A* or class *B* stations, or some of each.

The physical layer does not use Manchester encoding because 100-Mbps Manchester encoding requires 200 megabaud, which was deemed too expensive. Instead a scheme called **4 out of 5** encoding is used. Each group of 4 MAC symbols (0s, 1s, and certain nondata symbols such as start-of-frame) are encoded as a group of 5 bits on the medium. Sixteen of the 32 combinations are for data, 3 are for delimiters, 2 are for control, 3 are for hardware signaling, and 8 are unused (i.e., reserved for future versions of the protocol).

The advantage of this scheme is that it saves bandwidth, but the disadvantage is the loss of the self-clocking property of Manchester encoding. To compensate for this loss, a long preamble is used to synchronize the receiver to the sender's clock. Furthermore, all clocks are required to be stable to at least 0.005 percent. With this stability, frames up to 4500 bytes can be sent without danger of the receiver's clock drifting too far out of sync with the data stream.

The basic FDDI protocols are closely modeled on the 802.5 protocols. To transmit data, a station must first capture the token. Then it transmits a frame and removes it when it comes around again. One difference between FDDI and 802.5 is that in 802.5, a station may not generate a new token until its frame has gone all the way around and come back. In FDDI, with potentially 1000 stations and 200 km of fiber, the amount of time wasted waiting for the frame to circumnavigate the ring could be substantial. For this reason, it was decided to allow a station to put a new token back onto the ring as soon as it has finished transmitting its frames. In a large ring, several frames might be on the ring at the same time.

FDDI data frames are similar to 802.5 data frames. The FDDI format is shown in Fig. 4-46. The *Start delimiter* and *End delimiter* fields mark the frame boundaries. The *Frame control* field tells what kind of frame this is (data, control, etc.). The *Frame status* byte holds acknowledgement bits, similar to those of 802.5. The other fields are analogous to 802.5.

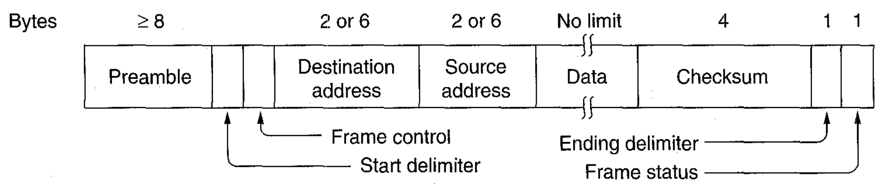

Fig. 4-46. FDDI frame format.

In addition to the regular (asynchronous) frames, FDDI also permits special synchronous frames for circuit-switched PCM or ISDN data. The synchronous frames are generated every 125 μsec by a master station to provide the 8000 samples/sec needed by PCM systems. Each of these frames has a header, 16 bytes of noncircuit-switched data, and up to 96 bytes of circuit-switched data (i.e., up to 96 PCM channels per frame).

The number 96 was chosen because it allows four T1 channels (4×24) at 1.544 Mbps or three CCITT E1 channels (3×32) at 2.048 Mbps to fit in a frame, thus making it suitable for use anywhere in the world. One synchronous frame every 125 μsec consumes 6.144 Mbps of bandwidth for the 96 circuit-switched channels. A maximum of 16 synchronous frames every 125 μsec allows up to 1536 PCM channels and eats up 98.3 Mbps.

Once a station has acquired one or more time slots in a synchronous frame, those slots are reserved for it until they are explicitly released. The total bandwidth not used by the synchronous frames is allocated on demand. A bit mask is present in each of these frames to indicate which slots are available for demand assignment. The nonsynchronous traffic is divided into priority classes, with the higher priorities getting first shot at the leftover bandwidth.

The FDDI MAC protocol uses three timers. The **token holding timer** determines how long a station may continue to transmit once it has acquired the token. This timer prevents a station from hogging the ring forever. The **token rotation timer** is restarted every time the token is seen. If this timer expires, it means that the token has not been sighted for too long an interval. Probably it has been lost, so the token recovery procedure is initiated. Finally, the **valid transmission timer** is used to time out and recover from certain transient ring errors.

FDDI also has a priority algorithm similar to 802.4. It determines which priority classes may transmit on a given token pass. If the token is ahead of schedule, all priorities may transmit, but if it is behind schedule, only the highest ones may send.

4.5.2. Fast Ethernet

FDDI was supposed to be the next generation LAN, but it never really caught on much beyond the backbone market (where it continues to do fine). The station management was too complicated, which led to complex chips and high prices. The substantial cost of FDDI chips made workstation manufacturers unwilling to make FDDI the standard network, so volume production never happened and FDDI never broke through to the mass market. The lesson that should have been learned here was KISS (Keep It Simple, Stupid).

In any event, the failure of FDDI to catch fire left a gap for a garden-variety LAN at speeds above 10 Mbps. Many installations needed more bandwidth and thus had numerous 10-Mbps LANs connected by a maze of repeaters, bridges, routers, and gateways, although to the network managers it sometimes felt that they were being held together by bubble gum and chicken wire.

It was in this environment that IEEE reconvened the 802.3 committee in 1992 with instructions to come up with a faster LAN. One proposal was to keep 802.3 exactly as it was, but just make it go faster. Another proposal was to redo it totally, to give it lots of new features, such as real-time traffic and digitized voice, but just keep the old name (for marketing reasons). After some wrangling, the committee decided to keep 802.3 the way it was, but just make it go faster. The people behind the losing proposal did what any computer-industry people would have done under these circumstances—they formed their own committee and standardized their LAN anyway (eventually as 802.12).

The three primary reasons that the 802.3 committee decided to go with a souped-up 802.3 LAN were:

1. The need to be backward compatible with thousands of existing LANs.
2. The fear that a new protocol might have unforeseen problems.
3. The desire to get the job done before the technology changed.

The work was done quickly (by standards committees' norms), and the result, **802.3u**, was officially approved by IEEE in June 1995. Technically, 802.3u is not a new standard, but an addendum to the existing 802.3 standard (to emphasize its backward compatibility). Since everyone calls it **fast Ethernet**, rather than 802.3u, we will do that, too.

The basic idea behind fast Ethernet was simple: keep all the old packet formats, interfaces, and procedural rules, but just reduce the bit time from 100 nsec to 10 nsec. Technically, it would have been possible to copy 10Base-5 or 10Base-2 and still detect collisions on time by just reducing the maximum cable length by a factor of ten. However, the advantages of 10Base-T wiring were so overwhelming, that fast Ethernet is based entirely on this design. Thus all fast Ethernet systems use hubs; multidrop cables with vampire taps or BNC connectors are not permitted.

Nevertheless, some choices still had to be made, the most important of which was which wire types to support. One contender was category 3 twisted pair. The argument for it was that practically every office in the Western world has at least four category 3 (or better) twisted pairs running from it to a telephone wiring closet within 100 meters. Sometimes two such cables exist. Thus using category 3 twisted pair would make it possible to wire up desktop computers using fast Ethernet without having to rewire the building, an enormous advantage for many organizations.

The main disadvantage of category 3 twisted pair is its inability to carry 200 megabaud signals (100 Mbps with Manchester encoding) 100 meters, the maximum computer-to-hub distance specified for 10Base-T (see Fig. 4-17). In contrast, category 5 twisted pair wiring can handle 100 meters easily, and fiber can go much further. The compromise chosen was to allow all three possibilities, as shown in Fig. 4-47, but to pep up the category 3 solution to give it the additional carrying capacity needed.

Name	Cable	Max. segment	Advantages
100Base-T4	Twisted pair	100 m	Uses category 3 UTP
100Base-TX	Twisted pair	100 m	Full duplex at 100 Mbps
100Base-FX	Fiber optics	2000 m	Full duplex at 100 Mbps; long runs

Fig. 4-47. Fast Ethernet cabling.

The category 3 UTP scheme, called **100Base-T4**, uses a signaling speed of 25 MHz, only 25 percent faster than standard 802.3's 20 MHz (remember that Manchester encoding, as shown in Fig. 4-20, requires two clock periods for each of the 10 million bits each second). To achieve the necessary bandwidth, 100Base-T4 requires four twisted pairs. Since standard telephone wiring for decades has had four twisted pairs per cable, most offices are able to handle this. Of course, it means giving up your office telephone, but that is surely a small price to pay for faster email.

Of the four twisted pairs, one is always to the hub, one is always from the hub, and other two are switchable to the current transmission direction. To get the necessary bandwidth, Manchester encoding is not used, but with modern clocks and such short distances, it is no longer needed. In addition, ternary signals are sent, so that during a single clock period the wire can contain a 0, a 1, or a 2. With three twisted pairs going in the forward direction and ternary signaling, any one of 27 possible symbols can be transmitted, making it possible to send 4 bits with some redundancy. Transmitting 4 bits in each of the 25 million clock cycles per second gives the necessary 100 Mbps. In addition, there is always a 33.3 Mbps reverse channel using the remaining twisted pair. This scheme, known as **8B6T**, (8 bits map to 6 trits) is not likely to win any prizes for elegance, but it works with the existing wiring plant.

For category 5 wiring, the design, **100Base-TX**, is simpler because the wires can handle clock rates up to 125 MHz and beyond. Only two twisted pairs per station are used, one to the hub and one from it. Rather than just use straight binary coding, a scheme called **4B5B** is used at 125 MHz. Every group of five clock periods is used to send 4 bits in order to give some redundancy, provide enough transitions to allow easy clock synchronization, create unique patterns for frame delimiting, and be compatible with FDDI in the physical layer. Consequently, 100Base-TX is a full-duplex system; stations can transmit at 100 Mbps and receive at 100 Mbps at the same time. In addition, you can have two telephones in your office for real communication in case the computer is fully occupied with surfing the Web.

The last option, **100Base-FX**, uses two strands of multimode fiber, one for each direction, so it, too, is full duplex with 100 Mbps in each direction. In addition, the distance between a station and the hub can be up to 2 km.

Two kinds of hubs are possible with 100Base-T4 and 100Base-TX, collectively known as **100Base-T**. In a shared hub, all the incoming lines (or at least all the lines arriving at one plug-in card) are logically connected, forming a single collision domain. All the standard rules, including the binary backoff algorithm, apply, so the system works just like old-fashioned 802.3. In particular, only one station at a time can be transmitting.

In a switched hub, each incoming frame is buffered on a plug-in line card. Although this feature makes the hub and cards more expensive, it also means that all stations can transmit (and receive) at the same time, greatly improving the total bandwidth of the system, often by an order of magnitude or more. Buffered frames are passed over a high-speed backplane from the source card to the destination card. The backplane has not been standardized, nor does it need to be, since it is entirely hidden deep inside the switch. If past experience is any guide, switch vendors will compete vigorously to produce ever faster backplanes in order to improve system throughput. Because 100Base-FX cables are too long for the normal Ethernet collision algorithm, they must be connected to buffered, switched hubs, so each one is a collision domain unto itself.

As a final note, virtually all switches can handle a mix of 10-Mbps and 100-Mbps stations, to make upgrading easier. As a site acquires more and more 100-Mbps workstations, all it has to do is buy the necessary number of new line cards and insert them into the switch.

More information about Fast Ethernet can be found in (Johnson, 1996). For a comparison of high-speed local area networks, in particular, FDDI, fast Ethernet, ATM, and VG-AnyLAN, see (Cronin et al., 1994).

4.5.3. HIPPI—High-Performance Parallel Interface

During the Cold War, Los Alamos National Laboratory, the U.S. government's nuclear weapons design center, routinely bought one of every supercomputer offered for sale. Los Alamos also collected fancy peripherals, such as massive storage devices and special graphics workstations for scientific visualization. At that time, each manufacturer had a different interface for connecting peripherals to its supercomputer, so it was not possible to share peripherals among machines or to connect two supercomputers together.

In 1987, researchers at Los Alamos began work on a standard supercomputer interface, with the intention of getting it standardized and then talking all the vendors into using it. (Given the size of Los Alamos' computing budget, when it talked, vendors listened.) The goal for the interface was an interface that everyone could implement quickly and efficiently. The guiding principle was KISS: Keep It Simple, Stupid. It was to have no options, not require any new chips to be designed, and have the performance of a fire hose.

The initial specification called for a data rate of 800 Mbps, because watching movies of bombs going off required frames of 1024×1024 pixels with 24 bits per pixel and 30 frames/sec, for an aggregate data rate of 750 Mbps. Later, one option crept in: a second data rate of 1600 Mbps. When this proposal, called **HIPPI** (**HIgh Performance Parallel Interface**) was later offered to ANSI for standardization, the proposers were regarded as the lunatic fringe because LANs in the 1980s meant 10-Mbps Ethernets.

HIPPI was originally designed to be a data channel rather than a LAN. Data channels operate point-to-point, from one master (a computer) to one slave (a peripheral), with dedicated wires and no switching. No contention is present and the environment is entirely predictable. Later, the need to be able to switch a peripheral from one supercomputer to another became apparent, and a crossbar switch was added to the HIPPI design, as illustrated in Fig. 4-48.

In order to achieve such enormous performance using only off-the-shelf chips, the basic interface was made 50 bits wide, 32 bits of data and 18 bits of control, so the HIPPI cable contains 50 twisted pairs. Every 40 nsec, a word is transferred in parallel across the interface. To achieve 1600 Mbps, two cables are used and two words are transferred per cycle. All transfers are simplex. To get two-way communication, two (or four) cables are needed. At these speeds, the maximum cable length is 25 meters.

After it got over some initial shock, the ANSI X3T9.3 committee produced a HIPPI standard based on the Los Alamos input. The standard covers the physical and data link layers. Everything above that is up to the users. The basic protocol is that to communicate, a host first asks the crossbar switch to set up a connection. Then it (usually) sends a single message and releases the connection.

Messages are structured with a control word, a header of up to 1016 bytes, and a data part of up to $2^{32} - 2$ bytes. For flow control reasons, messages are

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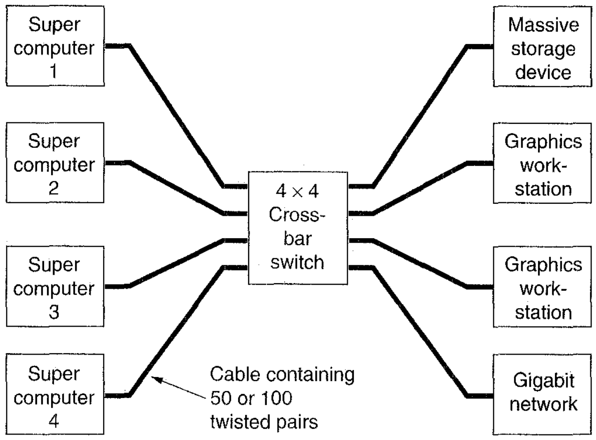

Fig. 4-48. HIPPI using a crossbar switch.

broken up into frames of 256 words. When the receiver is able to accept a frame, it signals the sender, which then sends a frame. Receivers can also ask for multiple frames at once. Error control consists of a horizontal parity bit per word and a vertical parity word at the end of each frame. Traditional checksums were regarded as unnecessary and too slow.

HIPPI was quickly implemented by dozens of vendors and has been the supercomputer interconnect standard for years. For more information about it, see (Hughes and Franta, 1994; Tolmie, 1992; and Tolmie and Renwick, 1993).

4.5.4. Fibre Channel

At the time HIPPI was designed, fiber optics was too expensive and not considered reliable enough, so the fastest LAN ever built was constructed from low-grade telephone wire. As time went on, fiber became cheaper and more reliable, so it was natural that there would eventually be an attempt to redo HIPPI using a single fiber instead of 50 or 100 twisted pairs. Unfortunately, the discipline that Los Alamos had in beating down proposed new features every time one reared its ugly head was lost along the way. The successor to HIPPI, called **fibre channel**, is far more complicated and more expensive to implement. Whether it will enjoy HIPPI's commercial success remains to be seen.

Fibre channel handles both data channel and network connections. In particular, it can be used to carry data channels including HIPPI, SCSI, and the multiplexor channel used on IBM mainframes. It can also carry network packets, including IEEE 802, IP, and ATM. Like HIPPI, the basic structure of fibre channel is a crossbar switch that connects inputs to outputs. Connections can be established for a single packet or for a much longer interval.

Fibre channel supports three service classes. The first class is pure circuit switching, with guaranteed delivery in order. The data channel modes use this service class. The second class is packet switching with guaranteed delivery. The third class is packet switching without guaranteed delivery.

Fibre channel has an elaborate protocol structure, as shown in Fig. 4-49. Here we see five layers, which together cover the physical and data link layers. The bottom layer deals with the physical medium. So far, it supports data rates of 100, 200, 400, and 800 Mbps. The second layer handles the bit encoding. The system used is somewhat like FDDI, but instead of 5 bits being used to encode 16 valid symbols, 10 bits are used to encode 256 valid symbols, providing a small amount of redundancy. Together, these two layers are functionally equivalent to the OSI physical layer.

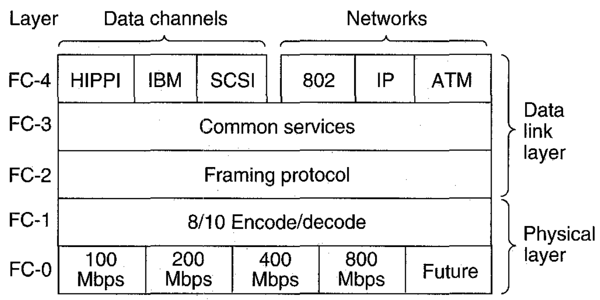

Fig. 4-49. The fibre channel protocol layers.

The middle layer defines the frame layout and header formats. Data are transmitted in frames whose payloads can be up to 2048 bytes. The next layer allows common services to be provided to the top layer in the future, as required. Finally, the top layer provides the interfaces to the various kinds of computers and peripherals supported.

As an aside, although fibre channel was designed in the United States, the spelling of the name was chosen by the editor of the standard, who was British. Additional information about fibre channel can be found in (Tolmie, 1992). A comparison of it with HIPPI and ATM is in (Tolmie, 1995).

4.6. SATELLITE NETWORKS

Although most multiple access channels are found in LANs, one kind of WAN also uses multiple access channels: communication satellite based WANs. In the following sections we will briefly study some of the problems that occur with satellite-based wide area networks. We will also look at some of the protocols that have been devised to deal with them.

Communication satellites generally have up to a dozen or so transponders. Each transponder has a beam that covers some portion of the earth below it, ranging from a wide beam 10,000 km across to a spot beam only 250 km across. Stations within the beam area can send frames to the satellite on the uplink frequency. The satellite then rebroadcasts them on the downlink frequency. Different frequencies are used for uplink and downlink to keep the transponder from going into oscillation. Satellites that do no on-board processing, but just echo whatever they hear (most of them), are often called **bent pipe** satellites.

Each antenna can aim itself at some area, transmit some frames, and then aim at a new area. Aiming is done electronically, but still takes some number of microseconds. The amount of time a beam is pointed to a given area is called the **dwell time**. For maximum efficiency, it should not be too short or too much time will be wasted moving the beam.

Just as with LANs, one of the key design issues is how to allocate the transponder channels. However, unlike LANs, carrier sensing is impossible due to the 270-msec propagation delay. When a station senses the state of a downlink channel, it hears what was going on 270 msec ago. Sensing the uplink channel is generally impossible. As a result, the CSMA/CD protocols (which assume that a transmitting station can detect collisions within the first few bit times, and then pull back if one is occurring) cannot be used with satellites. Hence the need for other protocols.

Five classes of protocols are used on the multiple access (uplink) channel: polling, ALOHA, FDM, TDM, and CDMA. Although we have studied each of these already, satellite operation sometimes adds new twists. The main problem is with the uplink channel, since the downlink channel has only a single sender (the satellite) and thus has no channel allocation problem.

4.6.1. Polling

The traditional way to allocate a single channel among competing users is for somebody to poll them. Having the satellite poll each station in turn to see if it has a frame is prohibitively expensive, given the 270-msec time required for each poll/response sequence.

However, if all the ground stations are also tied to a (typically low-bandwidth) packet-switching network, a minor variation of this idea is conceivable. The idea is to arrange all the stations in a logical ring, so each station knows its successor. Around this terrestrial ring circulates a token. The satellite never sees the token. A station is allowed to transmit on the uplink only when it has captured the token. If the number of stations is small and constant, the token transmission time is short, and the bursts sent on the uplink channel are much longer than the token rotation time, the scheme is moderately efficient.

4.6.2. ALOHA

Pure ALOHA is easy to implement: every station just sends whenever it wants to. The trouble is that the channel efficiency is only about 18 percent. Generally, such a low utilization factor is unacceptable for satellites that costs tens of millions of dollars each.

Using slotted ALOHA doubles the efficiency but introduces the problem of how to synchronize all the stations so they all know when each time slot begins. Fortunately, the satellite itself holds the answer, since it is inherently a broadcast medium. One ground station, the **reference station**, periodically transmits a special signal whose rebroadcast is used by all the ground stations as the time origin. If the time slots all have length ΔT, each station now knows that time slot k begins at a time $k\Delta T$ after the time origin. Since clocks run at slightly different rates, periodic resynchronization is necessary to keep everyone in phase. An additional complication is that the propagation time from the satellite is different for each ground station, but this effect can be corrected for.

To increase the utilization of the uplink channel above $1/e$, we could go from the single uplink channel of Fig. 4-50(a), to the dual uplink scheme of Fig. 4-50(b). A station with a frame to transmit chooses one of the two uplink channels at random and sends the frame in the next slot. Each uplink then operates an independent slotted ALOHA channel.

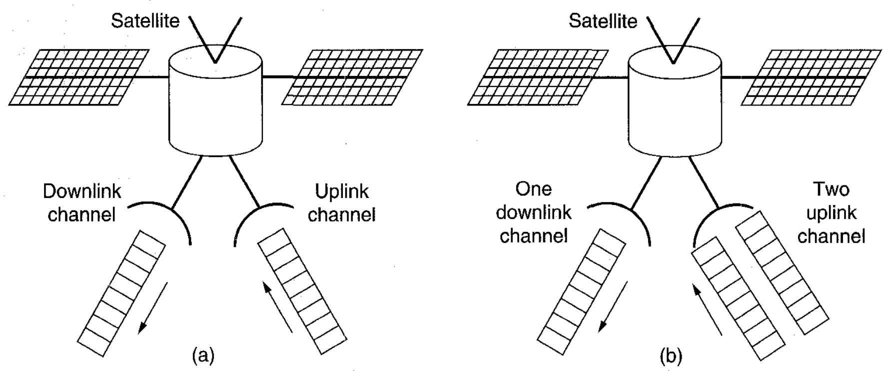

Fig. 4-50. (a) A standard ALOHA system. (b) Adding a second uplink channel.

If one of the uplink channels contains a single frame, it is just transmitted in the corresponding downlink slot later. If both channels are successful, the satellite can buffer one of the frames and transmit it during an idle slot later on. Working out the probabilities, it can be shown that given an infinite amount of buffer space, the downlink utilization can be gotten up to 0.736 at a cost of increasing the bandwidth requirements by one half.

4.6.3. FDM

Frequency division multiplexing is the oldest and probably still the most widely used channel allocation scheme. A typical 36-Mbps transponder might be divided statically into 500 or so 64,000-bps PCM channels, each one operating at its own unique frequency to avoid interfering with the others.

Although simple, FDM also has some drawbacks. First, guard bands are needed between the channels to keep the stations separated. This requirement exists because it is not possible to build transmitters that output all their energy in the main band and nothing in the side bands. The amount of bandwidth wasted in the guard bands can be a substantial fraction of the total.

Second, the stations must be carefully power controlled. If a station puts out too much power in the main band, it will also automatically put out too much power in the side bands, spilling over into adjacent channels and causing interference. Finally, FDM is entirely an analog technique and does not lend itself well to implementation in software.

If the number of stations is small and fixed, the frequency channels can be allocated statically in advance. However, if the number of stations, or the load on each one can fluctuate rapidly, some form of dynamic allocation of the frequency bands is needed. One such mechanism is the **SPADE** system used on some early Intelsat satellites. Each SPADE transponder was divided into 794 simplex (64-kbps) PCM voice channels, along with a 128-kbps common signaling channel. The PCM channels were used in pairs to provide full duplex service. The total transponder bandwidth used was 50 Mbps for the uplink portion and another 50 Mbps for the downlink.

The common signaling channel was divided into units of 50 msec. A unit contained 50 slots of 1 msec (128 bits). Each slot was "owned" by one of (not more than) 50 ground stations. When a ground station had data to send, it picked a currently unused channel at random and wrote the number of that channel in its next 128-bit slot. If the selected channel was still unused when the request was seen on the downlink, the channel was considered allocated and all other stations refrained from trying to acquire it. If two or more stations tried to allocate the same channel in the same frame, a collision occurred and they had to try again later. When a station was finished using its channel, it sent a deallocation message in its slot on the common channel.

4.6.4. TDM

Like FDM, TDM is well understood and widely used in practice. It requires time synchronization for the slots, but this can be provided by a reference station, as described for slotted ALOHA above. Similarly to FDM, for a small and unvarying number of stations, the slot assignment can be set up in advance and never changed, but for a varying number of stations, or a fixed number of stations with time-varying loads, time slots must be assigned dynamically.

Slot assignment can be done in a centralized or a decentralized way. As an example of centralized slot assignment, let us consider the experimental **ACTS** (**Advanced Communication Technology Satellite**), which was designed for a few dozen stations (Palmer and White, 1990). ACTS was launched in 1992 and has four independent 110-Mbps TDM channels, two uplink and two downlink. Each channel is organized as a sequence of 1-msec frames, each frame containing 1728 time slots. Each time slot has a 64-bit payload, allowing each one to hold a 64-kbps voice channel.

The beams can be switched from one geographical area to another, but since moving the beam takes several slot times, channels originating or terminating in the same geographic area are normally assigned to contiguous time slots to increase dwell time and minimize time lost to beam motion. Thus time slot management requires a thorough knowledge of station geography to minimize the number of wasted time slots. For this and other reasons, time slot management is done by one of the ground stations, the **MCS** (**Master Control Station**).

The basic operation of ACTS is a continuous three-step process, each step taking 1 msec. In step 1, the satellite receives a frame and stores it in a 1728-entry onboard RAM. In step 2, an onboard computer copies each input entry to the corresponding output entry (possibly for the other antenna). In step 3, the output frame is transmitted on the downlink.

Initially, each station is assigned at least one time slot. To acquire additional channels (for new voice calls), a station sends a short request message to the MCS. Similarly, it can release an existing channel with a message to the MCS. These messages make use of a small number of overhead bits and provide a special control channel to the MCS with a capacity of about 13 messages/sec per station. The channels are dedicated; there is no contention for them.

Dynamic TDM slot allocation is also possible. Below we will discuss three schemes. In each of these, TDM frames are divided into time slots, with each slot having a (temporary) owner. Only the owner may use a time slot.

The first scheme assumes that there are more slots than stations, so each station can be assigned a home slot (Binder, 1975). If there are more slots than stations, the extra slots are not assigned to anyone. If the owner of a slot does not want it during the current group, it goes idle. An empty slot is a signal to everyone else that the owner has no traffic. During the next frame, the slot becomes available to anyone who wants it, on a contention (ALOHA) basis.

If the owner wants to retrieve "his" home slot, he transmits a frame, thus forcing a collision (if there was other traffic). After a collision, everyone except the owner must desist from using the slot in the next frame. Thus the owner can always begin transmitting within two frame times in the worst case. At low channel utilization the system does not perform as well as normal slotted ALOHA, since after each collision, the collidees must abstain for one frame to see if the owner wants the slot back. Fig. 4-51(a) shows a frame with eight slots, seven of which are owned by *G*, *A*, *F*, *E*, *B*, *C*, and *D*, respectively. The eighth slot is not owned by anyone and can be fought over.

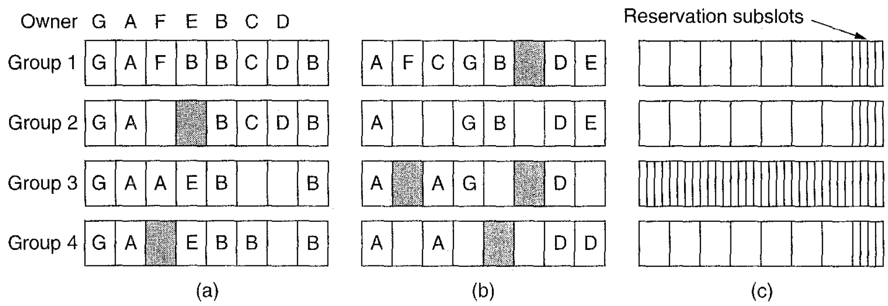

Fig. 4-51. Reservation schemes. (a) Binder. (b) Crowther. (c) Roberts. The shaded boxes indicate collisions. For each of the three schemes, four consecutive groups of slots are shown.

A second scheme is applicable even when the number of stations is unknown and varying (Crowther et al., 1973). In this method, slots do not have permanent owners, as in Binder's. Instead, stations compete for slots using slotted ALOHA. Whenever a transmission is successful, the station making the successful transmission is entitled to that slot in the next frame as well. Thus, as long as a station has data to send, it can continue doing so indefinitely (subject to some "Please-do-not-be-a-pig" rules). In essence the proposal allows a dynamic mix of slotted ALOHA and TDM, with the number of slots devoted to each varying with demand. Fig. 4-51(b) also shows a frame with eight slots. Initially, *E* is using the last slot, but after two frames, it no longer needs it. It lies idle for one frame, and then *D* picks it up and keeps it until it is done.

A third scheme, due to Roberts (1973), requires stations to make advance requests before transmitting. Each frame contains, say, one special slot [the last one in Fig. 4-51(c)] which is divided into *V* smaller subslots used to make reservations. When a station wants to send data, it broadcasts a short request frame in a randomly-chosen reservation subslot. If the reservation is successful (i.e., no collision), then the next regular slot (or slots) is reserved. At all times everyone must keep track of the queue length (number of slots reserved), so that when any station makes a successful reservation it will know how many data slots to skip before transmitting. Stations need not keep track of *who* is queued up; they merely need to know how long the queue is. When the queue length drops to zero, all slots revert to reservation subslots, to speed up the reservation process.

Although TDM is widely used, both with and without reservation schemes, it, too, has some shortcomings. For one, it requires all stations to synchronize in time, which is not entirely trivial in practice because satellites tend to drift in orbit, which changes the propagation time to each ground station. It also requires each ground station to be capable of extremely high burst speeds. For example, even though an ACTS station may have only one 64-kbps channel, it must be capable of putting out a 64-bit burst in a 578-nsec time slot. In other words, it must actually operate at 110 Mbps. In contrast, a 64-kbps FDM station really operates at 64 kbps.

4.6.5. CDMA

The final scheme is CDMA. CDMA avoids the time synchronization problem and also the channel allocation problem. It is completely decentralized and fully dynamic.

However, it has three main disadvantages. First, the capacity of a CDMA channel in the presence of noise and uncoordinated stations is typically lower than what TDM can achieve. Second, with 128 chips/bit (a common value), although the bit rate may not be high, the chip rate will be, necessitating a fast (read: expensive) transmitter. Third, few practicing engineers actually understand CDMA, which generally does not increase the chances of their using it, even if it is the best method for a particular application. Nevertheless, CDMA has been used by the military for decades and is now becoming more common in commercial applications as well.

4.7. SUMMARY

Some networks have a single channel that is used for all communication. In these networks, the key design issue is the allocation of this channel among the competing stations wishing to use it. Numerous channel allocation algorithms have been devised. A summary of some of the more important channel allocation methods is given in Fig. 4-52.

The simplest allocation schemes are FDM and TDM. These are efficient when the number of stations is small and the traffic is continuous. Both are widely used under these circumstances, for example, for dividing up the bandwidth in satellite links used as telephone trunks.

When the number of stations is large and variable or the traffic bursty, FDM and TDM are poor choices. The ALOHA protocol, with and without slotting and control, has been proposed as an alternative. ALOHA and its many variants and derivatives have been widely discussed, analyzed, and used in real systems.

When the state of the channel can be sensed, stations can avoid starting a transmission while another station is transmitting. This technique, carrier sensing, has led to a variety of protocols that can be used on LANs and MANs.

A class of protocols that eliminate contention altogether, or at least reduce it considerably, is known. Binary countdown completely eliminates contention.

Method	Description
FDM	Dedicate a frequency band to each station
TDM	Dedicate a time slot to each station
Pure ALOHA	Unsynchronized transmission at any instant
Slotted ALOHA	Random transmission in well-defined time slots
1-persistent CSMA	Standard carrier sense multiple access
Nonpersistent CSMA	Random delay when channel is sensed busy
P-persistent CSMA	CSMA, but with a probability of p of persisting
CSMA/CD	CSMA, but abort on detecting a collision
Bit map	Round robin scheduling using a bit map
Binary countdown	Highest numbered ready station goes next
Tree walk	Reduced contention by selective enabling
Wavelength division	A dynamic FDM scheme for fiber
MACA, MACAW	Wireless LAN protocols
GSM	FDM plus TDM for cellular radio
CDPD	Packet radio within an AMPS channel
CDMA	Everybody speak at once but in a different language
Ethernet	CSMA/CD with binary exponential backoff
Token bus	Logical ring on a physical bus
Token ring	Capture the token to send a frame
DQDB	Distributed queuing on a two-bus MAN
FDDI	Fiber-optic token ring
HIPPI	Crossbar using 50-100 twisted pairs
Fibre channel	Crossbar using fiber optics
SPADE	FDM with dynamic channel allocation
ACTS	TDM with centralized slot allocation
Binder	TDM with ALOHA when slot owner is not interested
Crowther	ALOHA with slot owner getting to keep it
Roberts	Channel time reserved in advance by ALOHA

Fig. 4-52. Channel allocation methods and systems for a common channel.

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The tree walk protocol reduces it by dynamically dividing the stations into two disjoint groups, one of which is permitted to transmit and one of which is not. It tries to make the division in such a way that only one station that is ready to send is permitted to do so.

Wireless LANs have their own problems and solutions. The biggest problem is caused by hidden stations, so CSMA does not work. One class of solutions, typified by MACA, attempts to stimulate transmissions around the destination, to make CSMA work better.

For mobile computers and telephones, cellular radio is the up-and-coming technology. GSM, CDPD, and CDMA are widely used.

The IEEE 802 LANs are: CSMA/CD, token bus, and token ring. Each of these has its own unique advantages and disadvantages, and each has found its own user community and will probably continue to serve that community for years to come. Convergence to a single LAN standard is an unlikely event. A new addition to this family is DQDB, being sold as a MAN in many cities.

An organization with multiple LANs often connects them with bridges. When a bridge connects two or more different kinds of LANs, new problems arise, some of them insoluble.

While the 802 LANs are the work horses of the day, the race horses are FDDI, fast Ethernet, HIPPI, and fibre channel. All of these offer bandwidth in the 100 Mbps range and up.

Finally, satellite networks also use multiple access channels (for the uplink). Various channel allocation methods are used here, including ALOHA, FDM, TDM, and CDMA.

PROBLEMS

1. A group of N stations share a 56-kbps pure ALOHA channel. Each station outputs a 1000-bit frame on an average of once every 100 sec, even if the previous one has not yet been sent (e.g., the stations are buffered). What is the maximum value of N?

2. Consider the delay of pure ALOHA versus slotted ALOHA at low load. Which one is less? Explain your answer.

3. Ten thousand airline reservation stations are competing for the use of a single slotted ALOHA channel. The average station makes 18 requests/hour. A slot is 125 μsec. What is the approximate total channel load?

4. A large population of ALOHA users manages to generate 50 requests/sec, including both originals and retransmissions. Time is slotted in units of 40 msec.
   (a) What is the chance of success on the first attempt?
   (b) What is the probability of exactly k collisions and then a success?
   (c) What is the expected number of transmission attempts needed?

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5. Measurements of a slotted ALOHA channel with an infinite number of users show that 10 percent of the slots are idle.
(a) What is the channel load, G?
(b) What is the throughput?
(c) Is the channel underloaded or overloaded?

6. In an infinite-population slotted ALOHA system, the mean number of slots a station waits between a collision and its retransmission is 4. Plot the delay versus throughput curve for this system.

7. A LAN uses Mok and Ward's version of binary countdown. At a certain instant, the ten stations have the virtual station numbers 8, 2, 4, 5, 1, 7, 3, 6, 9, and 0. The next three stations to send are 4, 3, and 9, in that order. What are the new virtual station numbers after all three have finished their transmissions?

8. Sixteen stations are contending for the use of a shared channel using the adaptive tree walk protocol. If all the stations whose addresses are prime numbers suddenly become ready at once, how many bit slots are needed to resolve the contention?

9. A collection of 2^n stations uses the adaptive tree walk protocol to arbitrate access to a shared cable. At a certain instant two of them become ready. What are the minimum, maximum, and mean number of slots to walk the tree if $2^n \gg 1$?

10. The wireless LANs that we studied used protocols such as MACA instead of CSMA/CD. Under what conditions would it be possible to use CSMA/CD instead?

11. What properties do the WDMA and GSM channel access protocols have in common?

12. Using the GSM framing structure as given in Fig. 4-14, determine how often any given user may send a data frame.

13. Suppose that A, B, and C are simultaneously transmitting 0 bits using a CDMA system with the chip sequences of Fig. 4-16(b). What is the resulting chip sequence?

14. In the discussion about orthogonality of CDMA chip sequences, it was stated that if $\mathbf{S} \bullet \mathbf{T} = 0$ then $\mathbf{S} \bullet \overline{\mathbf{T}}$ is also 0. Prove this.

15. Consider a different way of looking at the orthogonality property of CDMA chip sequences. Each bit in a pair of sequence can match or not match. Express the orthogonality property in terms of matches and mismatches.

16. A CDMA receiver gets the following chips: (−1 +1 −3 +1 −1 −3 +1 +1). Assuming the chip sequences defined in Fig. 4-16(b), which stations transmitted, and which bits did each one send?

17. A seven-story office building has 15 adjacent offices per floor. Each office contains a wall socket for a terminal in the front wall, so the sockets form a rectangular grid in the vertical plane, with a separation of 4 m between sockets, both horizontally and vertically. Assuming that it is feasible to run a straight cable between any pair of sockets, horizontally, vertically, or diagonally, how many meters of cable are needed to connect all sockets using
(a) a star configuration with a single router in middle?
(b) an 802.3 LAN?
(c) a ring net (without a wire center)?

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18. What is the baud rate of the standard 10-Mbps 802.3 LAN?

19. A 1-km-long, 10-Mbps CSMA/CD LAN (not 802.3) has a propagation speed of 200 m/μsec. Data frames are 256 bits long, including 32 bits of header, checksum, and other overhead. The first bit slot after a successful transmission is reserved for the receiver to capture the channel to send a 32-bit acknowledgement frame. What is the effective data rate, excluding overhead, assuming that there are no collisions?

20. Two CSMA/CD stations are each trying to transmit long (multiframe) files. After each frame is sent, they contend for the channel using the binary exponential backoff algorithm. What is the probability that the contention ends on round k, and what is the mean number of rounds per contention period?

21. Consider building a CSMA/CD network running at 1 Gbps over a 1-km cable with no repeaters. The signal speed in the cable is 200,000 km/sec. What is the minimum frame size?

22. Sketch the Manchester encoding for the bit stream: 0001110101.

23. Sketch the differential Manchester encoding for the bit stream of the previous problem. Assume the line is initially in the low state.

24. A token bus system works like this. When the token arrives at a station, a timer is reset to 0. The station then begins transmitting priority 6 frames until the timer reaches *T6*. Then it switches over to priority 4 frames until the timer reaches *T4*. This algorithm is then repeated with priority 2 and priority 0. If all stations have timer values of 40, 80, 90, and 100 msec for *T6* through *T0*, respectively, what fraction of the total bandwidth is reserved for each priority class?

25. What happens in a token bus if a station accepts the token and then crashes immediately? How does the protocol described in the text handle this case?

26. At a transmission rate of 5 Mbps and a propagation speed of 200 m/μsec, to how many meters of cable is the 1-bit delay in a token ring interface equivalent?

27. The delay around a token ring must be enough to contain the entire token. If the wire is not long enough, some artificial delay must be introduced. Explain why this extra delay is necessary in the content of a 24-bit token and a ring with only 16 bits of delay.

28. A very heavily loaded 1-km-long, 10-Mbps token ring has a propagation speed of 200 m/μsec. Fifty stations are uniformly spaced around the ring. Data frames are 256 bits, including 32 bits of overhead. Acknowledgements are piggybacked onto the data frames and are thus included as spare bits within the data frames and are effectively free. The token is 8 bits. Is the effective data rate of this ring higher or lower than the effective data rate of a 10-Mbps CSMA/CD network?

29. In a token ring the sender removes the frame. What modifications to the system would be needed to have the receiver remove the frame instead, and what would the consequences be?

30. A 4-Mbps token ring has a token-holding timer value of 10 msec. What is the longest frame that can be sent on this ring?

31. Does the use of a wire center have any influence on the performance of a token ring?

32. A fiber optic token ring used as a MAN is 200 km long and runs at 100 Mbps. After sending a frame, a station drains the frame from the ring before regenerating the token. The signal propagation speed in the fiber is 200,000 km/sec and the maximum frame size is 1K bytes. What is the maximum efficiency of the ring (ignoring all other sources of overhead)?

33. In Fig. 4-32, station *D* wants to send a cell. To which station does it want to send it?

34. The system of Fig. 4-32 is idle. A little later, stations *C*, *A*, and *B* become ready to send, in that order and in rapid succession. Assuming that no data frames are transmitted until all three have sent a request upstream, show the *RC* and *CD* values after each request and after the three data frames.

35. Ethernet is sometimes said to be inappropriate for real-time computing because the worst case retransmission interval is not bounded. Under what circumstances can the same argument be leveled at the token ring? Under what circumstances does the token ring have a known worst case? Assume the number of stations on the token ring is fixed and known.

36. Ethernet frames must be at least 64 bytes long to ensure that the transmitter is still going in the event of a collision at the far end of the cable. Fast Ethernet has the same 64 byte minimum frame size, but can get the bits out ten times faster. How is it possible to maintain the same minimum frame size?

37. Imagine two LAN bridges, both connecting a pair of 802.4 networks. The first bridge is faced with 1000 512-byte frames per second that must be forwarded. The second is faced with 200 4096-byte frames per second. Which bridge do you think will need the faster CPU? Discuss.

38. Suppose that the two bridges of the previous problem each connected an 802.4 LAN to an 802.5 LAN. Would that change have any influence on the previous answer?

39. A bridge between an 802.3 LAN and an 802.4 LAN has a problem with intermittent memory errors. Can this problem cause undetected errors with transmitted frames, or will these all be caught by the frame checksums?

40. A university computer science department has 3 Ethernet segments, connected by two transparent bridges into a linear network. One day the network administrator quits and is hastily replaced by someone from the computer center, which is an IBM token ring shop. The new administrator, noticing that the ends of the network are not connected, quickly orders a new transparent bridge and connects both loose ends to it, making a closed ring. What happens next?

41. A large FDDI ring has 100 stations and a token rotation time of 40 msec. The token-holding time is 10 msec. What is the maximum achievable efficiency of the ring?

42. Consider building a supercomputer interconnect using the HIPPI approach, but modern technology. The data path is now 64 bits wide, and a word can be sent every 10 nsec. What is the bandwidth of the channel?

43. In the text it was stated that a satellite with two uplink and one downlink slotted ALOHA channels can achieve a downlink utilization of 0.736, given an infinite amount of buffer space. Show how this result can be obtained.

5

THE NETWORK LAYER

The network layer is concerned with getting packets from the source all the way to the destination. Getting to the destination may require making many hops at intermediate routers along the way. This function clearly contrasts with that of the data link layer, which has the more modest goal of just moving frames from one end of a wire to the other. Thus the network layer is the lowest layer that deals with end-to-end transmission. For more information about it, see (Huitema, 1995; and Perlman, 1992).

To achieve its goals, the network layer must know about the topology of the communication subnet (i.e., the set of all routers) and choose appropriate paths through it. It must also take care to choose routes to avoid overloading some of the communication lines and routers while leaving others idle. Finally, when the source and destination are in different networks, it is up to the network layer to deal with these differences and solve the problems that result from them. In this chapter we will study all these issues and illustrate them with our two running examples, the Internet and ATM.

5.1. NETWORK LAYER DESIGN ISSUES

In the following sections we will provide an introduction to some of the issues that the designers of the network layer must grapple with. These issues include the service provided to the transport layer and the internal design of the subnet.

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5.1.1. Services Provided to the Transport Layer

The network layer provides services to the transport layer at the network layer/transport layer interface. This interface is often especially important for another reason: it frequently is the interface between the carrier and the customer, that is, the boundary of the subnet. The carrier often has control of the protocols and interfaces up to and including the network layer. Its job is to deliver packets given to it by its customers. For this reason, this interface must be especially well defined.

The network layer services have been designed with the following goals in mind.

1. The services should be independent of the subnet technology.
2. The transport layer should be shielded from the number, type, and topology of the subnets present.
3. The network addresses made available to the transport layer should use a uniform numbering plan, even across LANs and WANs.

Given these goals, the designers of the network layer have a lot of freedom in writing detailed specifications of the services to be offered to the transport layer. This freedom often degenerates into a raging battle between two warring factions. The discussion centers on the question of whether the network layer should provide connection-oriented service or connectionless service.

One camp (represented by the Internet community) argues that the subnet's job is moving bits around and nothing else. In their view (based on nearly 30 years of actual experience with a real, working computer network), the subnet is inherently unreliable, no matter how it is designed. Therefore, the hosts should accept the fact that it is unreliable and do error control (i.e., error detection and correction) and flow control themselves.

This viewpoint leads quickly to the conclusion that the network service should be connectionless, with primitives SEND PACKET and RECEIVE PACKET, and little else. In particular, no packet ordering and flow control should be done, because the hosts are going to do that anyway, and there is probably little to be gained by doing it twice. Furthermore, each packet must carry the full destination address, because each packet sent is carried independently of its predecessors, if any.

The other camp (represented by the telephone companies) argues that the subnet should provide a (reasonably) reliable, connection-oriented service. They claim 100 years of successful experience with the worldwide telephone system is a good guide. In this view, connections should have the following properties:

1. Before sending data, a network layer process on the sending side must set up a connection to its peer on the receiving side. This connection, which is given a special identifier, is then used until all the data have been sent, at which time it is explicitly released.

2. When a connection is set up, the two processes can enter into a negotiation about the parameters, quality, and cost of the service to be provided.

3. Communication is in both directions, and packets are delivered in sequence.

4. Flow control is provided automatically to prevent a fast sender from dumping packets into the pipe at a higher rate than the receiver can take them out, thus leading to overflow.

Other properties, such as guaranteed delivery, explicit confirmation of delivery, and high priority packets are optional. As we pointed out in Chap. 1, connectionless service is like the postal system, and connection-oriented service is like the telephone system.

The argument between connection-oriented and connectionless service really has to do with where to put the complexity. In the connection-oriented service, it is in the network layer (subnet); in the connectionless service, it is in the transport layer (hosts). Supporters of connectionless service say that user computing power has become cheap, so that there is no reason not to put the complexity in the hosts. Furthermore, they argue that the subnet is a major (inter)national investment that will last for decades, so it should not be cluttered up with features that may become obsolete quickly but will have to be calculated into the price structure for many years. Furthermore, some applications, such as digitized voice and real-time data collection may regard *speedy* delivery as much more important than *accurate* delivery.

On the other hand, supporters of connection-oriented service say that most users are not interested in running complex transport layer protocols in their machines. What they want is reliable, trouble-free service, and this service can be best provided with network layer connections. Furthermore, some services, such as real time audio and video are much easier to provide on top of a connection-oriented network layer than on top of a connectionless network layer.

Although it is rarely discussed in these terms, two separate issues are involved here. First, whether the network is connection-oriented (setup required) or connectionless (no setup required). Second, whether it is reliable (no lost, duplicated, or garbled packets) or unreliable (packets can be lost, duplicated, or garbled). In theory, all four combinations exist, but the dominant combinations are reliable connection-oriented and unreliable connectionless, so the other two tend to get lost in the noise.

These two camps are represented by our two running examples. The Internet has a connectionless network layer, and ATM networks have a connection-oriented network layer. An obvious question arises about how the Internet works when it runs over an ATM-based, carrier-provided subnet. The answer is that the source host first establishes an ATM network layer connection to the destination host and then sends independent (IP) packets over it, as shown in Fig. 5-1. Although this approach works, it is inefficient because certain functionality is in both layers. For example, the ATM network layer guarantees that packets are always delivered in order, but the TCP code still contains the full mechanism for managing and reordering out-of-order packets. For more information about how to run IP over ATM, see RFC 1577 and (Armitage and Adams, 1995).

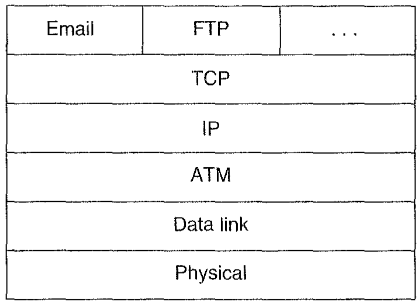

Fig. 5-1. Running TCP/IP over an ATM subnet.

5.1.2. Internal Organization of the Network Layer

Having looked at the two classes of service the network layer can provide to its users, it is time to see how it works inside. There are basically two different philosophies for organizing the subnet, one using connections and the other working connectionless. In the context of the *internal* operation of the subnet, a connection is usually called a **virtual circuit**, in analogy with the physical circuits set up by the telephone system. The independent packets of the connectionless organization are called **datagrams**, in analogy with telegrams.

Virtual circuits are generally used in subnets whose primary service is connection-oriented, so we will describe them in that context. The idea behind virtual circuits is to avoid having to choose a new route for every packet or cell sent. Instead, when a connection is established, a route from the source machine to the destination machine is chosen as part of the connection setup and remembered. That route is used for all traffic flowing over the connection, exactly the same way that the telephone system works. When the connection is released, the virtual circuit is also terminated.

In contrast, with a datagram subnet no routes are worked out in advance, even if the service is connection-oriented. Each packet sent is routed independently of its predecessors. Successive packets may follow different routes. While datagram subnets have to do more work, they are also generally more robust and adapt to failures and congestion more easily than virtual circuit subnets. We will discuss the pros and cons of the two approaches later.

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If packets flowing over a given virtual circuit always take the same route through the subnet, each router must remember where to forward packets for each of the currently open virtual circuits passing through it. Every router must maintain a table with one entry per open virtual circuit passing through it. Each packet traveling through the subnet must contain a virtual circuit number field in its header, in addition to sequence numbers, checksums, and the like. When a packet arrives at a router, the router knows on which line it arrived and what the virtual circuit number is. Based on only this information, the packet must be forwarded on the correct output line.

When a network connection is set up, a virtual circuit number not already in use on that machine is chosen as the connection identifier. Since each machine chooses virtual circuit numbers independently, these numbers have only local significance. If they were globally significant over the whole network, it is likely that two virtual circuits bearing the same global virtual circuit number might pass through some intermediate router, leading to ambiguities.

Because virtual circuits can be initiated from either end, a problem occurs when call setups are propagating in both directions at once along a chain of routers. At some point they have arrived at adjacent routers. Each router must now pick a virtual circuit number to use for the (full-duplex) circuit it is trying to establish. If they have been programmed to choose the lowest number not already in use on the link, they will pick the same number, causing two unrelated virtual circuits over the same physical line to have the same number. When a data packet arrives later, the receiving router has no way of telling whether it is a forward packet on one circuit or a reverse packet on the other. If circuits are simplex, there is no ambiguity.

Note that every process must be required to indicate when it is through using a virtual circuit, so that the virtual circuit can be purged from the router tables to recover the space. In public networks, the motivation is the stick rather than the carrot: users are invariably charged for connect time as well as for data transported. In addition, some provision must be made for dealing with machines that terminate their virtual circuits by crashing rather than politely releasing them when done.

So much for the use of virtual circuits internal to the subnet. The other possibility is to use datagrams internally, in which case the routers do not have a table with one entry for each open virtual circuit. Instead, they have a table telling which outgoing line to use for each possible destination router. These tables are also needed when virtual circuits are used internally, to determine the route for a setup packet.

Each datagram must contain the full destination address. For a large network, these addresses can be quite long (e.g., a dozen bytes or more). When a packet comes in, the router looks up the outgoing line to use and sends the packet on its way. Also, the establishment and release of network or transport layer connections do not require any special work on the part of the routers.

5.1.3. Comparison of Virtual Circuit and Datagram Subnets

Both virtual circuits and datagrams have their supporters and their detractors. We will now attempt to summarize the arguments both ways. The major issues are listed in Fig. 5-2, although purists could probably find a counterexample for everything in the figure.

Issue	Datagram subnet	VC subnet
Circuit setup	Not needed	Required
Addressing	Each packet contains the full source and destination address	Each packet contains a short VC number
State information	Subnet does not hold state information	Each VC requires subnet table space
Routing	Each packet is routed independently	Route chosen when VC is set up; all packets follow this route
Effect of router failures	None, except for packets lost during the crash	All VCs that passed through the failed router are terminated
Congestion control	Difficult	Easy if enough buffers can be allocated in advance for each VC

Fig. 5-2. Comparison of datagram and virtual circuit subnets.

Inside the subnet, several trade-offs exist between virtual circuits and datagrams. One trade-off is between router memory space and bandwidth. Virtual circuits allow packets to contain circuit numbers instead of full destination addresses. If the packets tend to be fairly short, a full destination address in every packet may represent a significant amount of overhead, and hence wasted bandwidth. The price paid for using virtual circuits internally is the table space within the routers. Depending upon the relative cost of communication circuits versus router memory, one or the other may be cheaper.

Another trade-off is setup time versus address parsing time. Using virtual circuits requires a setup phase, which takes time and consumes resources. However, figuring out what to do with a data packet in a virtual circuit subnet is easy: the router just uses the circuit number to index into a table to find out where the packet goes. In a datagram subnet, a more complicated procedure is required to determine where the packet goes.

Virtual circuits have some advantages in avoiding congestion within the subnet because resources can be reserved in advance, when the connection is established. Once the packets start arriving, the necessary bandwidth and router capacity will be there. With a datagram subnet, congestion avoidance is more difficult.

For transaction processing systems (e.g., stores calling up to verify credit card purchases), the overhead required to set up and clear a virtual circuit may easily dwarf the use of the circuit. If the majority of the traffic is expected to be of this kind, the use of switched virtual circuits inside the subnet makes little sense. On the other hand, permanent virtual circuits, which are set up manually and last for months or years, may be useful here.

Virtual circuits also have a vulnerability problem. If a router crashes and loses its memory, even if it comes back up a second later, all the virtual circuits passing through it will have to be aborted. In contrast, if a datagram router goes down, only those users whose packets were queued up in the router at the time will suffer, and maybe not even all those, depending upon whether they have already been acknowledged or not. The loss of a communication line is fatal to virtual circuits using it but can be easily compensated for if datagrams are used. Datagrams also allow the routers to balance the traffic throughout the subnet, since routes can be changed halfway through a connection.

It is worth explicitly pointing out that the service offered (connection-oriented or connectionless) is a separate issue from the subnet structure (virtual circuit or datagram). In theory, all four combinations are possible. Obviously, a virtual circuit implementation of a connection-oriented service and a datagram implementation of a connectionless service are reasonable. Implementing connections using datagrams also makes sense when the subnet is trying to provide a highly robust service.

The fourth possibility, a connectionless service on top of a virtual circuit subnet, seems strange but certainly occurs. The obvious example is running IP over an ATM subnet. Here it is desired to run an existing connectionless protocol over a new connection-oriented network layer. As mentioned earlier, this is more of an ad hoc solution to a problem than a good design. In a new system designed to run over an ATM subnet, one would not normally put a connectionless protocol like IP over a connection-oriented network layer like ATM and then layer a connection-oriented transport protocol on top of the connectionless protocol. Examples of all four cases are shown in Fig. 5-3.

5.2. ROUTING ALGORITHMS

The main function of the network layer is routing packets from the source machine to the destination machine. In most subnets, packets will require multiple hops to make the journey. The only notable exception is for broadcast

Upper layer	Type of subnet: Datagram	Type of subnet: Virtual circuit
Connectionless	UDP over IP	UDP over IP over ATM
Connection-oriented	TCP over IP	ATM AAL1 over ATM

Fig. 5-3. Examples of different combinations of service and subnet structure.

networks, but even here routing is an issue if the source and destination are not on the same network. The algorithms that choose the routes and the data structures that they use are a major area of network layer design.

The **routing algorithm** is that part of the network layer software responsible for deciding which output line an incoming packet should be transmitted on. If the subnet uses datagrams internally, this decision must be made anew for every arriving data packet since the best route may have changed since last time. If the subnet uses virtual circuits internally, routing decisions are made only when a new virtual circuit is being set up. Thereafter, data packets just follow the previously established route. The latter case is sometimes called **session routing**, because a route remains in force for an entire user session (e.g., a login session at a terminal or a file transfer).

Regardless of whether routes are chosen independently for each packet or only when new connections are established, there are certain properties that are desirable in a routing algorithm: correctness, simplicity, robustness, stability, fairness, and optimality. Correctness and simplicity hardly require comment, but the need for robustness may be less obvious at first. Once a major network comes on the air, it may be expected to run continuously for years without systemwide failures. During that period there will be hardware and software failures of all kinds. Hosts, routers, and lines will go up and down repeatedly, and the topology will change many times. The routing algorithm should be able to cope with changes in the topology and traffic without requiring all jobs in all hosts to be aborted and the network to be rebooted every time some router crashes.

Stability is also an important goal for the routing algorithm. There exist routing algorithms that never converge to equilibrium, no matter how long they run. Fairness and optimality may sound obvious—surely no one would oppose them—but as it turns out, they are often contradictory goals. As a simple example of this conflict, look at Fig. 5-4. Suppose that there is enough traffic between A and A', between B and B', and between C and C' to saturate the horizontal links. To maximize the total flow, the X to X' traffic should be shut off altogether.

Unfortunately, X and X' may not see it that way. Evidently, some compromise between global efficiency and fairness to individual connections is needed.

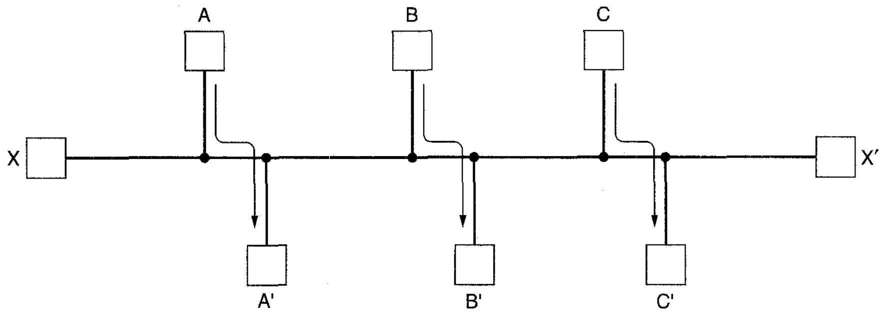


Fig. 5-4. Conflict between fairness and optimality.

Before we can even attempt to find trade-offs between fairness and optimality, we must decide what it is we seek to optimize. Minimizing mean packet delay is an obvious candidate, but so is maximizing total network throughput. Furthermore, these two goals are also in conflict, since operating any queueing system near capacity implies a long queueing delay. As a compromise, many networks attempt to minimize the number of hops a packet must make, because reducing the number of hops tends to improve the delay and also reduce the amount of bandwidth consumed, which tends to improve the throughput as well.

Routing algorithms can be grouped into two major classes: nonadaptive and adaptive. **Nonadaptive algorithms** do not base their routing decisions on measurements or estimates of the current traffic and topology. Instead, the choice of the route to use to get from I to J (for all I and J) is computed in advance, off-line, and downloaded to the routers when the network is booted. This procedure is sometimes called **static routing**.

Adaptive algorithms, in contrast, change their routing decisions to reflect changes in the topology, and usually the traffic as well. Adaptive algorithms differ in where they get their information (e.g., locally, from adjacent routers, or from all routers), when they change the routes (e.g., every ΔT sec, when the load changes, or when the topology changes), and what metric is used for optimization (e.g., distance, number of hops, or estimated transit time). In the following sections we will discuss a variety of routing algorithms, both static and dynamic.

5.2.1. The Optimality Principle

Before getting into specific algorithms, it may be helpful to note that one can make a general statement about optimal routes without regard to network topology or traffic. This statement is known as the **optimality principle**. It states that if router J is on the optimal path from router I to router K, then the optimal path from J to K also falls along the same route. To see this, call the part of the route from I to J r_1 and the rest of the route r_2. If a route better than r_2 existed from J to K, it could be concatenated with r_1 to improve the route from I to K, contradicting our statement that $r_1 r_2$ is optimal.

As a direct consequence of the optimality principle, we can see that the set of optimal routes from all sources to a given destination form a tree rooted at the destination. Such a tree is called a **sink tree** and is illustrated in Fig. 5-5 where the distance metric is the number of hops. Note that a sink tree is not necessarily unique; other trees with the same path lengths may exist. The goal of all routing algorithms is to discover and use the sink trees for all routers.

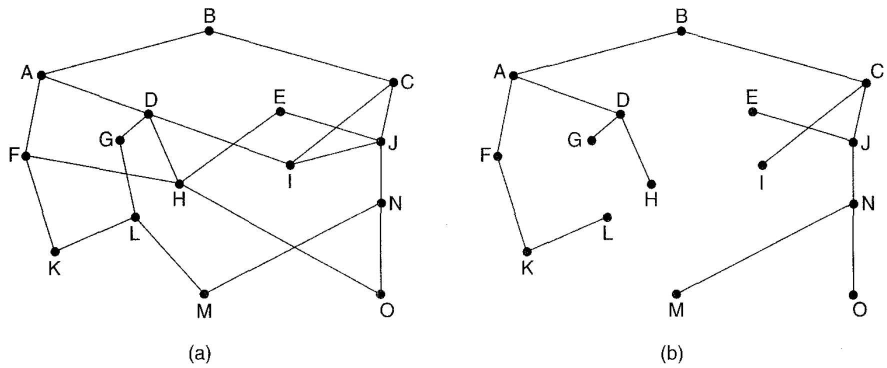


Fig. 5-5. (a) A subnet. (b) A sink tree for router B.

Since a sink tree is indeed a tree, it does not contain any loops, so each packet will be delivered within a finite and bounded number of hops. In practice, life is not quite this easy. Links and routers can go down and come back up during operation, so different routers may have different ideas about the current topology. Also, we have quietly finessed the issue of whether each router has to individually acquire the information on which to base its sink tree computation, or whether this information is collected by some other means. We will come back to these issues shortly. Nevertheless, the optimality principle and the sink tree provide a benchmark against which other routing algorithms can be measured.

In the next three sections, we will look at three different static routing algorithms. After that we will move on to adaptive ones.

5.2.2. Shortest Path Routing

Let us begin our study of routing algorithms with a technique that is widely used in many forms because it is simple and easy to understand. The idea is to build a graph of the subnet, with each node of the graph representing a router and each arc of the graph representing a communication line (often called a link). To choose a route between a given pair of routers, the algorithm just finds the shortest path between them on the graph.

The concept of a **shortest path** deserves some explanation. One way of measuring path length is the number of hops. Using this metric, the paths *ABC* and *ABE* in Fig. 5-6 are equally long. Another metric is the geographic distance in kilometers, in which case *ABC* is clearly much longer than *ABE* (assuming the figure is drawn to scale).

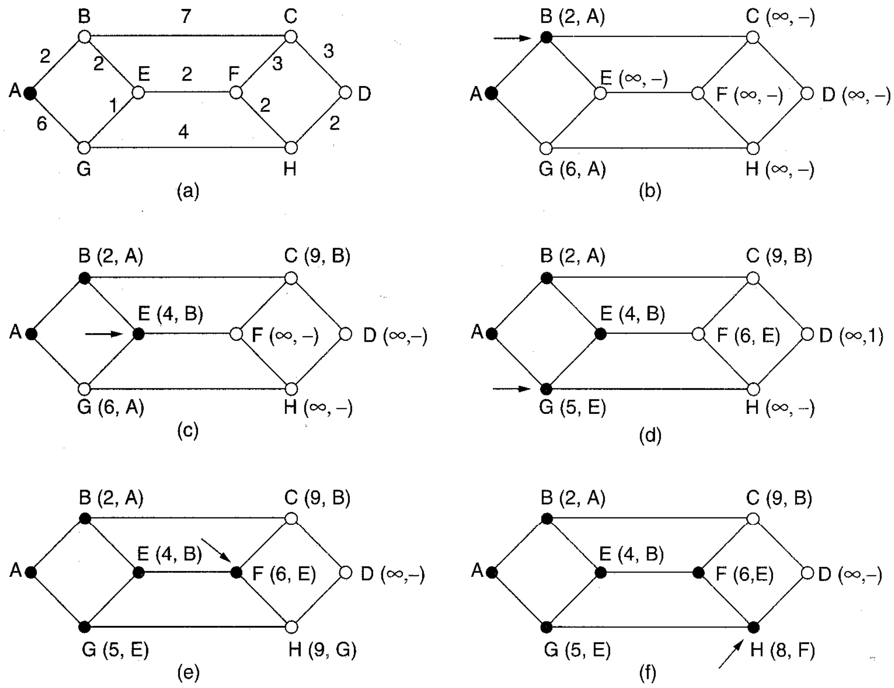


Fig. 5-6. The first five steps used in computing the shortest path from *A* to *D*. The arrows indicate the working node.

However, many other metrics are also possible besides hops and physical distance. For example, each arc could be labeled with the mean queueing and transmission delay for some standard test packet as determined by hourly test runs. With this graph labeling, the shortest path is the fastest path, rather than the path with the fewest arcs or kilometers.

In the most general case, the labels on the arcs could be computed as a function of the distance, bandwidth, average traffic, communication cost, mean queue length, measured delay, and other factors. By changing the weighting function, the algorithm would then compute the "shortest" path measured according to any one of a number of criteria, or a combination of criteria.

Several algorithms for computing the shortest path between two nodes of a graph are known. This one is due to Dijkstra (1959). Each node is labeled (in parentheses) with its distance from the source node along the best known path. Initially, no paths are known, so all nodes are labeled with infinity. As the algorithm proceeds and paths are found, the labels may change, reflecting better paths. A label may be either tentative or permanent. Initially, all labels are tentative. When it is discovered that a label represents the shortest possible path from the source to that node, it is made permanent and never changed thereafter.

To illustrate how the labeling algorithm works, look at the weighted, undirected graph of Fig. 5-6(a), where the weights represent, for example, distance. We want to find the shortest path from *A* to *D*. We start out by marking node *A* as permanent, indicated by a filled in circle. Then we examine, in turn, each of the nodes adjacent to *A* (the working node), relabeling each one with the distance to *A*. Whenever a node is relabeled, we also label it with the node from which the probe was made, so we can reconstruct the final path later. Having examined each of the nodes adjacent to *A*, we examine all the tentatively labeled nodes in the whole graph and make the one with the smallest label permanent, as shown in Fig. 5-6(b). This one becomes the new working node.

We now start at *B*, and examine all nodes adjacent to it. If the sum of the label on *B* and the distance from *B* to the node being considered is less than the label on that node, we have a shorter path, so the node is relabeled.

After all the nodes adjacent to the working node have been inspected and the tentative labels changed if possible, the entire graph is searched for the tentatively labeled node with the smallest value. This node is made permanent and becomes the working node for the next round. Figure 5-6 shows the first five steps of the algorithm.

To see why the algorithm works, look at Fig. 5-6(c). At that point we have just made *E* permanent. Suppose that there were a shorter path than *ABE*, say *AXYZE*. There are two possibilities: either node *Z* has already been made permanent, or it has not been. If it has, then *E* has already been probed (on the round following the one when *Z* was made permanent), so the *AXYZE* path has not escaped our attention.

Now consider the case where *Z* is still tentatively labeled. Either the label at *Z* is greater than or equal to that at *E*, in which case *AXYZE* cannot be a shorter path than *ABE*, or it is less than that of *E*, in which case *Z* and not *E* will become permanent first, allowing *E* to be probed from *Z*.

This algorithm is given in Fig. 5-7. The only difference between the program and the algorithm described above is that in Fig. 5-7, we compute the shortest path starting at the terminal node, *t*, rather than at the source node, *s*. Since the shortest path from *t* to *s* in an undirected graph is the same as the shortest path from *s* to *t*, it does not matter at which end we begin (unless there are several shortest paths, in which case reversing the search might discover a different one). The reason for searching backward is that each node is labeled with its predecessor rather than its successor. When copying the final path into the output variable, *path*, the path is thus reversed. By reversing the search, the two effects cancel, and the answer is produced in the correct order.

5.2.3. Flooding

Another static algorithm is **flooding**, in which every incoming packet is sent out on every outgoing line except the one it arrived on. Flooding obviously generates vast numbers of duplicate packets, in fact, an infinite number unless some measures are taken to damp the process. One such measure is to have a hop counter contained in the header of each packet, which is decremented at each hop, with the packet being discarded when the counter reaches zero. Ideally, the hop counter should be initialized to the length of the path from source to destination. If the sender does not know how long the path is, it can initialize the counter to the worst case, namely, the full diameter of the subnet.

An alternative technique for damming the flood is to keep track of which packets have been flooded, to avoid sending them out a second time. One way to achieve this goal is to have the source router put a sequence number in each packet it receives from its hosts. Each router then needs a list per source router telling which sequence numbers originating at that source have already been seen. If an incoming packet is on the list, it is not flooded.

To prevent the list from growing without bound, each list should be augmented by a counter, k, meaning that all sequence numbers through k have been seen. When a packet comes in, it is easy to check if the packet is a duplicate; if so, it is discarded. Furthermore, the full list below k is not needed, since k effectively summarizes it.

A variation of flooding that is slightly more practical is **selective flooding**. In this algorithm the routers do not send every incoming packet out on every line, only on those lines that are going approximately in the right direction. There is usually little point in sending a westbound packet on an eastbound line unless the topology is extremely peculiar.

Flooding is not practical in most applications, but it does have some uses. For example, in military applications, where large numbers of routers may be blown to bits at any instant, the tremendous robustness of flooding is highly desirable. In distributed database applications, it is sometimes necessary to update all the databases concurrently, in which case flooding can be useful. A third possible use of flooding is as a metric against which other routing algorithms can be compared. Flooding always chooses the shortest path, because it chooses every possible path in parallel. Consequently, no other algorithm can produce a shorter delay (if we ignore the overhead generated by the flooding process itself).

```
#define MAX_NODES 1024                  /* maximum number of nodes */
#define INFINITY 1000000000             /* a number larger than every maximum path */
int n, dist[MAX_NODES][MAX_NODES];/* dist[i][j] is the distance from i to j */

void shortest_path(int s, int t, int path[])
{ struct state {                        /* the path being worked on */
      int predecessor;                  /* previous node */
      int length;                       /* length from source to this node */
      enum {permanent, tentative} label; /* label state */
  } state[MAX_NODES];

  int i, k, min;
  struct state *
                p;
  for (p = &state[0]; p < &state[n]; p++) { /* initialize state */
      p->predecessor = -1;
      p->length = INFINITY;
      p->label = tentative;
  }
  state[t].length = 0;  state[t].label = permanent;
  k = t;                                /* k is the initial working node */
  do {                                  /* Is there a better path from k? */
      for (i = 0; i < n; i++)           /* this graph has n nodes */
           if (dist[k][i] != 0 && state[i].label == tentative) {
                 if (state[k].length + dist[k][i] < state[i].length) {
                      state[i].predecessor = k;
                      state[i].length = state[k].length + dist[k][i];
                 }
           }

      /* Find the tentatively labeled node with the smallest label. */
      k = 0; min = INFINITY;
      for (i = 0; i < n; i++)
           if (state[i].label == tentative && state[i].length < min) {
                 min = state[i].length;
                 k = i;
           }
      state[k].label = permanent;
  } while (k != s);

  /* Copy the path into the output array. */
  i = 0;  k = s;
  do {path[i++] = k; k = state[k].predecessor; } while (k >= 0);
}
```

Fig. 5-7. Dijkstra's algorithm to compute the shortest path through a graph.

5.2.4. Flow-Based Routing

The algorithms studied so far take only the topology into account. They do not consider the load. If, for example, there is always a huge amount of traffic from A to B, in Fig. 5-6, then it may be better to route traffic from A to C via $AGEFC$, even though this path is much longer than ABC. In this section we will study a static algorithm that uses both topology and load for routing. It is called **flow-based routing**.

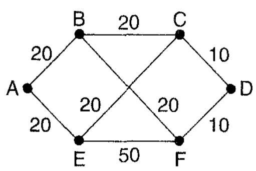
In some networks, the mean data flow between each pair of nodes is relatively stable and predictable. For example, in a corporate network for a retail store chain, each store might send orders, sales reports, inventory updates, and other well-defined types of messages to known sites in a predefined pattern, so that the total volume of traffic varies little from day to day. Under conditions in which the average traffic from i to j is known in advance and, to a reasonable approximation, constant in time, it is possible to analyze the flows mathematically to optimize the routing.

The basic idea behind the analysis is that for a given line, if the capacity and average flow are known, it is possible to compute the mean packet delay on that line from queueing theory. From the mean delays on all the lines, it is straightforward to calculate a flow-weighted average to get the mean packet delay for the whole subnet. The routing problem then reduces to finding the routing algorithm that produces the minimum average delay for the subnet. Fig. 5-8.

To use this technique, certain information must be known in advance. First the subnet topology must be known. Second, the traffic matrix, F_{ij}, must be given. Third, the line capacity matrix, C_{ij}, specifying the capacity of each line in bps must be available. Finally, a (possibly tentative) routing algorithm must be chosen.

As an example of this method, consider the full-duplex subnet of Fig. 5-8(a). The weights on the arcs give the capacities, C_{ij}, in each direction measured in kbps. The matrix of Fig. 5-8(b) has an entry for each source-destination pair. The entry for source i to destination j shows the route to be used for i-j traffic, and also the number of packets/sec to be sent from source i to destination j. For example, 3 packets/sec go from B to D, and they use route BFD to get there. Notice that some routing algorithm has already been applied to derive the routes shown in the matrix.

Given this information, it is straightforward to calculate the total in line i, λ_i. For example, the B-D traffic contributes 3 packets/sec to the BF line and also 3 packets/sec to the FD line. Similarly, the A-D traffic contributes 1 packet/sec to each of three lines. The total traffic in each eastbound line is shown in the λ_i column of Fig. 5-9. In this example, all the traffic is symmetric, that is, the XY traffic is identical to the YX traffic, for all X and Y. In real networks this condition does not always hold. The figure also shows the mean number of packets/sec on each line, μC_i assuming a mean packet size of $1/\mu = 800$ bits.



(a)

Destination

Source	A	B	C	D	E	F
A		9 AB	4 ABC	1 ABFD	7 AE	4 AEF
B	9 BA		8 BC	3 BFD	2 BFE	4 BF
C	4 CBA	8 CB		3 CD	3 CE	2 CEF
D	1 DFBA	3 DFB	3 DC		3 DCE	4 DF
E	7 EA	2 EFB	3 EC	3 ECD		5 EF
F	4 FEA	4 FB	2 FEC	4 FD	5 FE	

(b)

Fig. 5-8. (a) A subnet with line capacities shown in kbps. (b) The traffic in packets/sec and the routing matrix.

The next-to-last column of Fig. 5-9 gives the mean delay for each line derived from the queueing theory formula

$$T = \frac{1}{\mu C - \lambda}$$

where $1/\mu$ is the mean packet size in bits, C is the capacity in bps, and λ is the mean flow in packets/sec. For example, with a capacity $\mu C = 25$ packets/sec and an actual flow $\lambda = 14$ packets/sec, the mean delay is 91 msec. Note that with $\lambda = 0$, the mean delay is still 40 msec, because the capacity is 25 packets/sec. In other words, the "delay" includes both queueing and service time.

To compute the mean delay time for the entire subnet, we take the weighted sum of each of the eight lines, with the weight being the fraction of the total traffic using that line. In this example, the mean turns out to be 86 msec.

To evaluate a different routing algorithm, we can repeat the entire process, only with different flows to get a new average delay. If we restrict ourselves to only single path routing algorithms, as we have done so far, there are only a finite number of ways to route packets from each source to each destination. It is always possible to write a program to simply try them all, one after another, and find out which one has the smallest mean delay. Since this calculation can be done off-line in advance, the fact that it may be time consuming is not necessarily a serious problem. This one is then the best routing algorithm. Bertsekas and Gallager (1992) discuss flow-based routing in detail.

i	Line	λ_i (pkts/sec)	C_i (kbps)	μC_i (pkts/sec)	T_i (msec)	Weight
1	AB	14	20	25	91	0.171
2	BC	12	20	25	77	0.146
3	CD	6	10	12.5	154	0.073
4	AE	11	20	25	71	0.134
5	EF	13	50	62.5	20	0.159
6	FD	8	10	12.5	222	0.098
7	BF	10	20	25	67	0.122
8	EC	8	20	25	59	0.098

Fig. 5-9. Analysis of the subnet of Fig. 5-8 using a mean packet size of 800 bits. The reverse traffic (*BA*, *CB*, etc.) is the same as the forward traffic.

5.2.5. Distance Vector Routing

Modern computer networks generally use dynamic routing algorithms rather than the static ones described above. Two dynamic algorithms in particular, distance vector routing and link state routing, are the most popular. In this section we will look at the former algorithm. In the following one we will study the latter one.

Distance vector routing algorithms operate by having each router maintain a table (i.e, a vector) giving the best known distance to each destination and which line to use to get there. These tables are updated by exchanging information with the neighbors.

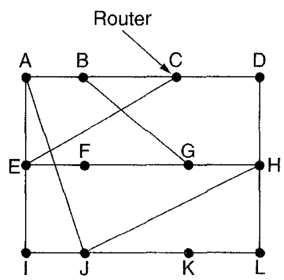
The distance vector routing algorithm is sometimes called by other names, including the distributed **Bellman-Ford** routing algorithm and the **Ford-Fulkerson** algorithm, after the researchers who developed it (Bellman, 1957; and Ford and Fulkerson, 1962). It was the original ARPANET routing algorithm and was also used in the Internet under the name RIP and in early versions of DECnet and Novell's IPX. AppleTalk and Cisco routers use improved distance vector protocols.

In distance vector routing, each router maintains a routing table indexed by, and containing one entry for, each router in the subnet. This entry contains two parts: the preferred outgoing line to use for that destination, and an estimate of the time or distance to that destination. The metric used might be number of hops, time delay in milliseconds, total number of packets queued along the path, or something similar.

The router is assumed to know the "distance" to each of its neighbors. If the metric is hops, the distance is just one hop. If the metric is queue length, the router simply examines each queue. If the metric is delay, the router can measure it directly with special ECHO packets that the receiver just timestamps and sends back as fast as it can.

As an example, assume that delay is used as a metric and that the router knows the delay to each of its neighbors. Once every T msec each router sends to each neighbor a list of its estimated delays to each destination. It also receives a similar list from each neighbor. Imagine that one of these tables has just come in from neighbor X, with X_i being X's estimate of how long it takes to get to router i. If the router knows that the delay to X is m msec, it also knows that it can reach router i via X in $X_i + m$ msec via X. By performing this calculation for each neighbor, a router can find out which estimate seems the best and use that estimate and the corresponding line in its new routing table. Note that the old routing table is not used in the calculation.

This updating process is illustrated in Fig. 5-10. Part (a) shows a subnet. The first four columns of part (b) show the delay vectors received from the neighbors of router J. A claims to have a 12-msec delay to B, a 25-msec delay to C, a 40-msec delay to D, etc. Suppose that J has measured or estimated its delay to its neighbors, A, I, H, and K as 8, 10, 12, and 6 msec, respectively.



(a)

To	A	I	H	K	New estimated delay from J	Line
A	0	24	20	21	8	A
B	12	36	31	28	20	A
C	25	18	19	36	28	I
D	40	27	8	24	20	H
E	14	7	30	22	17	I
F	23	20	19	40	30	I
G	18	31	6	31	18	H
H	17	20	0	19	12	H
I	21	0	14	22	10	I
J	9	11	7	10	0	–
K	24	22	22	0	6	K
L	29	33	9	9	15	K
	JA delay is 8	JI delay is 10	JH delay is 12	JK delay is 6	New routing table for J	

Vectors received from J's four neighbors

(b)

Fig. 5-10. (a) A subnet. (b) Input from A, I, H, K, and the new routing table for J.

Consider how J computes its new route to router G. It knows that it can get to A in 8 msec, and A claims to be able to get to G in 18 msec, so J knows it can count on a delay of 26 msec to G if it forwards packets bound for G to A.

Similarly, it computes the delay to G via I, H, and K as 41 (31 + 10), 18 (6 + 12), and 37 (31 + 6) msec respectively. The best of these values is 18, so it makes an entry in its routing table that the delay to G is 18 msec, and that the route to use is via H. The same calculation is performed for all the other destinations, with the new routing table shown in the last column of the figure.

The Count-to-Infinity Problem

Distance vector routing works in theory but has a serious drawback in practice: although it converges to the correct answer, it may do so slowly. In particular, it reacts rapidly to good news, but leisurely to bad news. Consider a router whose best route to destination X is large. If on the next exchange neighbor A suddenly reports a short delay to X, the router just switches over to using the line to A to send traffic to X. In one vector exchange, the good news is processed.

To see how fast good news propagates, consider the five-node (linear) subnet of Fig. 5-11, where the delay metric is the number of hops. Suppose A is down initially and all the other routers know this. In other words, they have all recorded the delay to A as infinity.

A	B	C	D	E	
	∞	∞	∞	∞	Initially
	1	∞	∞	∞	After 1 exchange
	1	2	∞	∞	After 2 exchanges
	1	2	3	∞	After 3 exchanges
	1	2	3	4	After 4 exchanges

(a)

A	B	C	D	E	
	1	2	3	4	Initially
	3	2	3	4	After 1 exchange
	3	4	3	4	After 2 exchanges
	5	4	5	4	After 3 exchanges
	5	6	5	6	After 4 exchanges
	7	6	7	6	After 5 exchanges
	7	8	7	8	After 6 exchanges
		⋮			
	∞	∞	∞	∞	

(b)

Fig. 5-11. The count-to-infinity problem.

When A comes up, the other routers learn about it via the vector exchanges. For simplicity we will assume that there is a gigantic gong somewhere that is struck periodically to initiate a vector exchange at all routers simultaneously. At the time of the first exchange, B learns that its left neighbor has zero delay to A. B now makes an entry in its routing table that A is one hop away to the left. All the other routers still think that A is down. At this point, the routing table entries for A are as shown in the second row of Fig. 5-11(a). On the next exchange, C learns that B has a path of length 1 to A, so it updates its routing table to indicate a path of length 2, but D and E do not hear the good news until later. Clearly, the good news is spreading at the rate of one hop per exchange. In a subnet whose longest

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path is of length N hops, within N exchanges everyone will know about newly revived lines and routers.

Now let us consider the situation of Fig. 5-11(b), in which all the lines and routers are initially up. Routers B, C, D, and E have distances to A of 1, 2, 3, and 4, respectively. Suddenly A goes down, or alternatively, the line between A and B is cut, which is effectively the same thing from B's point of view.

At the first packet exchange, B does not hear anything from A. Fortunately, C says "Do not worry. I have a path to A of length 2." Little does B know that C's path runs through B itself. For all B knows, C might have ten outgoing lines all with independent paths to A of length 2. As a result, B now thinks it can reach A via C, with a path length of 3. D and E do not update their entries for A on the first exchange.

On the second exchange, C notices that each of its neighbors claims to have a path to A of length 3. It picks one of the them at random and makes its new distance to A 4, as shown in the third row of Fig. 5-11(b). Subsequent exchanges produce the history shown in the rest of Fig. 5-11(b).

From this figure, its should be clear why bad news travels slowly: no router ever has a value more than one higher than the minimum of all its neighbors. Gradually, all the routers work their way up to infinity, but the number of exchanges required depends on the numerical value used for infinity. For this reason, it is wise to set infinity to the longest path plus 1. If the metric is time delay, there is no well-defined upper bound, so a high value is needed to prevent a path with a long delay from being treated as down. Not entirely surprisingly, this problem is known as the **count-to-infinity** problem.

The Split Horizon Hack

Many ad hoc solutions to the count-to-infinity problem have been proposed in the literature, each one more complicated and less useful than the one before it. We will describe just one of them here and then tell why it, too, fails. The **split horizon** algorithm works the same way as distance vector routing, except that the distance to X is not reported on the line that packets for X are sent on (actually, it is reported as infinity). In the initial state of Fig. 5-11(b), for example, C tells D the truth about the distance to A, but C tells B that its distance to A is infinite. Similarly, D tells the truth to E but lies to C.

Now let us see what happens when A goes down. On the first exchange, B discovers that the direct line is gone, and C is reporting an infinite distance to A as well. Since neither of its neighbors can get to A, B sets its distance to infinity as well. On the next exchange, C hears that A is unreachable from both of its neighbors, so it marks A as unreachable too. Using split horizon, the bad news propagates one hop per exchange. This rate is much better than without split horizon.

The real bad news is that split horizon, although widely used, sometimes fails.

Consider, for example, the four-node subnet of Fig. 5-12. Initially, both *A* and *B* have a distance 2 to *D*, and *C* has a distance 1 there.

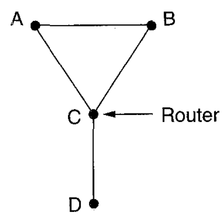

Fig. 5-12. An example where split horizon fails.

Now suppose that the *CD* line goes down. Using split horizon, both *A* and *B* tell *C* that they cannot get to *D*. Thus *C* immediately concludes that *D* is unreachable and reports this to both *A* and *B*. Unfortunately, *A* hears that *B* has a path of length 2 to *D*, so it assumes it can get to *D* via *B* in 3 hops. Similarly, *B* concludes it can get to *D* via *A* in 3 hops. On the next exchange, they each set their distance to *D* to 4. Both of them gradually count to infinity, precisely the behavior we were trying to avoid.

5.2.6. Link State Routing

Distance vector routing was used in the ARPANET until 1979, when it was replaced by link state routing. Two primary problems caused its demise. First, since the delay metric was queue length, it did not take line bandwidth into account when choosing routes. Initially, all the lines were 56 kbps, so line bandwidth was not an issue, but after some lines had been upgraded to 230 kbps and others to 1.544 Mbps, not taking bandwidth into account was a major problem. Of course, it would have been possible to change the delay metric to factor in line bandwidth, but a second problem also existed, namely, the algorithm often took too long to converge, even with tricks like split horizon. For these reasons, it was replaced by an entirely new algorithm now called **link state routing**. Variants of link state routing are now widely used.

The idea behind link state routing is simple and can be stated as five parts. Each router must

1. Discover its neighbors and learn their network addresses.
2. Measure the delay or cost to each of its neighbors.
3. Construct a packet telling all it has just learned.
4. Send this packet to all other routers.
5. Compute the shortest path to every other router.

In effect, the complete topology and all delays are experimentally measured and distributed to every router. Then Dijkstra's algorithm can be used to find the shortest path to every other router. Below we will consider each of these five steps in more detail.

Learning about the Neighbors

When a router is booted, its first task is to learn who its neighbors are. It accomplishes this goal by sending a special HELLO packet on each point-to-point line. The router on the other end is expected to send back a reply telling who it is. These names must be globally unique because when a distant router later hears that three routers are all connected to *F*, it is essential that it can determine whether or not all three mean the same *F*.

When two or more routers are connected by a LAN, the situation is slightly more complicated. Fig. 5-13(a) illustrates a LAN to which three routers, *A*, *C*, and *F*, are directly connected. Each of these routers is connected to one or more additional routers, as shown.

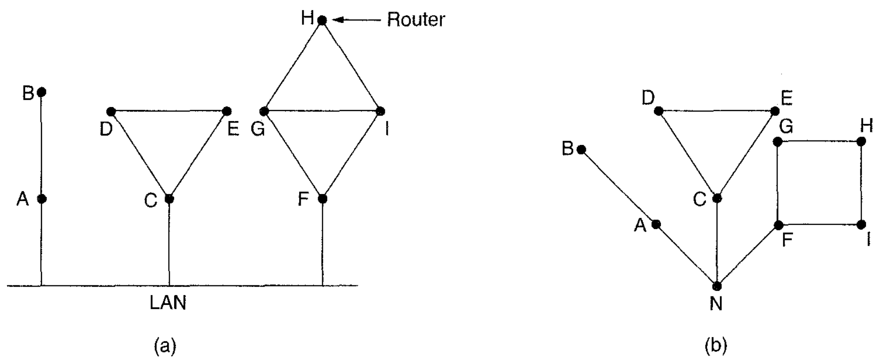

Fig. 5-13. (a) Nine routers and a LAN. (b) A graph model of (a).

One way to model the LAN is to consider it as a node itself, as shown in Fig. 5-13(b). Here we have introduced a new, artificial node, *N*, to which *A*, *C*, and *F* are connected. The fact that it is possible to go from *A* to *C* on the LAN is represented by the path *ANC* here.

Measuring Line Cost

The link state routing algorithm requires each router to know, or at least have a reasonable estimate, of the delay to each of its neighbors. The most direct way to determine this delay is to send a special ECHO packet over the line that the other side is required to send back immediately. By measuring the round-trip time and dividing it by two, the sending router can get a reasonable estimate of the delay. For even better results, the test can be conducted several times, and the average used.

An interesting issue is whether or not to take the load into account when measuring the delay. To factor the load in, the round-trip timer must be started when the ECHO packet is queued. To ignore the load, the timer should be started when the ECHO packet reaches the front of the queue.

Arguments can be made both ways. Including traffic-induced delays in the measurements means that when a router has a choice between two lines with the same bandwidth, one of which is heavily loaded all the time and one of which is not, it will regard the route over the unloaded line as a shorter path. This choice will result in better performance.

Unfortunately, there is also an argument against including the load in the delay calculation. Consider the subnet of Fig. 5-14, which is divided up into two parts, East and West, connected by two lines, *CF* and *EI*. Suppose that most of the traffic between East and West is using line *CF*, and as a result, this line is heavily loaded with long delays. Including queueing delay in the shortest path calculation will make *EI* more attractive. After the new routing tables have been installed, most of the East-West traffic will now go over *EI*, overloading this line. Consequently, in the next update, *CF* will appear to be the shortest path. As a result, the routing tables may oscillate wildly, leading to erratic routing and many potential problems. If load is ignored and only bandwidth is considered, this problem does not occur. Alternatively, the load can be spread over both lines, but this solution does not fully utilize the best path.

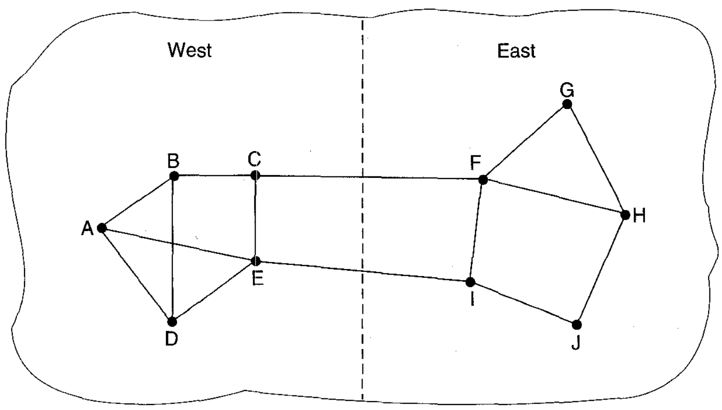

Fig. 5-14. A subnet in which the East and West parts are connected by two lines.

Building Link State Packets

Once the information needed for the exchange has been collected, the next step is for each router to build a packet containing all the data. The packet starts with the identity of the sender, followed by a sequence number and age (to be described later), and a list of neighbors. For each neighbor, the delay to that neighbor is given. An example subnet is given in Fig. 5-15(a) with delays shown in the lines. The corresponding link state packets for all six routers are shown in Fig. 5-15(b).

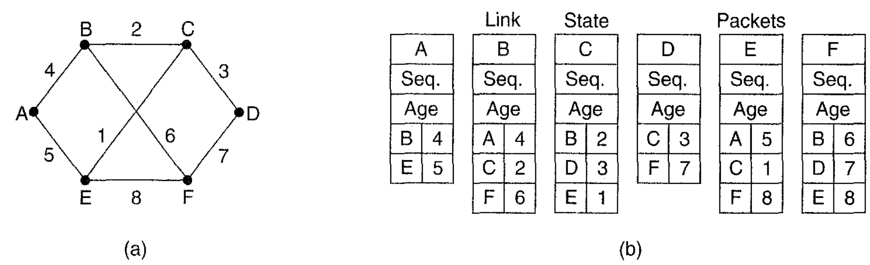

Fig. 5-15. (a) A subnet. (b) The link state packets for this subnet.

Building the link state packets is easy. The hard part is determining when to build them. One possibility is to build them periodically, that is, at regular intervals. Another possibility is when some significant event occurs, such as a line or neighbor going down or coming back up again, or changing its properties appreciably.

Distributing the Link State Packets

The trickiest part of the algorithm is distributing the link state packets reliably. As the packets are distributed and installed, the routers getting the first ones will change their routes. Consequently, the different routers may be using different versions of the topology, which can lead to inconsistencies, loops, unreachable machines, and other problems.

First we will describe the basic distribution algorithm. Later we will give some refinements. The fundamental idea is to use flooding to distribute the link state packets. To keep the flood in check, each packet contains a sequence number that is incremented for each new packet sent. Routers keep track of all the (source router, sequence) pairs they see. When a new link state packet comes in, it is checked against the list of packets already seen. If it is new, it is forwarded on all lines except the one it arrived on. If it is a duplicate, it is discarded.

If a packet with a sequence number lower than the highest one seen so far ever arrives, it is rejected as being obsolete.

This algorithm has a few problems, but they are manageable. First, if the sequence numbers wrap around, confusion will reign. The solution here is to use a 32-bit sequence number. With one link state packet per second, it would take 137 years to wrap around, so this possibility can be ignored.

Second, if a router ever crashes, it will lose track of its sequence number. If it starts again at 0, the next packet will be rejected as a duplicate.

Third, if a sequence number is ever corrupted and 65,540 is received instead of 4 (a 1-bit error), packets 5 through 65,540 will be rejected as obsolete, since the current sequence number is thought to be 65,540.

The solution to all these problems is to include the age of each packet after the sequence number and decrement it once per second. When the age hits zero, the information from that router is discarded. Normally, a new packet comes in, say, every 10 minutes, so router information only times out when a router is down (or six consecutive packets have been lost, an unlikely event). The age field is also decremented by each router during the initial flooding process, to make sure no packet can get lost and live for an indefinite period of time (a packet whose age is zero is discarded).

Some refinements to this algorithm make it more robust. When a link state packet comes in to a router for flooding, it is not queued for transmission immediately. Instead it is put in a holding area to wait a short while first. If another link state packet from the same source comes in before it is transmitted, their sequence numbers are compared. If they are equal, the duplicate is discarded. If they are different, the older one is thrown out. To guard against errors on the router-router lines, all link state packets are acknowledged. When a line goes idle, the holding area is scanned in round robin order to select a packet or acknowledgement to send.

The data structure used by router *B* for the subnet shown in Fig. 5-15(a) is depicted in Fig. 5-16. Each row here corresponds to a recently arrived, but as yet not fully processed, link state packet. The table records where the packet originated, its sequence number and age, and the data. In addition, there are send and acknowledgement flags for each of *B*'s three lines (to *A*, *C*, and *F*, respectively). The send flags mean that the packet must be sent on the indicated line. The acknowledgement flags mean that it must be acknowledged there.

In Fig. 5-16, the link state packet from *A* arrived directly, so it must be sent to *C* and *F* and acknowledged to *A*, as indicated by the flag bits. Similarly, the packet from *F* has to be forwarded to *A* and *C* and acknowledged to *F*.

However, the situation with the third packet, from *E*, is different. It arrived twice, once via *EAB* and once via *EFB*. Consequently, it has to be sent only to *C*, but acknowledged to both *A* and *F*, as indicated by the bits.

If a duplicate arrives while the original is still in the buffer, bits have to be changed. For example, if a copy of *C*'s state arrives from *F* before the fourth

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Source	Seq.	Age	Send flags A	Send flags C	Send flags F	ACK flags A	ACK flags C	ACK flags F	Data
A	21	60	0	1	1	1	0	0	
F	21	60	1	1	0	0	0	1	
E	21	59	0	1	0	1	0	1	
C	20	60	1	0	1	0	1	0	
D	21	59	1	0	0	0	1	1	

Fig. 5-16. The packet buffer for router *B* in Fig. 5-15.

entry in the table has been forwarded, the six bits will be changed to 100011 to indicate that the packet must be acknowledged to *F* but not sent there.

Computing the New Routes

Once a router has accumulated a full set of link state packets, it can construct the entire subnet graph because every link is represented. Every link is, in fact, represented twice, once for each direction. The two values can be averaged or used separately.

Now Dijkstra's algorithm can be run locally to construct the shortest path to all possible destinations. The results of this algorithm can be installed in the routing tables, and normal operation resumed.

For a subnet with n routers, each of which has k neighbors, the memory required to store the input data is proportional to kn. For large subnets, this can be a problem. Also, the computation time can be an issue. Nevertheless, in many practical situations, link state routing works well.

However, problems with the hardware or software can wreak havoc with this algorithm (also with other ones). For example, if a router claims to have a line it does not have, or forgets a line it does have, the subnet graph will be incorrect. If a router fails to forward packets, or corrupts them while forwarding them, trouble will arise. Finally, if it runs out of memory or does the routing calculation wrong, bad things will happen. As the subnet grows into the range of tens or hundreds of thousands of nodes, the probability of some router failing occasionally becomes nonnegligible. The trick is to try to arrange to limit the damage when the inevitable happens. Perlman (1988) discusses these problems and their solutions in detail.

Link state routing is widely used in actual networks, so a few words about some example protocols using it are in order. The OSPF protocol, which is increasingly being used in the Internet, uses a link state algorithm. We will describe OSPF in Sec. 5.5.5.

Another important link state protocol is **IS-IS** (**Intermediate System-Intermediate System**), which was designed for DECnet and later adopted by ISO for use with its connectionless network layer protocol, CLNP. Since then it has been modified to handle other protocols as well, most notably, IP. IS-IS is used in numerous Internet backbones (including the old NSFNET backbone), and in some digital cellular systems such as CDPD. Novell NetWare uses a minor variant of IS-IS (NLSP) for routing IPX packets.

Basically IS-IS distributes a picture of the router topology, from which the shortest paths are computed. Each router announces, in its link state information, which network layer addresses it can reach directly. These addresses can be IP, IPX, AppleTalk, or any other addresses. IS-IS can even support multiple network layer protocols at the same time.

Many of the innovations designed for IS-IS were adopted by OSPF (OSPF was designed several years after IS-IS). These include a self-stabilizing method of flooding link state updates, the concept of a designated router on a LAN, and the method of computing and supporting path splitting and multiple metrics. As a consequence, there is very little difference between IS-IS and OSPF. The most important difference is that IS-IS is encoded in such a way that it is easy and natural to simultaneously carry information about multiple network layer protocols, a feature OSPF does not have. This advantage is especially valuable in large multiprotocol environments.

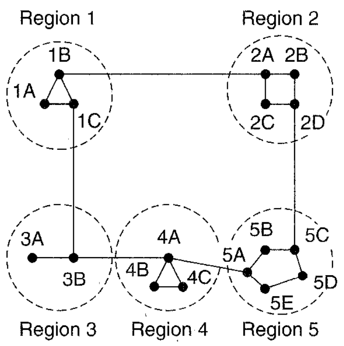
5.2.7. Hierarchical Routing

As networks grow in size, the router routing tables grow proportionally. Not only is router memory consumed by ever increasing tables, but more CPU time is needed to scan them and more bandwidth is needed to send status reports about them. At a certain point the network may grow to the point where it is no longer feasible for every router to have an entry for every other router, so the routing will have to be done hierarchically, as it is in the telephone network.

When hierarchical routing is used, the routers are divided into what we will call **regions**, with each router knowing all the details about how to route packets to destinations within its own region, but knowing nothing about the internal structure of other regions. When different networks are connected together, it is natural to regard each one as a separate region in order to free the routers in one network from having to know the topological structure of the other ones.

For huge networks, a two-level hierarchy may be insufficient; it may be necessary to group the regions into clusters, the clusters into zones, the zones into groups, and so on, until we run out of names for aggregations. As an example of a multilevel hierarchy, consider how a packet might be routed from Berkeley, California to Malindi, Kenya. The Berkeley router would know the detailed topology within California but would send all out-of-state traffic to the Los Angeles router. The Los Angeles router would be able to route traffic to other domestic routers, but would send foreign traffic to New York. The New York router would be programmed to direct all traffic to the router in the destination country responsible for handling foreign traffic, say in Nairobi. Finally, the packet would work its way down the tree in Kenya until it got to Malindi.

Figure 5-17 gives a quantitative example of routing in a two-level hierarchy with five regions. The full routing table for router 1*A* has 17 entries, as shown in Fig. 5-17(b). When routing is done hierarchically, as in Fig. 5-17(c), there are entries for all the local routers as before, but all other regions have been condensed into a single router, so all traffic for region 2 goes via the 1*B*–2*A* line, but the rest of the remote traffic goes via the 1*C*–3*B* line. Hierarchical routing has reduced the table from 17 to 7 entries. As the ratio of the number of regions to the number of routers per region grows, the savings in table space increase.



(a)

Full table for 1A

Dest.	Line	Hops
1A	–	–
1B	1B	1
1C	1C	1
2A	1B	2
2B	1B	3
2C	1B	3
2D	1B	4
3A	1C	3
3B	1C	2
4A	1C	3
4B	1C	4
4C	1C	4
5A	1C	4
5B	1C	5
5C	1B	5
5D	1C	6
5E	1C	5

(b)

Hierarchical table for 1A

Dest.	Line	Hops
1A	–	–
1B	1B	1
1C	1C	1
2	1B	2
3	1C	2
4	1C	3
5	1C	4

(c)

Fig. 5-17. Hierarchical routing.

Unfortunately, these gains in space are not free. There is a penalty to be paid, and this penalty is in the form of increased path length. For example, the best route from 1*A* to 5*C* is via region 2, but with hierarchical routing all traffic to region 5 goes via region 3, because that is better for most destinations in region 5.

When a single network becomes very large, an interesting question is: How many levels should the hierarchy have? For example, consider a subnet with 720

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routers. If there is no hierarchy, each router needs 720 routing table entries. If the subnet is partitioned into 24 regions of 30 routers each, each router needs 30 local entries plus 23 remote entries for a total of 53 entries. If a three-level hierarchy is chosen, with eight clusters, each containing 9 regions of 10 routers, each router needs 10 entries for local routers, 8 entries for routing to other regions within its own cluster, and 7 entries for distant clusters, for a total of 25 entries. Kamoun and Kleinrock (1979) have discovered that the optimal number of levels for an N router subnet is $\ln N$, requiring a total of $e \ln N$ entries per router. They have also shown that the increase in effective mean path length caused by hierarchical routing is sufficiently small that it is usually acceptable.

5.2.8. Routing for Mobile Hosts

Millions of people have portable computers nowadays, and they generally want to read their email and access their normal file systems wherever in the world they may be. These mobile hosts introduce a new complication: to route a packet to a mobile host, the network first has to find it. The subject of incorporating mobile hosts into a network is very young, but in this section we will sketch some of the issues here and give a possible solution.

The model of the world that network designers typically use is shown in Fig. 5-18. Here we have a WAN consisting of routers and hosts. Connected to the WAN are LANs, MANs, and wireless cells of the type we studied in Chap. 2.

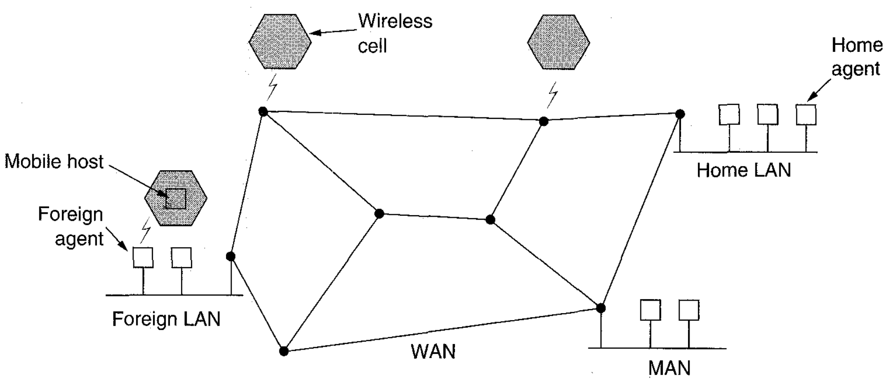

Fig. 5-18. A WAN to which LANs, MANs, and wireless cells are attached.

Users who never move are said to be stationary. They are connected to the network by copper wires or fiber optics. In contrast, we can distinguish two other kinds of users. Migratory users are basically stationary users who move from one fixed site to another from time to time but use the network only when they are physically connected to it. Roaming users actually compute on the run and want to maintain their connections as they move around. We will use the term **mobile users** to mean either of the latter two categories, that is, all users who are away from home.

All users are assumed to have a permanent **home location** that never changes. Users also have a permanent home address that can be used to determine their home locations, analogous to the way the telephone number 1-212-5551212 indicates the United States (country code 1) and Manhattan (212). The routing goal in systems with mobile users is to make it possible to send packets to mobile users using their home addresses, and have the packets efficiently reach them wherever they may be. The trick, of course, is to find them.

In the model of Fig. 5-18, the world is divided up (geographically) into small units. Let us call them areas, where an area is typically a LAN or wireless cell. Each area has one or more **foreign agents**, which keep track of all mobile users visiting the area. In addition, each area has a **home agent**, which keeps track of users whose home is in the area, but who are currently visiting another area.

When a new user enters an area, either by connecting to it (e.g., plugging into the LAN), or just wandering into the cell, his computer must register itself with the foreign agent there. The registration procedure typically works like this:

1. Periodically, each foreign agent broadcasts a packet announcing its existence and address. A newly arrived mobile host may wait for one of these messages, but if none arrives quickly enough, the mobile host can broadcast a packet saying: "Are there any foreign agents around?"

2. The mobile host registers with the foreign agent, giving its home address, current data link layer address, and some security information.

3. The foreign agent contacts the mobile host's home agent and says: "One of your hosts is over here." The message from the foreign agent to the home agent contains the foreign agent's network address. It also includes the security information, to convince the home agent that the mobile host is really there.

4. The home agent examines the security information, which contains a timestamp, to prove that it was generated within the past few seconds. If it is happy, it tells the foreign agent to proceed.

5. When the foreign agent gets the acknowledgement from the home agent, it makes an entry in its tables and informs the mobile host that it is now registered.

Ideally, when a user leaves an area, that, too, should be announced to allow deregistration, but many users abruptly turn off their computers when done.

When a packet is sent to a mobile user, it is routed to the user's home LAN because that is what the address says should be done, as illustrated in step 1 of Fig. 5-19. Packets sent to the mobile user on its home LAN are intercepted by the home agent. The home agent then looks up the mobile user's new (temporary) location and finds the address of the foreign agent handling the mobile user. The home agent then does two things. First, it encapsulates the packet in the payload field of an outer packet and sends the latter to the foreign agent (step 2 in Fig. 5-19). This mechanism is called tunneling; we will look at it in more detail later. After getting the encapsulated packet, the foreign agent removes the original packet from the payload field and sends it to the mobile user as a data link frame.

Second, the home agent tells the sender to henceforth send packets to the mobile host by encapsulating them in the payload of packets explicitly addressed to the foreign agent, instead of just sending them to the mobile user's home address (step 3). Subsequent packets can now be routed directly to the user via the foreign agent (step 4), bypassing the home location entirely.

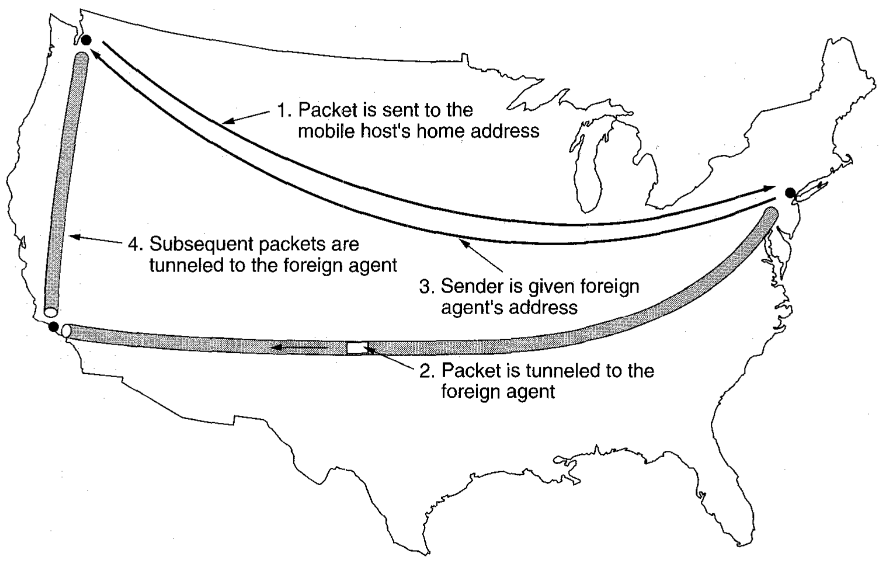

Fig. 5-19. Packet routing for mobile users.

The various schemes that have been proposed differ in several ways. First, there is the issue of how much of this protocol is carried out by the routers and how much by the hosts, and in the latter case, by which layer in the hosts. Second, a few schemes, routers along the way record mapped addresses so they can intercept and redirect traffic even before it gets to the home location. Third, in some schemes each visitor is given a unique temporary address; in others, the temporary address refers to an agent that handles traffic for all visitors.

Fourth, the schemes differ in how they actually manage to arrange for packets that are addressed to one destination to be delivered to a different one. One choice is changing the destination address and just retransmitting the modified packet. Alternatively, the whole packet, home address and all, can be encapsulated inside the payload of another packet sent to the temporary address. Finally, the schemes differ in their security aspects. In general, when a host or router gets a message of the form "Starting right now, please send all of Cayla's mail to me," it might have a couple of questions about whom it was talking to and whether or not this is a good idea. Several mobile host protocols are discussed and compared in (Ioannidis and Maguire, 1993; Myles and Skellern, 1993; Perkins, 1993; Teraoka et al., 1993; and Wada et al., 1993).

5.2.9. Broadcast Routing

For some applications, hosts need to send messages to many or all other hosts. For example, a service distributing weather reports, stock market updates, or live radio programs might work best by broadcasting to all machines and letting those that are interested read the data. Sending a packet to all destinations simultaneously is called **broadcasting**; various methods have been proposed for doing it.

One broadcasting method that requires no special features from the subnet is for the source to simply send a distinct packet to each destination. Not only is the method wasteful of bandwidth, but it also requires the source to have a complete list of all destinations. In practice this may be the only possibility, but it is the least desirable of the methods.

Flooding is another obvious candidate. Although flooding is ill-suited for ordinary point-to-point communication, for broadcasting it might rate serious consideration, especially if none of the methods described below are applicable. The problem with flooding as a broadcast technique is the same problem it has as a point-to-point routing algorithm: it generates too many packets and consumes too much bandwidth.

A third algorithm is **multidestination routing**. If this method is used, each packet contains either a list of destinations or a bit map indicating the desired destinations. When a packet arrives at a router, the router checks all the destinations to determine the set of output lines that will be needed. (An output line is needed if it is the best route to at least one of the destinations.) The router generates a new copy of the packet for each output line to be used and includes in each packet only those destinations that are to use the line. In effect, the destination set is partitioned among the output lines. After a sufficient number of hops, each packet will carry only one destination and can be treated as a normal packet. Multidestination routing is like separately addressed packets, except that when several packets must follow the same route, one of them pays full fare and the rest ride free.

A fourth broadcast algorithm makes explicit use of the sink tree for the router initiating the broadcast, or any other convenient spanning tree for that matter. A

spanning tree is a subset of the subnet that includes all the routers but contains no loops. If each router knows which of its lines belong to the spanning tree, it can copy an incoming broadcast packet onto all the spanning tree lines except the one it arrived on. This method makes excellent use of bandwidth, generating the absolute minimum number of packets necessary to do the job. The only problem is that each router must have knowledge of some spanning tree for it to be applicable. Sometimes this information is available (e.g., with link state routing) but sometimes it is not (e.g., with distance vector routing).

Our last broadcast algorithm is an attempt to approximate the behavior of the previous one, even when the routers do not know anything at all about spanning trees. The idea is remarkably simple once it has been pointed out. When a broadcast packet arrives at a router, the router checks to see if the packet arrived on the line that is normally used for sending packets *to* the source of the broadcast. If so, there is an excellent chance that the broadcast packet itself followed the best route from the router and is therefore the first copy to arrive at the router. This being the case, the router forwards copies of it onto all lines except the one it arrived on. If, however, the broadcast packet arrived on a line other than the preferred one for reaching the source, the packet is discarded as a likely duplicate.

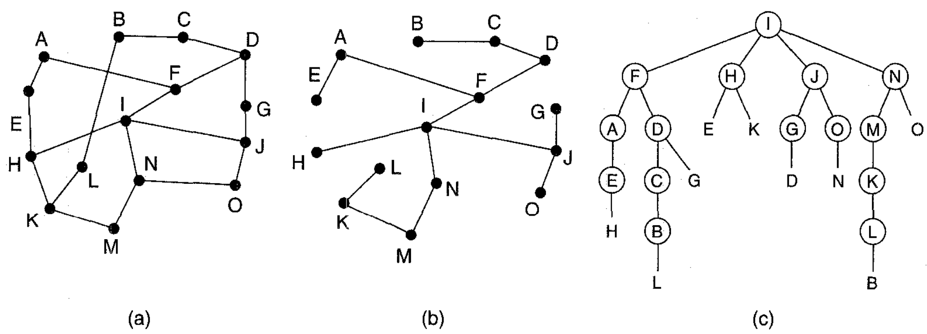

Fig. 5-20. Reverse path forwarding. (a) A subnet. (b) A spanning tree. (c) The tree built by reverse path forwarding.

An example of the algorithm, called **reverse path forwarding**, is shown in Fig. 5-20. Part (a) shows a subnet, part (b) shows a sink tree for router *I* of that subnet, and part (c) shows how the reverse path algorithm works. On the first hop, *I* sends packets to *F*, *H*, *J*, and *N*, as indicated by the second row of the tree. Each of these packets arrives on the preferred path to *I* (assuming that the preferred path falls along the sink tree) and is so indicated by a circle around the letter. On the second hop, eight packets are generated, two by each of the routers that received a packet on the first hop. As it turns out, all eight of these arrive at previously unvisited routers, and five of these arrive along the preferred line. Of the six packets generated on the third hop, only three arrive on the preferred path (at *C*, *E*, and *K*); the others are duplicates. After five hops and 23 packets, the broadcasting terminates, compared with four hops and 14 packets had the sink tree been followed exactly.

The principal advantage of reverse path forwarding is that it is both reasonably efficient and easy to implement. It does not require routers to know about spanning trees, nor does it have the overhead of a destination list or bit map in each broadcast packet as does multidestination addressing. Nor does it require any special mechanism to stop the process, as flooding does (either a hop counter in each packet and a priori knowledge of the subnet diameter, or a list of packets already seen per source).

5.2.10. Multicast Routing

For some applications, widely-separated processes work together in groups, for example, a group of processes implementing a distributed database system. It frequently is necessary for one process to send a message to all the other members of the group. If the group is small, it can just send each other member a point-to-point message. If the group is large, this strategy is expensive. Sometimes broadcasting can be used, but using broadcasting to inform 1000 machines on a million-node network is inefficient because most receivers are not interested in the message (or worse yet, they are definitely interested but are not supposed to see it). Thus we need a way to send messages to well-defined groups that are numerically large in size but small compared to the network as a whole.

Sending a message to such a group is called **multicasting**, and its routing algorithm is called **multicast routing**. In this section we will describe one way of doing multicast routing. For additional information, see (Deering and Cheriton, 1990; Deering et al., 1994; and Rajagopalan, 1992).

To do multicasting, group management is required. Some way is needed to create and destroy groups, and for processes to join and leave groups. How these tasks are accomplished is not of concern to the routing algorithm. What is of concern is that when a process joins a group, it informs its host of this fact. It is important that routers know which of their hosts belong to which groups. Either hosts must inform their routers about changes in group membership, or routers must query their hosts periodically. Either way, routers learn about which of their hosts are in which groups. Routers tell their neighbors, so the information propagates through the subnet.

To do multicast routing, each router computes a spanning tree covering all other routers in the subnet. For example, in Fig. 5-21(a) we have a subnet with two groups, 1 and 2. Some routers are attached to hosts that belong to one or both of these groups, as indicated in the figure. A spanning tree for the leftmost router is shown in Fig. 5-21(b).

When a process sends a multicast packet to a group, the first router examines its spanning tree and prunes it, removing all lines that do not lead to hosts that are

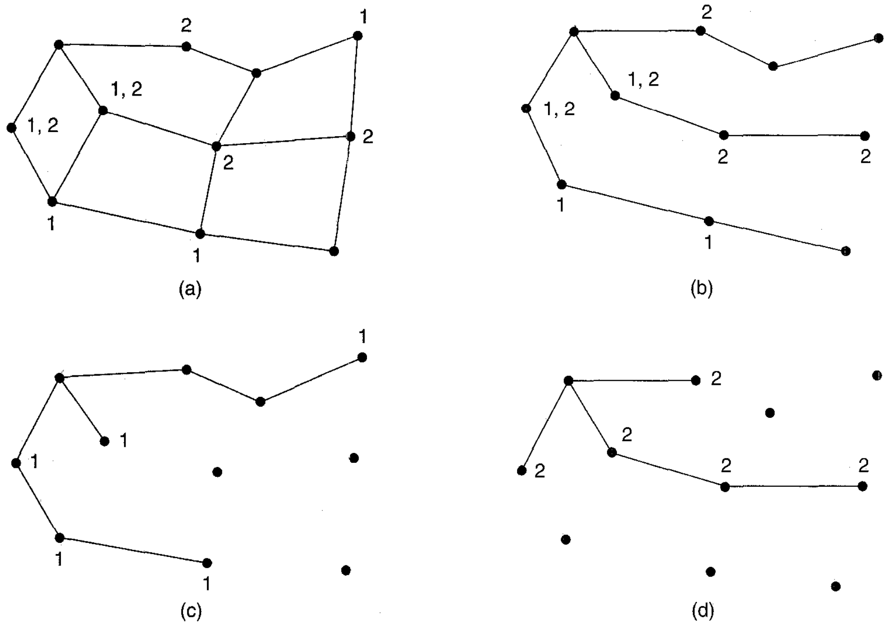

Fig. 5-21. (a) A subnet. (b) A spanning tree for the leftmost router. (c) A multicast tree for group 1. (d) A multicast tree for group 2.

members of the group. In our example, Fig. 5-21(c) shows the pruned spanning tree for group 1. Similarly, Fig. 5-21(d) shows the pruned spanning tree for group 2. Multicast packets are forwarded only along the appropriate spanning tree.

Various ways of pruning the spanning tree are possible. The simplest one can be used if link state routing is used, and each router is aware of the complete subnet topology, including which hosts belong to which groups. Then the spanning tree can be pruned by starting at the end of each path and working toward the root, removing all routers that do not belong to the group in question.

With distance vector routing, a different pruning strategy can be followed. The basic algorithm is reverse path forwarding. However, whenever a router with no hosts interested in a particular group and no connections to other routers receives a multicast message for that group, it responds with a PRUNE message, telling the sender not to send it any more multicasts for that group. When a router with no group members among its own hosts has received such messages on all its lines, it, too, can respond with a PRUNE message. In this way, the subnet is recursively pruned.

One potential disadvantage of this algorithm is that it scales poorly to large networks. Suppose that a network has n groups, each with an average of m

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members. For each group, *m* pruned spanning trees must be stored, for a total of *mn* trees. When many large groups exist, considerable storage is needed to store all the trees.

An alternative design uses **core-base trees** (Ballardie et al., 1993). Here, a single spanning tree per group is computed, with the root (the core) near the middle of the group. To send a multicast message, a host sends it to the core, which then does the multicast along the spanning tree. Although this tree will not be optimal for all sources, the reduction in storage costs from *m* trees to one tree per group is a major saving.

5.3. CONGESTION CONTROL ALGORITHMS

When too many packets are present in (a part of) the subnet, performance degrades. This situation is called **congestion**. Figure 5-22 depicts the symptom. When the number of packets dumped into the subnet by the hosts is within its carrying capacity, they are all delivered (except for a few that are afflicted with transmission errors), and the number delivered is proportional to the number sent. However, as traffic increases too far, the routers are no longer able to cope, and they begin losing packets. This tends to make matters worse. At very high traffic, performance collapses completely, and almost no packets are delivered.

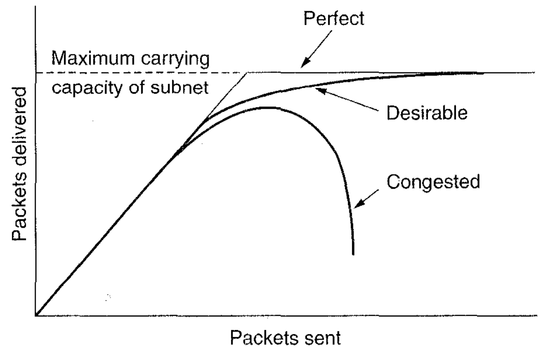


Fig. 5-22. When too much traffic is offered, congestion sets in and performance degrades sharply.

Congestion can be brought about by several factors. If all of a sudden, streams of packets begin arriving on three or four input lines and all need the same output line, a queue will build up. If there is insufficient memory to hold all of them, packets will be lost. Adding more memory may help up to a point, but Nagle (1987) discovered that if routers have an infinite amount of memory, congestion gets worse, not better, because by the time packets get to the front of the queue, they have already timed out (repeatedly), and duplicates have been sent. All these packets will be dutifully forwarded to the next router, increasing the load all the way to the destination.

Slow processors can also cause congestion. If the routers' CPUs are slow at performing the bookkeeping tasks required of them (queueing buffers, updating tables, etc.), queues can build up, even though there is excess line capacity. Similarly, low-bandwidth lines can also cause congestion. Upgrading the lines but not changing the processors, or vice versa, often helps a little, but frequently just shifts the bottleneck. Also, upgrading part, but not all, of the system, often just moves the bottleneck somewhere else. The real problem is frequently a mismatch between parts of the system. This problem will persist until all the components are in balance.

Congestion tends to feed upon itself and become worse. If a router has no free buffers, it must ignore newly arriving packets. When a packet is discarded, the sending router (a neighbor) may time out and retransmit it, perhaps ultimately many times. Since it cannot discard the packet until it has been acknowledged, congestion at the receiver's end forces the sender to refrain from releasing a buffer it would have normally freed. In this manner, congestion backs up, like cars approaching a toll booth.

It is worth explicitly pointing out the difference between congestion control and flow control, as the relationship is subtle. Congestion control has to do with making sure the subnet is able to carry the offered traffic. It is a global issue, involving the behavior of all the hosts, all the routers, the store-and-forwarding processing within the routers, and all the other factors that tend to diminish the carrying capacity of the subnet.

Flow control, in contrast, relates to the point-to-point traffic between a given sender and a given receiver. Its job is to make sure that a fast sender cannot continually transmit data faster than the receiver can absorb it. Flow control nearly always involves some direct feedback from the receiver to the sender to tell the sender how things are doing at the other end.

To see the difference between these two concepts, consider a fiber optic network with a capacity of 1000 gigabits/sec on which a supercomputer is trying to transfer a file to a personal computer at 1 Gbps. Although there is no congestion (the network itself is not in trouble), flow control is needed to force the supercomputer to stop frequently to give the personal computer a chance to breathe.

At the other extreme, consider a store-and-forward network with 1-Mbps lines and 1000 large computers, half of which are trying to transfer files at 100 kbps to the other half. Here the problem is not that of fast senders overpowering slow receivers, but simply that the total offered traffic exceeds what the network can handle.

The reason congestion control and flow control are often confused is that some congestion control algorithms operate by sending messages back to the various sources telling them to slow down when the network gets into trouble. Thus a host can get a "slow down" message either because the receiver cannot handle the load, or because the network cannot handle it. We will come back to this point later.

We will start our study of congestion control by looking at a general model for dealing with it. Then we will look at broad approaches to preventing it in the first place. After that, we will look at various dynamic algorithms for coping with it once it has set in.

5.3.1. General Principles of Congestion Control

Many problems in complex systems, such as computer networks, can be viewed from a control theory point of view. This approach leads to dividing all solutions into two groups: open loop and closed loop. Open loop solutions attempt to solve the problem by good design, in essence, to make sure it does not occur in the first place. Once the system is up and running, midcourse corrections are not made.

Tools for doing open-loop control include deciding when to accept new traffic, deciding when to discard packets and which ones, and making scheduling decisions at various points in the network. All of these have in common the fact that they make decisions without regard to the current state of the network.

In contrast, closed loop solutions are based on the concept of a feedback loop. This approach has three parts when applied to congestion control:

1. Monitor the system to detect when and where congestion occurs.
2. Pass this information to places where action can be taken.
3. Adjust system operation to correct the problem.

Various metrics can be used to monitor the subnet for congestion. Chief among these are the percentage of all packets discarded for lack of buffer space, the average queue lengths, the number of packets that time out and are retransmitted, the average packet delay, and the standard deviation of packet delay. In all cases, rising numbers indicate growing congestion.

The second step in the feedback loop is to transfer the information about the congestion from the point where it is detected to the point where something can be done about it. The obvious way is for the router detecting the congestion to send a packet to the traffic source or sources, announcing the problem. Of course, these extra packets increase the load at precisely the moment that more load is not needed, namely, when the subnet is congested.

However, other possibilities also exist. For example, a bit or field can be reserved in every packet for routers to fill in whenever congestion gets above some threshold level. When a router detects this congested state, it fills in the field in all outgoing packets, to warn the neighbors.

Still another approach is to have hosts or routers send probe packets out periodically to explicitly ask about congestion. This information can then be used to route traffic around problem areas. Some radio stations have helicopters flying around their cities to report on road congestion in the hope that their listeners will route their packets (cars) around hot spots.

In all feedback schemes, the hope is that knowledge of congestion will cause the hosts to take appropriate action to reduce the congestion. To work correctly, the time scale must be adjusted carefully. If every time two packets arrive in a row, a router yells STOP, and every time a router is idle for 20 μsec it yells GO, the system will oscillate wildly and never converge. On the other hand, if it waits 30 minutes to make sure before saying anything, the congestion control mechanism will react too sluggishly to be of any real use. To work well, some kind of averaging is needed, but getting the time constant right is a nontrivial matter.

Many congestion control algorithms are known. To provide a way to organize them in a sensible way, Yang and Reddy (1995) have developed a taxonomy for congestion control algorithms. They begin by dividing all algorithms into open loop or closed loop, as described above. They further divide the open loop algorithms into ones that act at the source versus ones that act at the destination. The closed loop algorithms are also divided into two subcategories: explicit feedback versus implicit feedback. In explicit feedback algorithms, packets are sent back from the point of congestion to warn the source. In implicit algorithms, the source deduces the existence of congestion by making local observations, such as the time needed for acknowledgements to come back.

The presence of congestion means that the load is (temporarily) greater than the resources (in part of the system) can handle. Two solutions come to mind: increase the resources or decrease the load. For example, the subnet may start using dial-up telephone lines to temporarily increase the bandwidth between certain points. In systems like SMDS (see Chap. 1), it may ask the carrier for additional bandwidth for a while. On satellite systems, increasing transmission power often gives higher bandwidth. Splitting traffic over multiple routes instead of always using the best one may also effectively increase the bandwidth. Finally, spare routers that are normally used only as backups (to make the system fault tolerant) can be put on-line to give more capacity when serious congestion appears.

However, sometimes it is not possible to increase the capacity, or it has already been increased to the limit. The only way then to beat back the congestion is to decrease the load. Several ways exist to reduce the load, including denying service to some users, degrading service to some or all users, and having users schedule their demands in a more predictable way.

Some of these methods, which we will study shortly, can best be applied to virtual circuits. For subnets that use virtual circuits internally, these methods can be used at the network layer. For datagram subnets, they can nevertheless sometimes be used on transport layer connections. In this chapter, we will focus on their use in the network layer. In the next one, we will see what can be done at the transport layer to manage congestion.

5.3.2. Congestion Prevention Policies

Let us begin our study of methods to control congestion by looking at open loop systems. These systems are designed to minimize congestion in the first place, rather than letting it happen and reacting after the fact. They try to achieve their goal by using appropriate policies at various levels. In Fig. 5-23 we see different data link, network, and transport policies that can affect congestion (Jain, 1990).

Layer	Policies
Transport	• Retransmission policy • Out-of-order caching policy • Acknowledgement policy • Flow control policy • Timeout determination
Network	• Virtual circuits versus datagram inside the subnet • Packet queueing and service policy • Packet discard policy • Routing algorithm • Packet lifetime management
Data link	• Retransmission policy • Out-of-order caching policy • Acknowledgement policy • Flow control policy

Fig. 5-23. Policies that affect congestion.

Let us start at the data link layer and work our way upward. The retransmission policy deals with how fast a sender times out and what it transmits upon timeout. A jumpy sender that times out quickly and retransmits all outstanding packets using go back n will put a heavier load on the system than a leisurely sender that uses selective repeat. Closely related to this is caching policy. If receivers routinely discard all out-of-order packets, these packets will have to be transmitted again later, creating extra load.

Acknowledgement policy also affects congestion. If each packet is acknowledged immediately, the acknowledgement packets generate extra traffic. However, if acknowledgements are saved up to piggyback onto reverse traffic, extra timeouts and retransmissions may result. A tight flow control scheme (e.g., a small window) reduces the data rate and thus helps fight congestion.

At the network layer, the choice between virtual circuits and datagrams affects congestion, since many congestion control algorithms work only with virtual circuit subnets. Packet queueing and service policy relates to whether routers have one queue per input line, one queue per output line, or both. It also relates to the order packets are processed (e.g., round robin, or priority based). Discard policy is the rule telling which packet is dropped when there is no space. A good policy can help alleviate congestion and a bad one can make it worse.

The routing algorithm can help avoid congestion by spreading the traffic over all the lines, whereas a bad one can send too much traffic over already congested lines. Finally, packet lifetime management deals with how long a packet may live before being discarded. If it is too long, lost packets may clog up the works for a long time, but if it is too short, packets may sometimes time out before reaching their destination, thus inducing retransmissions.

In the transport layer, the same issues occur as in the data link layer, but in addition, determining the timeout interval is harder because the transit time across the network is less predictable than the transit time over a wire between two routers. If it is too short, extra packets will be sent unnecessarily. If it is too long, congestion will be reduced, but the response time will suffer whenever a packet is lost.

5.3.3. Traffic Shaping

One of the main causes of congestion is that traffic is often bursty. If hosts could be made to transmit at a uniform rate, congestion would be less common. Another open loop method to help manage congestion is forcing the packets to be transmitted at a more predictable rate. This approach to congestion management is widely used in ATM networks and is called **traffic shaping**.

Traffic shaping is about regulating the average *rate* (and burstiness) of data transmission. In contrast, the sliding window protocols we studied earlier limit the amount of data in transit at once, not the rate at which it is sent. When a virtual circuit is set up, the user and the subnet (i.e., the customer and the carrier) agree on a certain traffic pattern (i.e., shape) for that circuit. As long as the customer fulfills her part of the bargain and only sends packets according to the agreed upon contract, the carrier promises to deliver them all in a timely fashion. Traffic shaping reduces congestion and thus helps the carrier live up to its promise. Such agreements are not so important for file transfers but are of great importance for real-time data, such as audio and video connections, which do not tolerate congestion well.

In effect, with traffic shaping the customer says to the carrier: "My transmission pattern will look like this. Can you handle it?" If the carrier agrees, the issue arises of how the carrier can tell if the customer is following the agreement, and what to do if the customer is not. Monitoring a traffic flow is called **traffic policing**. Agreeing to a traffic shape and policing it afterward are easier with virtual circuit subnets than with datagram subnets. However, even with datagram subnets, the same ideas can be applied to transport layer connections.

The Leaky Bucket Algorithm

Imagine a bucket with a small hole in the bottom, as illustrated in Fig. 5-24(a). No matter at what rate water enters the bucket, the outflow is at a constant rate, ρ, when there is any water in the bucket, and zero when the bucket is empty. Also, once the bucket is full, any additional water entering it spills over the sides and is lost (i.e., does not appear in the output stream under the hole).

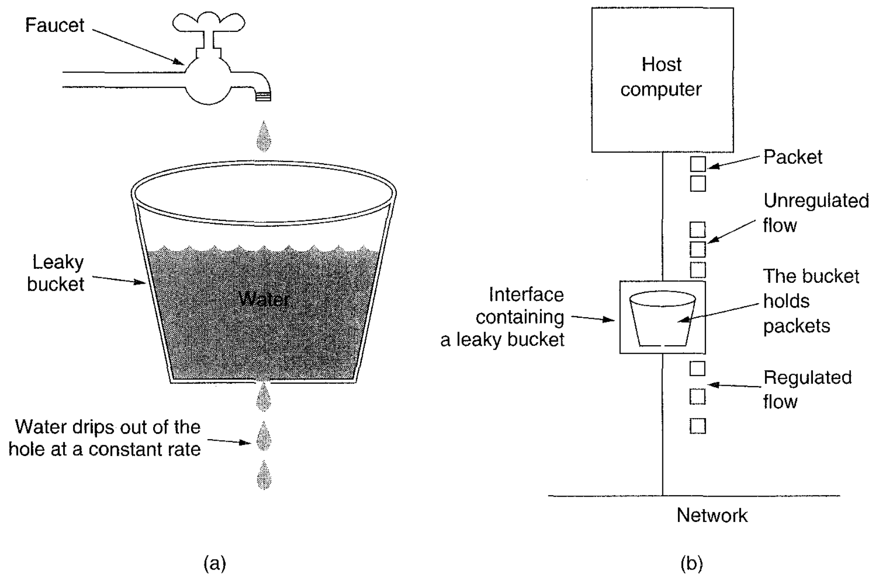


Fig. 5-24. (a) A leaky bucket with water. (b) A leaky bucket with packets.

The same idea can be applied to packets, as shown in Fig. 5-24(b). Conceptually, each host is connected to the network by an interface containing a leaky bucket, that is, a finite internal queue. If a packet arrives at the queue when it is full, the packet is discarded. In other words, if one or more processes within the host try to send a packet when the maximum number are already queued, the new packet is unceremoniously discarded. This arrangement can be built into the hardware interface or simulated by the host operating system. It was first proposed by Turner (1986) and is called the **leaky bucket algorithm**. In fact, it is nothing other than a single-server queueing system with constant service time.

The host is allowed to put one packet per clock tick onto the network. Again, this can be enforced by the interface card or by the operating system. This mechanism turns an uneven flow of packets from the user processes inside the host into an even flow of packets onto the network, smoothing out bursts and greatly reducing the chances of congestion.

When the packets are all the same size (e.g., ATM cells), this algorithm can be used as described. However, when variable-sized packets are being used, it is often better to allow a fixed number of bytes per tick, rather than just one packet. Thus if the rule is 1024 bytes per tick, a single 1024-byte packet can be admitted on a tick, two 512-byte packets, four 256-byte packets, and so on. If the residual byte count is too low, the next packet must wait until the next tick.

Implementing the original leaky bucket algorithm is easy. The leaky bucket consists of a finite queue. When a packet arrives, if there is room on the queue it is appended to the queue; otherwise, it is discarded. At every clock tick, one packet is transmitted (unless the queue is empty).

The byte-counting leaky bucket is implemented almost the same way. At each tick, a counter is initialized to n. If the first packet on the queue has fewer bytes than the current value of the counter, it is transmitted, and the counter is decremented by that number of bytes. Additional packets may also be sent, as long as the counter is high enough. When the counter drops below the length of the next packet on the queue, transmission stops until the next tick, at which time the residual byte count is overwritten and lost.

As an example of a leaky bucket, imagine that a computer can produce data at 25 million bytes/sec (200 Mbps) and that the network also runs at this speed. However, the routers can handle this data rate only for short intervals. For long intervals, they work best at rates not exceeding 2 million bytes/sec. Now suppose data comes in 1-million-byte bursts, one 40-msec burst every second. To reduce the average rate to 2 MB/sec, we could use a leaky bucket with $\rho = 2$ MB/sec and a capacity, C, of 1 MB. This means that bursts of up to 1 MB can be handled without data loss, and that such bursts are spread out over 500 msec, no matter how fast they come in.

In Fig. 5-25(a) we see the input to the leaky bucket running at 25 MB/sec for 40 msec. In Fig. 5-25(b) we see the output draining out at a uniform rate of 2 MB/sec for 500 msec.

The Token Bucket Algorithm

The leaky bucket algorithm enforces a rigid output pattern at the average rate, no matter how bursty the traffic is. For many applications, it is better to allow the output to speed up somewhat when large bursts arrive, so a more flexible algorithm is needed, preferably one that never loses data. One such algorithm is the **token bucket algorithm**. In this algorithm, the leaky bucket holds tokens, generated by a clock at the rate of one token every ΔT sec. In Fig. 5-26(a) we see a bucket holding three tokens, with five packets waiting to be transmitted. For a packet to be transmitted, it must capture and destroy one token. In Fig. 5-26(b)

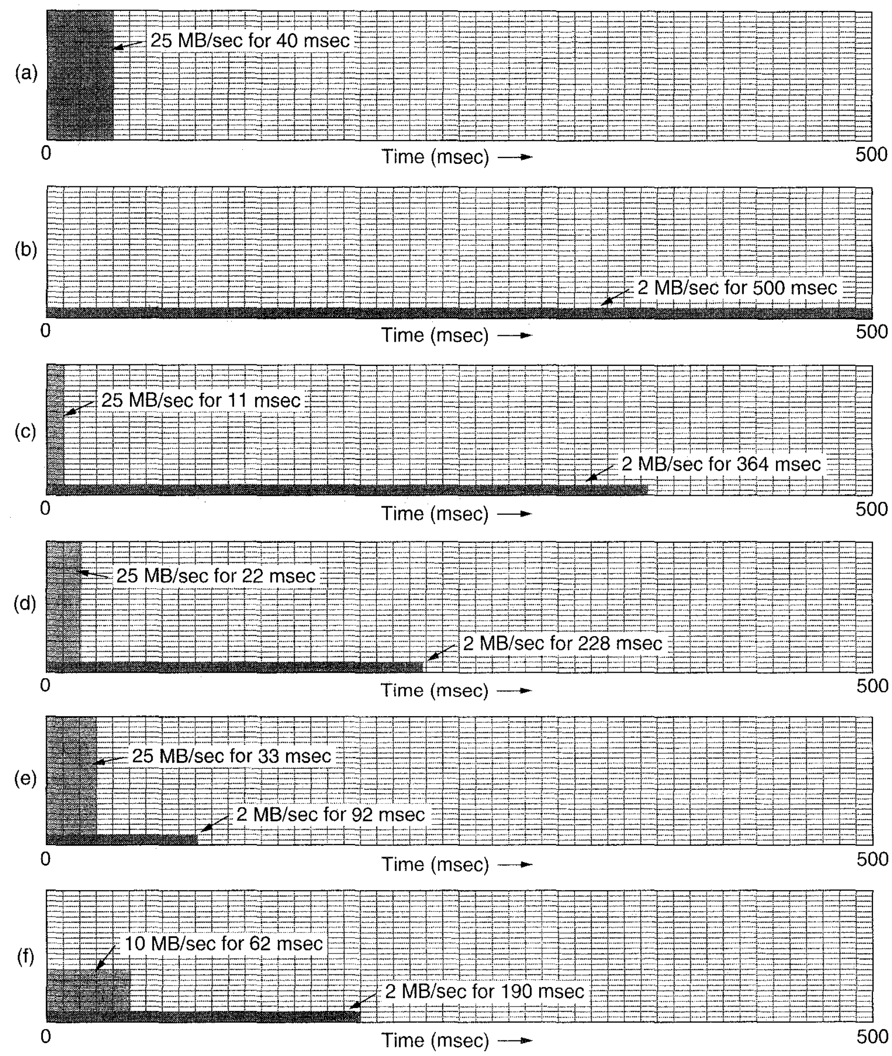


Fig. 5-25. (a) Input to a leaky bucket. (b) Output from a leaky bucket. (c) - (e) Output from a token bucket with capacities of 250KB, 500KB, and 750KB. (f) Output from a 500KB token bucket feeding a 10 MB/sec leaky bucket.

we see that three of the five packets have gotten through, but the other two are stuck waiting for two more tokens to be generated.

The token bucket algorithm provides a different kind of traffic shaping than the leaky bucket algorithm. The leaky bucket algorithm does not allow idle hosts

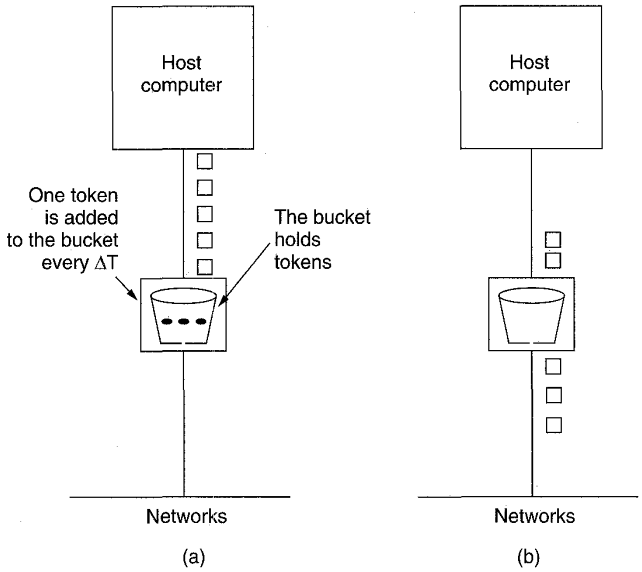


Fig. 5-26. The token bucket algorithm. (a) Before. (b) After.

to save up permission to send large bursts later. The token bucket algorithm does allow saving, up to the maximum size of the bucket, n. This property means that bursts of up to n packets can be sent at once, allowing some burstiness in the output stream and giving faster response to sudden bursts of input.

Another difference between the two algorithms is that the token bucket algorithm throws away tokens when the bucket fills up but never discards packets. In contrast, the leaky bucket algorithm discards packets when the bucket fills up.

Here too, a minor variant is possible, in which each token represents the right to send not one packet, but k bytes. A packet can only be transmitted if enough tokens are available to cover its length in bytes. Fractional tokens are kept for future use.

The leaky bucket and token bucket algorithms can also be used to smooth traffic between routers, as well as being used to regulate host output as in our examples. However, one clear difference is that a token bucket regulating a host can make the host stop sending when the rules say it must. Telling a router to stop sending while its input keeps pouring in may result in lost data.

The implementation of the basic token bucket algorithm is just a variable that counts tokens. The counter is incremented by one every ΔT and decremented by one whenever a packet is sent. When the counter hits zero, no packets may be sent. In the byte-count variant, the counter is increment by k bytes every ΔT and decremented by the length of each packet sent.

Essentially what the token bucket does is allow bursts, but up to a regulated maximum length. Look at Fig. 5-25(c) for example. Here we have a token bucket with a capacity of 250 KB. Tokens arrive at a rate allowing output at 2 MB/sec. Assuming the token bucket is full when the 1-MB burst arrives, the bucket can drain at the full 25 MB/sec for about 11 msec. Then it has to cut back to 2 MB/sec until the entire input burst has been sent.

Calculating the length of the maximum rate burst is slightly tricky. It is not just 1 MB divided by 25 MB/sec because while the burst is being output, more tokens arrive. If we call the burst length S sec, the token bucket capacity C bytes, the token arrival rate ρ bytes/sec, and the maximum output rate M bytes/sec, we see that an output burst contains a maximum of $C + \rho S$ bytes. We also know that the number of bytes in a maximum-speed burst of length S seconds is MS. Hence we have

$$C + \rho S = MS$$

We can solve this equation to get $S = C/(M - \rho)$. For our parameters of $C = 250$ KB, $M = 25$ MB/sec, and $\rho = 2$ MB/sec, we get a burst time of about 11 msec. Figure 5-25(d) and Fig. 5-25(e) show the token bucket for capacities of 500-KB and 750 KB, respectively.

A potential problem with the token bucket algorithm is that it allows large bursts again, even though the maximum burst interval can be regulated by careful selection of ρ and M. Frequently it is desirable to reduce the peak rate, but without going back to the low value of the original leaky bucket.

One way to get smoother traffic is to put a leaky bucket after the token bucket. The rate of the leaky bucket should be higher than the token bucket's ρ but lower than the maximum rate of the network. Figure 5-25(f) shows the output for a 500 KB token bucket followed by a 10-MB/sec leaky bucket.

Policing all these schemes can be a bit tricky. Essentially, the network has to simulate the algorithm and make sure that no more packets or bytes are being sent than are permitted. Excess packets are then discarded or downgraded, as discussed later.

5.3.4. Flow Specifications

Traffic shaping is most effective when the sender, receiver, and subnet all agree to it. To get agreement, it is necessary to specify the traffic pattern in a precise way. Such an agreement is called a **flow specification**. It consists of a data structure that describes both the pattern of the injected traffic and the quality of service desired by the applications. A flow specification can apply either to the packets sent on a virtual circuit, or to a sequence of datagrams sent between a source and a destination (or even to multiple destinations).

In this section we will describe an example flow specification designed by Partridge (1992). It is shown in Fig. 5-27. The idea is that before a connection is established or before a sequence of datagrams are sent, the source gives the flow specification to the subnet for approval. The subnet can either accept it, reject it, or come back with a counterproposal ("I cannot give you 100 msec average delay; can you live with 150 msec?"). Once the sender and subnet have struck a deal, the sender can ask the receiver if it, too, agrees.

Characteristics of the Input	Service Desired
Maximum packet size (bytes)	Loss sensitivity (bytes)
Token bucket rate (bytes/sec)	Loss interval (μsec)
Token bucket size (bytes)	Burst loss sensitivity (packets)
Maximum transmission rate (bytes/sec)	Minimum delay noticed (μsec)
	Maximum delay variation (μsec)
	Quality of guarantee

Fig. 5-27. An example flow specification.

Let us now examine the parameters of our example flow specification starting with the traffic specification. The *Maximum packet size* tells how big packets may be. The next two parameters implicitly assume that traffic will be shaped by the token bucket algorithm working in bytes. They tell how many bytes are put into the token bucket per second, and how big the bucket is. If the rate is r bytes/sec and the bucket size is b bytes, then in any arbitrary time interval Δt, the maximum number of bytes that may be sent is $b + r\Delta t$. Here the first term represents the maximum possible contents of the bucket at the start of the interval and the second one represents the new credits that come in during the interval. The *Maximum transmission rate* is the top rate the host is capable of producing under any conditions and implicitly specifies the shortest time interval in which the token bucket could be emptied.

The second column specifies what the application wants from the subnet. The first and second parameters represent the numerator and denominator of a fraction giving the maximum acceptable loss rate (e.g., 1 byte per hour). Alternatively, they can indicate that the flow is insensitive to packet loss. The *Burst loss sensitivity* tells how many consecutive lost packets can be tolerated.

The next two service parameters deal with delay. The *Minimum delay noticed* says how long a packet can be delayed without the application noticing. For a file transfer, it might be a second, but for an audio stream 3 msec might be the limit. The *Maximum delay variation* tries to quantify the fact that some applications are not sensitive to the actual delay but are highly sensitive to the **jitter**, that is, the amount of variation in the end-to-end packet transit time. It is two times the number of microseconds a packet's delay may vary from the average. Thus a value of 2000 means that a packet may be up to 1 msec early or late, but no more.

Finally, the *Quality of guarantee* indicates whether or not the application really means it. On the one hand, the loss and delay characteristics might be ideal goals, but no harm is done if they are not met. On the other hand, they might be so important that if they cannot be met, the application simply terminates. Intermediate positions are also possible.

Although we have looked at the flow specification as a request from the application to the subnet, it can also be a return value telling what the subnet can do. Thus it can potentially be used for an extended negotiation about the service level.

A problem inherent with any flow specification is that the application may not know what it really wants. For example, an application program running in New York might be quite happy with a delay of 200 msec to Sydney, but most unhappy with the same 200-msec delay to Boston. Here the "minimum service" is clearly a function of what is thought to be possible.

5.3.5. Congestion Control in Virtual Circuit Subnets

The congestion control methods described above are basically open loop: they try to prevent congestion from occurring in the first place, rather than dealing with it after the fact. In this section we will describe some approaches to dynamically controlling congestion in virtual circuit subnets. In the next two, we will look at techniques that can be used in any subnet.

One technique that is widely used to keep congestion that has already started from getting worse is **admission control**. The idea is simple: once congestion has been signaled, no more virtual circuits are set up until the problem has gone away. Thus, attempts to set up new transport layer connections fail. Letting more people in just makes matters worse. While this approach is crude, it is simple and easy to carry out. In the telephone system, when a switch gets overloaded, it also practices admission control, by not giving dial tones.

An alternative approach is to allow new virtual circuits but carefully route all new virtual circuits around problem areas. For example, consider the subnet of Fig. 5-28(a), in which two routers are congested, as indicated.

Suppose that a host attached to router *A* wants to set up a connection to a host attached to router *B*. Normally, this connection would pass through one of the congested routers. To avoid this situation, we can redraw the subnet as shown in Fig. 5-28(b), omitting the congested routers and all of their lines. The dashed line shows a possible route for the virtual circuit that avoids the congested routers.

Another strategy relating to virtual circuits is to negotiate an agreement between the host and subnet when a virtual circuit is set up. This agreement normally specifies the volume and shape of the traffic, quality of service required, and other parameters. To keep its part of the agreement, the subnet will typically reserve resources along the path when the circuit is set up. These resources can include table and buffer space in the routers and bandwidth on the lines. In this

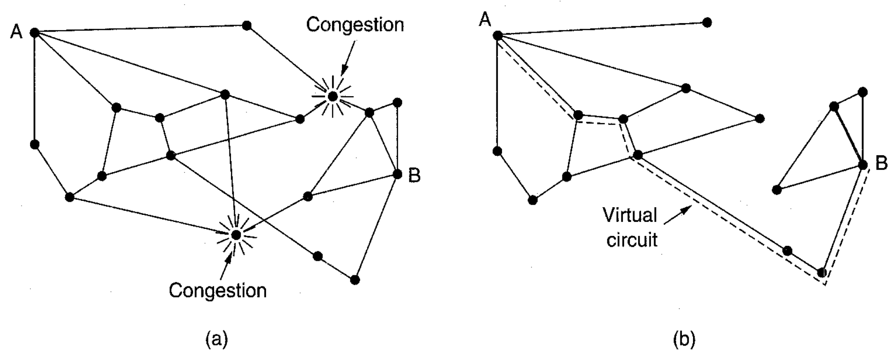


Fig. 5-28. (a) A congested subnet. (b) A redrawn subnet that eliminates the congestion and a virtual circuit from A to B.

way, congestion is unlikely to occur on the new virtual circuits because all the necessary resources are guaranteed to be available.

This kind of reservation can be done all the time as standard operating procedure, or only when the subnet is congested. A disadvantage of doing it all the time is that it tends to waste resources. If six virtual circuits that might use 1 Mbps all pass through the same physical 6-Mbps line, the line has to be marked as full, even though it may rarely happen that all six virtual circuits are transmitting at the same time. Consequently, the price of the congestion control is unused bandwidth.

5.3.6. Choke Packets

Let us now turn to an approach that can be used in both virtual circuit and datagram subnets. Each router can easily monitor the utilization of its output lines and other resources. For example, it can associate with each line a real variable, u, whose value, between 0.0 and 1.0, reflects the recent utilization of that line. To maintain a good estimate of u, a sample of the instantaneous line utilization, f (either 0 or 1), can be made periodically and u updated according to

$$u_{\text{new}} = au_{\text{old}} + (1 - a)f$$

where the constant a determines how fast the router forgets recent history.

Whenever u moves above the threshold, the output line enters a "warning" state. Each newly arriving packet is checked to see if its output line is in warning state. If so, the router sends a **choke packet** back to the source host, giving it the destination found in the packet. The original packet is tagged (a header bit is turned on) so that it will not generate any more choke packets further along the path and is then forwarded in the usual way.

When the source host gets the choke packet, it is required to reduce the traffic sent to the specified destination by X percent. Since other packets aimed at the same destination are probably already under way and will generate yet more choke packets, the host should ignore choke packets referring to that destination for a fixed time interval. After that period has expired, the host listens for more choke packets for another interval. If one arrives, the line is still congested, so the host reduces the flow still more and begins ignoring choke packets again. If no choke packets arrive during the listening period, the host may increase the flow again. The feedback implicit in this protocol can help prevent congestion yet not throttle any flow unless trouble occurs.

Hosts can reduce traffic by adjusting their policy parameters, for example, window size or leaky bucket output rate. Typically, the first choke packet causes the data rate to be reduced to 0.50 of its previous rate, the next one causes a reduction to 0.25, and so on. Increases are done in smaller increments to prevent congestion from reoccurring quickly.

Several variations on this congestion control algorithm have been proposed. For one, the routers can maintain several thresholds. Depending on which threshold has been crossed, the choke packet can contain a mild warning, a stern warning, or an ultimatum.

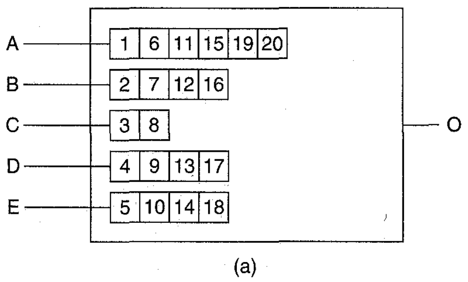
Another variation is to use queue lengths or buffer utilization instead of line utilization as the trigger signal. The same exponential weighting can be used with this metric as with u, of course.

Weighted Fair Queueing

A problem with using choke packets is that the action to be taken by the source hosts is voluntary. Suppose that a router is being swamped by packets from four sources, and it sends choke packets to all of them. One of them cuts back, as it is supposed to, but the other three just keep blasting away. The result is that the honest host gets an even smaller share of the bandwidth than it had before.

To get around this problem, and thus make compliance more attractive, Nagle (1987) proposed a **fair queueing** algorithm. The essence of the algorithm is that routers have multiple queues for each output line, one for each source. When a line becomes idle, the router scans the queues round robin, taking the first packet on the next queue. In this way, with n hosts competing for a given output line, each host gets to send one out of every n packets. Sending more packets will not improve this fraction. Some ATM switches use this algorithm.

Although a start, the algorithm has a problem: it gives more bandwidth to hosts that use large packets than to hosts that use small packets. Demers et al. (1990) suggested an improvement in which the round robin is done in such a way as to simulate a byte-by-byte round robin, instead of a packet-by-packet round robin. In effect, it scans the queues repeatedly, byte-for-byte, until it finds the tick on which each packet will be finished. The packets are then sorted in order of their finishing and sent in that order. The algorithm is illustrated in Fig. 5-29.



Packet	Finishing time
C	8
B	16
D	17
E	18
A	20

(b)

Fig. 5-29. (a) A router with five packets queued for line *O*. (b) Finishing times for the five packets.

In Fig. 5-29(a) we see packets of length 2 to 6 bytes. At (virtual) clock tick 1, the first byte of the packet on line *A* is sent. Then goes the first byte of the packet on line *B*, and so on. The first packet to finish is *C*, after eight ticks. The sorted order is given in Fig. 5-29(b). In the absence of new arrivals, the packets will be sent in the order listed, from *C* to *A*.

One problem with this algorithm is that it gives all hosts the same priority. In many situations, it is desirable to give the file and other servers more bandwidth than clients, so they can be given two or more bytes per tick. This modified algorithm is called **weighted fair queueing** and is widely used. Sometimes the weight is equal to the number of virtual circuits or flows coming out of a machine, so each process gets equal bandwidth. An efficient implementation of the algorithm is discussed in (Shreedhar and Varghese, 1995).

Hop-by-Hop Choke Packets

At high speeds and over long distances, sending a choke packet to the source hosts does not work well because the reaction is so slow. Consider, for example, a host in San Francisco (router *A* in Fig. 5-30) that is sending traffic to a host in New York (router *D* in Fig. 5-30) at 155 Mbps. If the New York host begins to run out of buffers, it will take about 30 msec for a choke packet to get back to San Francisco to tell it to slow down. The choke packet propagation is shown as the second, third, and fourth steps in Fig. 5-30(a). In those 30 msec, another 4.6 megabits (e.g., over 10,000 ATM cells) will have been sent. Even if the host in San Francisco completely shuts down immediately, the 4.6 megabits in the pipe will continue to pour in and have to be dealt with. Only in the seventh diagram in Fig. 5-30(a) will the New York router notice a slower flow.

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An alternative approach is to have the choke packet take effect at every hop it passes through, as shown in the sequence of Fig. 5-30(b). Here, as soon as the choke packet reaches *F*, *F* is required to reduce the flow to *D*. Doing so will require *F* to devote more buffers to the flow, since the source is still sending away at full blast, but it gives *D* immediate relief, like a headache remedy in a television commercial. In the next step, the choke packet reaches *E*, which tells *E* to reduce the flow to *F*. This action puts a greater demand on *E*'s buffers but gives *F* immediate relief. Finally, the choke packet reaches *A* and the flow genuinely slows down.

The net effect of this hop-by-hop scheme is to provide quick relief at the point of congestion at the price of using up more buffers upstream. In this way congestion can be nipped in the bud without losing any packets. The idea is discussed in more detail and simulation results are given in (Mishra and Kanakia, 1992).

5.3.7. Load Shedding

When none of the above methods make the congestion disappear, routers can bring out the heavy artillery: load shedding. **Load shedding** is a fancy way of saying that when routers are being inundated by packets that they cannot handle, they just throw them away. The term comes from the world of electrical power generation where it refers to the practice of utilities intentionally blacking out certain areas to save the entire grid from collapsing on hot summer days when the demand for electricity greatly exceeds the supply.

A router drowning in packets can just pick packets at random to drop, but usually it can do better than that. Which packet to discard may depend on the applications running. For file transfer, an old packet is worth more than a new one because dropping packet 6 and keeping packets 7 through 10 will cause a gap at the receiver that may force packets 6 through 10 to be retransmitted (if the receiver routinely discards out-of-order packets). In a 12-packet file, dropping 6 may require 7 through 12 to be retransmitted, whereas dropping 10 may require only 10 through 12 to be retransmitted. In contrast, for multimedia, a new packet is more important than an old one. The former policy (old is better than new) is often called **wine** and the latter (new is better than old) is often called **milk**.

A step above this in intelligence requires cooperation from the senders. For many applications, some packets are more important than others. For example, certain algorithms for compressing video periodically transmit an entire frame and then send subsequent frames as differences from the last full frame. In this case, dropping a packet that is part of a difference is preferable to dropping one that is part of a full frame. As another example, consider transmitting a document containing ASCII text and pictures. Losing a line of pixels in some image is far less damaging than losing a line of readable text.

To implement an intelligent discard policy, applications must mark their packets in priority classes to indicate how important they are. If they do this, when

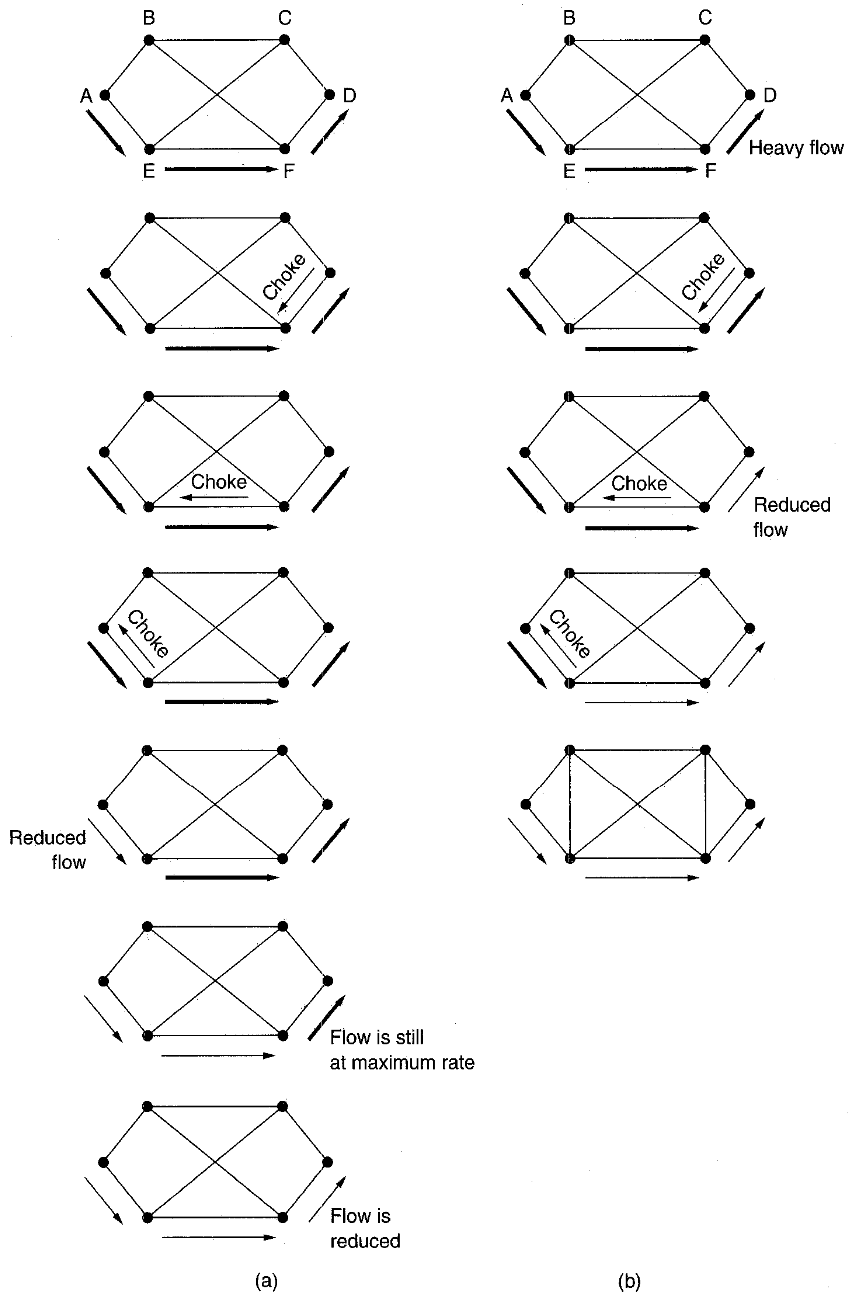

Fig. 5-30. (a) A choke packet that affects only the source. (b) A choke packet that affects each hop it passes through.

packets have to be discarded, routers can first drop packets from the lowest class, then the next lowest class, and so on. Of course, unless there is some significant incentive to mark packets as anything other than VERY IMPORTANT—NEVER, EVER DISCARD, nobody will do it.

The incentive might be in the form of money, with the low-priority packets being cheaper to send than the high-priority ones. Alternatively, priority classes could be coupled with traffic shaping. For example, there might be a rule saying that when the token bucket algorithm is being used and a packet arrives at a moment when no token is available, it may still be sent, provided that it is marked as the lowest possible priority, and thus subject to discard the instant trouble appears. Under conditions of light load, users might be happy to operate in this way, but as the load increases and packets actually begin to be discarded, they might cut back and only send packets when tokens are available.

Another option is to allow hosts to exceed the limits specified in the agreement negotiated when the virtual circuit was set up (e.g., use a higher bandwidth than allowed), but subject to the condition that all excess traffic be marked as low priority. Such a strategy is actually not a bad idea, because it makes more efficient use of idle resources, allowing hosts to use them as long as nobody else is interested, but without establishing a right to them when times get tough.

Marking packets by class requires one or more header bits in which to put the priority. ATM cells have 1 bit reserved in the header for this purpose, so every ATM cell is labeled either as low priority or high priority. ATM switches indeed use this bit when making discard decisions.

In some networks, packets are grouped together into larger units that are used for retransmission purposes. For example, in ATM networks, what we have been calling "packets" are fixed-length cells. These cells are just fragments of "messages." When a cell is dropped, ultimately the entire "message" will be retransmitted, not just the missing cell. Under these conditions, a router that drops a cell might as well drop all the rest of the cells in that message, since transmitting them costs bandwidth and wins nothing—even if they get through they will still be retransmitted later.

Simulation results show that when a router senses trouble on the horizon, it is better off starting to discard packets early, rather than wait until it becomes completely clogged up (Floyd and Jacobson, 1993; Romanow and Floyd, 1994). Doing so may prevent the congestion from getting a foothold.

5.3.8. Jitter Control

For applications such as audio and video transmission, it does not matter much if the packets take 20 msec or 30 msec to be delivered, as long as the transit time is constant. Having some packets taking 20 msec and others taking 30 msec will give an uneven quality to the sound or image. Thus the agreement might be that 99 percent of the packets be delivered with a delay in the range of 24.5 msec to 25.5 msec. The mean value chosen must be feasible, of course. In other words, an average amount of congestion must be calculated in.

The jitter can be bounded by computing the expected transit time for each hop along the path. When a packet arrives at a router, the router checks to see how much the packet is behind or ahead of its schedule. This information is stored in the packet and updated at each hop. If the packet is ahead of schedule, it is held just long enough to get it back on schedule. If it is behind schedule, the router tries to get it out the door quickly. In fact, the algorithm for determining which of several packets competing for an output line should go next can always choose the packet furthest behind in its schedule. In this way, packets that are ahead of schedule get slowed down and packets that are behind schedule get speeded up, in both cases reducing the amount of jitter.

5.3.9. Congestion Control for Multicasting

All of the congestion control algorithms discussed so far deal with messages from a single source to a single destination. In this section we will describe a way of managing multicast flows from multiple sources to multiple destinations. For example, imagine several closed-circuit television stations transmitting audio and video streams to a group of receivers, each of whom can view one or more stations at once and are free to switch from station to station at will. An application of this technology might be a video conference, in which each participant could focus on the current speaker or on the boss' expression, as desired.

In many multicast applications, groups can change membership dynamically, for example, as people enter a video conference or get bored and switch to a soap opera. Under these conditions, the approach of having the senders reserve bandwidth in advance does not work well, as it would require each sender to track all entries and exits of its audience and regenerate the spanning tree at each change. For a system designed to transmit cable television, with millions of subscribers, it would not work at all.

RSVP—Resource reSerVation Protocol

One interesting solution that can handle this environment is the **RSVP** protocol (Zhang et al., 1993). It allows multiple senders to transmit to multiple groups of receivers, permits individual receivers to switch channels freely, and optimizes bandwidth use while at the same time eliminating congestion.

In its simplest form, the protocol uses multicast routing using spanning trees, as discussed earlier. Each group is assigned a group address. To send to a group, a sender puts the group's address in its packets. The standard multicast routing algorithm then builds a spanning tree covering all group members. The routing algorithm is not part of RSVP. The only difference with normal multicasting is a little extra information that is multicast to the group periodically to tell the routers along the tree to maintain certain data structures in their memories.

As an example, consider the network of Fig. 5-31(a). Hosts 1 and 2 are multicast senders, and hosts 3, 4, and 5 are multicast receivers. In this example, the senders and receivers are disjoint, but in general, the two sets may overlap. The multicast trees for hosts 1 and 2 are shown in Fig. 5-31(b) and Fig. 5-31(c), respectively.

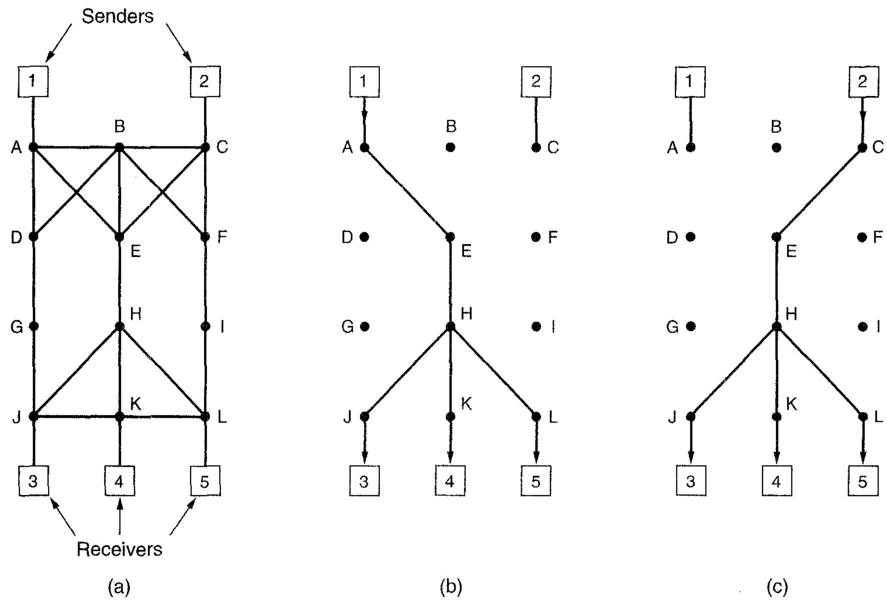

Fig. 5-31. (a) A network. (b) The multicast spanning tree for host 1. (c) The multicast spanning tree for host 2.

To get better reception and eliminate congestion, any of the receivers in a group can send a reservation message up the tree to the sender. The message is propagated using the reverse path forwarding algorithm discussed earlier. At each hop, the router notes the reservation and reserves the necessary bandwidth. If insufficient bandwidth is available, it reports back failure. By the time the message gets back to the source, bandwidth has been reserved all the way from the sender to the receiver making the reservation request along the spanning tree.

An example of such a reservation is shown in Fig. 5-32(a). Here host 3 has requested a channel to host 1. Once it has been established, packets can flow from 1 to 3 without congestion. Now consider what happens if host 3 next reserves a channel to the other sender, host 2, so the user can watch two television programs at once. A second path is reserved, as illustrated in Fig. 5-32(b). Note that two separate channels are needed from host 3 to router *E* because two independent streams are being transmitted.

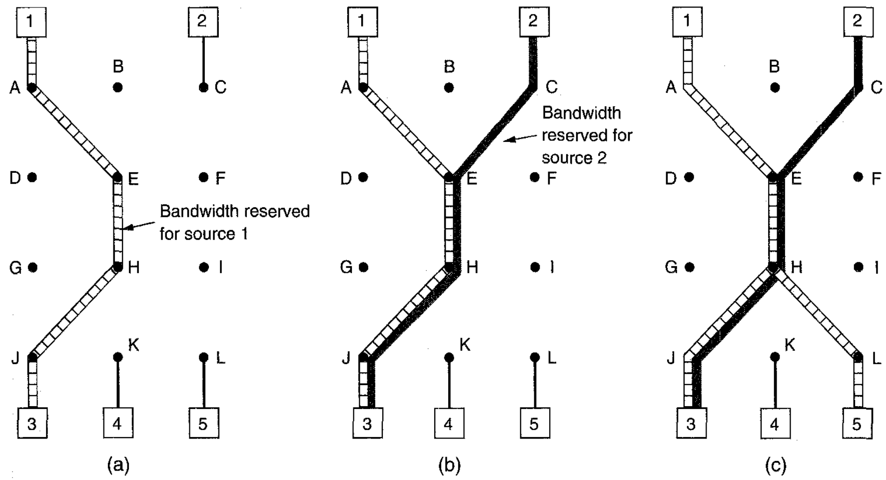

Fig. 5-32. (a) Host 3 requests a channel to host 1. (b) Host 3 then requests a second channel, to host 2. (c) Host 5 requests a channel to host 1.

Finally, in Fig. 5-32(c), host 5 decides to watch the program being transmitted by host 1 and also makes a reservation. First, dedicated bandwidth is reserved as far as router *H*. However, this router sees that it already has a feed from host 1, so if the necessary bandwidth has already been reserved, it does not have to reserve any more. Note that hosts 3 and 5 might have asked for different amounts of bandwidth (e.g., 3 has a black-and-white television set, so it does not want the color information), so the capacity reserved must be large enough to satisfy the greediest receiver.

When making a reservation, a receiver can (optionally) specify one or more sources that it wants to receive from. It can also specify whether these choices are fixed for the duration of the reservation, or whether the receiver wants to keep open the option of changing sources later. The routers use this information to optimize bandwidth planning. In particular, two receivers are only set up to share a path if they both agree not to change sources later on.

The reason for this strategy in the fully dynamic case is that reserved bandwidth is decoupled from the choice of source. Once a receiver has reserved bandwidth, it can switch to another source and keep that portion of the existing path that is valid for the new source. If host 2 is transmitting several video streams, for example, host 3 may switch between them at will without changing its reservation: the routers do not care what program the receiver is watching.

5.4. INTERNETWORKING

Up until now, we have implicitly assumed that there is a single homogeneous network, with each machine using the same protocol in each layer. Unfortunately, this assumption is wildly optimistic. Many different networks exist, including LANs, MANs, and WANs. Numerous protocols are in widespread use in every layer. In the following sections we will take a careful look at the issues that arise when two or more networks are together to form an **internet**.

Considerable controversy exists about the question of whether today's abundance of network types is a temporary condition that will go away as soon as everyone realizes how wonderful [fill in your favorite network] is, or whether it is an inevitable, but permanent feature of the world that is here to stay. Having different networks invariably means having different protocols.

We believe that a variety of different networks (and thus protocols) will always be around, for the following reasons. First of all, the installed base of different networks is large and growing. Nearly all UNIX shops run TCP/IP. Many large businesses still have mainframes running SNA. DEC is still developing DECnet. Personal computer LANs often use Novell NCP/IPX or AppleTalk. ATM systems are starting to be widespread. Finally, specialized protocols are often used on satellite, cellular, and infrared networks. This trend will continue for years due to the large number of existing networks and because not all vendors perceive it in their interest for their customers to be able to easily migrate to another vendor's system.

Second, as computers and networks get cheaper, the place where decisions get made moves downward. Many companies have a policy to the effect that purchases costing over a million dollars have to be approved by top management, purchases costing over 100,000 dollars have to be approved by middle management, but purchases under 100,000 dollars can be made by department heads without any higher approval. This can easily lead to the accounting department installing an Ethernet, the engineering department installing a token bus, and the personnel department installing a token ring.

Third, different networks (e.g., ATM and wireless) have radically different technology, so it should not be surprising that as new hardware developments occur, new software will be created to fit the new hardware. For example, the average home now is like the average office ten years ago: it is full of computers that do not talk to one another. In the future, it may be commonplace for the telephone, the television set, and other appliances all to be networked together, so they can be controlled remotely. This new technology will undoubtedly bring new protocols.

As an example of how different networks interact, consider the following example. At most universities, the computer science and electrical engineering departments have their own LANs, often different. In addition, the university computer center often has a mainframe and supercomputer, the former for faculty members in the humanities who do not wish to get into the computer maintenance business, and the latter for physicists who want to crunch numbers. As a consequence of these various networks and facilities, the following scenarios are easy to imagine:

1. LAN-LAN: A computer scientist downloading a file to engineering.
2. LAN-WAN: A computer scientist sending mail to a distant physicist.
3. WAN-WAN: Two poets exchanging sonnets.
4. LAN-WAN-LAN: Engineers at different universities communicating.

Figure 5-33 illustrates these four types of connections as dotted lines. In each case, it is necessary to insert a "black box" at the junction between two networks, to handle the necessary conversions as packets move from one network to the other.

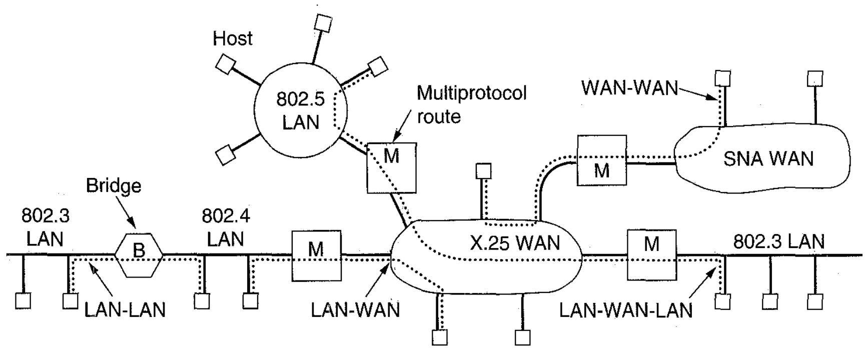

Fig. 5-33. Network interconnection.

The name used for the black box connecting two networks depends on the layer that does the work. Some common names are given below (although there is not much agreement on terminology in this area).

Layer 1: Repeaters copy individual bits between cable segments.

Layer 2: Bridges store and forward data link frames between LANs.

Layer 3: Multiprotocol routers forward packets between dissimilar networks.

Layer 4: Transport gateways connect byte streams in the transport layer.

Above 4: Application gateways allow interworking above layer 4.

For convenience, we will sometimes use the term "gateway" to mean any device that connects two or more dissimilar networks.

Repeaters are low-level devices that just amplify or regenerate weak signals. They are needed to provide current to drive long cables. In 802.3, for example, the timing properties of the MAC protocol (the value of τ chosen) allow cables up to 2.5 km, but the transceiver chips can only provide enough power to drive 500 meters. The solution is to use repeaters to extend the cable length where that is desired.

Unlike repeaters, which copy the bits as they arrive, **bridges** are store-and-forward devices. A bridge accepts an entire frame and passes it up to the data link layer where the checksum is verified. Then the frame is sent down to the physical layer for forwarding on a different network. Bridges can make minor changes to the frame before forwarding it, such as adding or deleting some fields from the frame header. Since they are data link layer devices, they do not deal with headers at layer 3 and above and cannot make changes or decisions that depend on them.

Multiprotocol routers are conceptually similar to bridges, except that they are found in the network layer. They just take incoming packets from one line and forward them on another, just as all routers do, but the lines may belong to different networks and use different protocols (e.g., IP, IPX, and the OSI connectionless packet protocol, CLNP). Like all routers, multiprotocol routers operate at the level of the network layer.

Transport gateways make a connection between two networks at the transport layer. We will discuss this possibility later when we come to concatenated virtual circuits.

Finally, **application gateways** connect two parts of an application in the application layer. For example, to send mail from an Internet machine using the Internet mail format to an ISO MOTIS mailbox, one could send the message to a mail gateway. The mail gateway would unpack the message, convert it to MOTIS format, and then forward it on the second network using the network and transport protocols used there.

When a gateway is between two WANs run by different organizations, possibly in different countries, the joint operation of one workstation-class machine can lead to a lot of finger pointing. To eliminate these problems, a slightly different approach can be taken. The gateway is effectively ripped apart in the middle and the two parts are connected with a wire. Each of the halves is called a **half-gateway** and each one is owned and operated by one of the network operators. The whole problem of gatewaying then reduces to agreeing to a common protocol to use on the wire, one that is neutral and does not favor either party. Figure 5-34 shows both full and half-gateways. Either kind can be used in any layer (e.g., half-bridges also exist).

That all said, the situation is murkier in practice than it is in theory. Many devices on the market combine bridge and router functionality. The key property of a pure bridge is that it examines data link layer frame headers and does not inspect or modify the network layer packets inside the frames. A bridge cannot

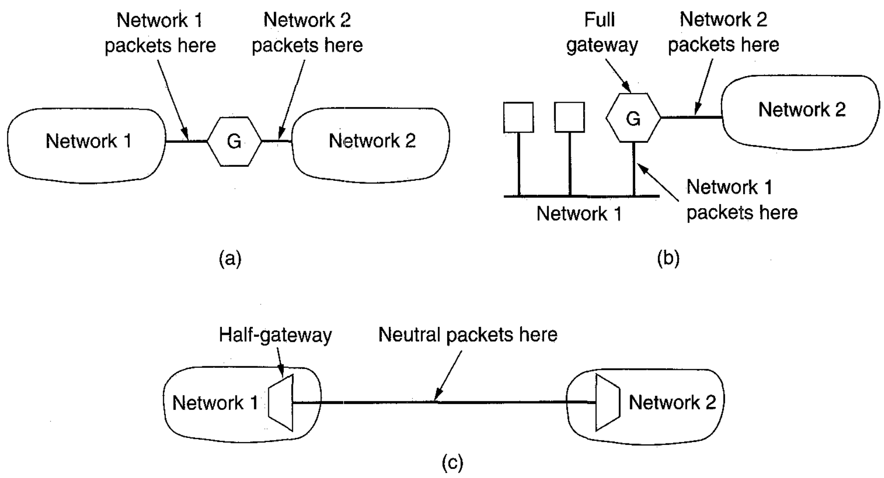

Fig. 5-34. (a) A full gateway between two WANs. (b) A full gateway between a LAN and a WAN. (c) Two half-gateways.

tell and does not care whether the frame it is forwarding from an 802.x LAN to an 802.y contains an IP, IPX, or CLNP packet in the payload field.

A router, in contrast, knows very well whether it is an IP router, an IPX router, a CLNP router, or all three combined. It examines these headers and makes decisions based on the addresses found there. On the other hand, when a pure router hands off a packet to the data link layer, it does not know or care whether it will be carried in an Ethernet frame or a token ring frame. That is the data link layer's responsibility.

The confusion in the industry comes from two sources. First, functionally, bridges and routers are not all that different. They each accept incoming PDUs (Protocol Data Units), examine some header fields, and make decisions about where to send the PDUs based on header information and internal tables.

Second, many commercial products are sold under the wrong label or combine the functionality of both bridges and routers. For example, source routing bridges are not really bridges at all, since they involve a protocol layer above the data link layer to do their job. For an illuminating discussion of bridges versus routers, see Chap. 12 of (Perlman, 1992).

5.4.1. How Networks Differ

Networks can differ in many ways. In Fig. 5-35 we list some of the differences that can occur in the network layer. It is papering over these differences that makes internetworking more difficult than operating within a single network.

Item	Some Possibilities
Service offered	Connection-oriented versus connectionless
Protocols	IP, IPX, CLNP, AppleTalk, DECnet, etc.
Addressing	Flat (802) versus hierarchical (IP)
Multicasting	Present or absent (also broadcasting)
Packet size	Every network has its own maximum
Quality of service	May be present or absent; many different kinds
Error handling	Reliable, ordered, and unordered delivery
Flow control	Sliding window, rate control, other, or none
Congestion control	Leaky bucket, choke packets, etc.
Security	Privacy rules, encryption, etc.
Parameters	Different timeouts, flow specifications, etc.
Accounting	By connect time, by packet, by byte, or not at all

Fig. 5-35. Some of the many ways networks can differ.

When packets sent by a source on one network must transit one or more foreign networks before reaching the destination network (which also may be different from the source network), many problems can occur at the interfaces between networks. To start with, when packets from a connection-oriented network must transit a connectionless one, they may be reordered, something the sender does not expect and the receiver is not prepared to deal with. Protocol conversions will often be needed, which can be difficult if the required functionality cannot be expressed. Address conversions will also be needed, which may require some kind of directory system. Passing multicast packets through a network that does not support multicasting requires generating separate packets for each destination.

The differing maximum packet sizes used by different networks is a major headache. How do you pass an 8000-byte packet through a network whose maximum size is 1500 bytes? Differing qualities of service is an issue when a packet that has real-time delivery constraints passes through a network that does offer any real-time guarantees.

Error, flow, and congestion control frequently differ among different networks. If the source and destination both expect all packets to be delivered in sequence without error, yet an intermediate network just discards packets whenever it smells congestion on the horizon, or packets can wander around aimlessly for a while and then suddenly emerge and be delivered, many applications will break. Different security mechanisms, parameter settings, and accounting rules, and even national privacy laws also can cause problems.

5.4.2. Concatenated Virtual Circuits

Two styles of internetworking are common: a connection-oriented concatenation of virtual circuit subnets, and a datagram internet style. We will now examine these in turn. In the concatenated virtual circuit model, shown in Fig. 5-36, a connection to a host in a distant network is set up in a way similar to the way connections are normally established. The subnet sees that the destination is remote and builds a virtual circuit to the router nearest the destination network. Then it constructs a virtual circuit from that router to an external "gateway" (multiprotocol router). This gateway records the existence of the virtual circuit in its tables and proceeds to build another virtual circuit to a router in the next subnet. This process continues until the destination host has been reached.

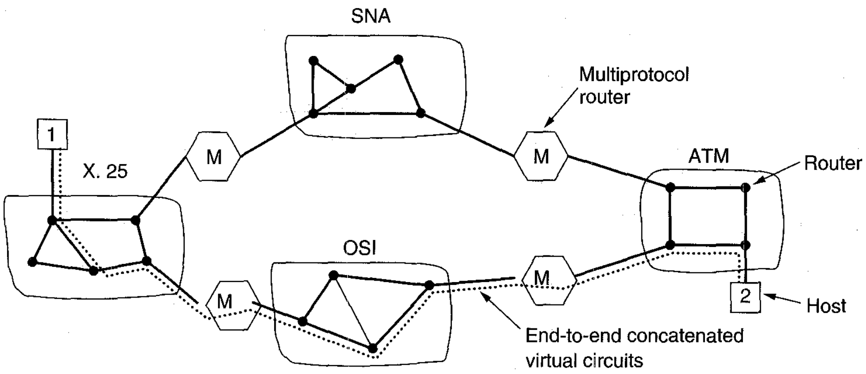

Fig. 5-36. Internetworking using concatenated virtual circuits.

Once data packets begin flowing along the path, each gateway relays incoming packets, converting between packet formats and virtual circuit numbers as needed. Clearly, all data packets must traverse the same sequence of gateways, and thus arrive in order.

The essential feature of this approach is that a sequence of virtual circuits is set up from the source through one or more gateways to the destination. Each gateway maintains tables telling which virtual circuits pass through it, where they are to be routed, and what the new virtual circuit number is.

Although Fig. 5-36 shows the connection made with a full gateway, it could equally well be done with half-gateways.

This scheme works best when all the networks have roughly the same properties. For example, if all of them guarantee reliable delivery of network layer packets, then barring a crash somewhere along the route, the flow from source to destination will also be reliable. Similarly, if none of them guarantee reliable delivery, then the concatenation of the virtual circuits is not reliable either. On the other hand, if the source machine is on a network that does guarantee reliable delivery, but one of the intermediate networks can lose packets, the concatenation has fundamentally changed the nature of the service.

Concatenated virtual circuits are also common in the transport layer. In particular, it is possible to build a bit pipe using, say, OSI, which terminates in a gateway, and have a TCP connection go from the gateway to the next gateway. In this manner, an end-to-end virtual circuit can be built spanning different networks and protocols.

5.4.3. Connectionless Internetworking

The alternative internetwork model is the datagram model, shown in Fig. 5-37. In this model, the only service the network layer offers to the transport layer is the ability to inject datagrams into the subnet and hope for the best. There is no notion of a virtual circuit at all in the network layer, let alone a concatenation of them. This model does not require all packets belonging to one connection to traverse the same sequence of gateways. In Fig. 5-37 datagrams from host 1 to host 2 are shown taking different routes through the internetwork. A routing decision is made separately for each packet, possibly depending on the traffic at the moment the packet is sent. This strategy can use multiple routes and thus achieve a higher bandwidth than the concatenated virtual circuit model. On the other hand, there is no guarantee that the packets arrive at the destination in order, assuming that they arrive at all.

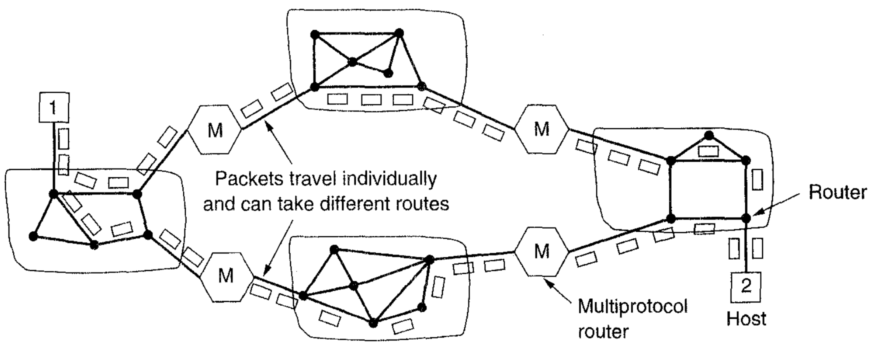

Fig. 5-37. A connectionless internet.

The model of Fig. 5-37 is not quite as simple as it looks. For one thing, if each network has its own network layer protocol, it is not possible for a packet from one network to transit another one. One could imagine the multiprotocol routers actually trying to translate from one format to another, but unless the two formats are close relatives with the same information fields, such conversions will always be incomplete and often doomed to failure. For this reason, conversion is rarely attempted.

A second, and more serious problem, is addressing. Imagine a simple case: a host on the Internet is trying to send an IP packet to a host on an adjoining OSI host. The OSI datagram protocol, CLNP, was based on IP and is close enough to it that a conversion might well work. The trouble is that IP packets all carry the 32-bit Internet address of the destination host in a header field. OSI hosts do not have 32-bit Internet addresses. They use decimal addresses similar to telephone numbers.

To make it possible for the multiprotocol router to convert between formats, someone would have to assign a 32-bit Internet address to each OSI host. Taken to the limit, this approach would mean assigning an Internet address to every machine in the world that an Internet host might want to talk to. It would also mean assigning an OSI address to every machine in the world that an OSI host might want to talk to. The same problem occurs with every other address space (SNA, AppleTalk, etc.). The problems here are insurmountable. In addition, someone would have to maintain a database mapping everything to everything.

Another idea is to design a universal "internet" packet and have all routers recognize it. This approach is, in fact, what IP is—a packet designed to be carried through many networks. The only problem is that IPX, CLNP, and other "universal" packets exist too, making all of them less than universal. Getting everybody to agree to a single format is just not possible.

Let us now briefly recap the two ways internetworking can be attacked. The concatenated virtual circuit model has essentially the same advantages as using virtual circuits within a single subnet: buffers can be reserved in advance, sequencing can be guaranteed, short headers can be used, and the troubles caused by delayed duplicate packets can be avoided.

It also has the same disadvantages: table space required in the routers for each open connection, no alternate routing to avoid congested areas, and vulnerability to router failures along the path. It also has the disadvantage of being difficult, if not impossible, to implement if one of the networks involved is an unreliable datagram network.

The properties of the datagram approach to internetworking are the same as those of datagram subnets: more potential for congestion, but also more potential for adapting to it, robustness in the face of router failures, and longer headers needed. Various adaptive routing algorithms are possible in an internet, just as they are within a single datagram network.

A major advantage of the datagram approach to internetworking is that it can be used over subnets that do not use virtual circuits inside. Many LANs, mobile networks (e.g., aircraft and naval fleets), and even some WANs fall into this category. When an internet includes one of these, serious problems occur if the internetworking strategy is based on virtual circuits.

5.4.4. Tunneling

Handling the general case of making two different networks interwork is exceedingly difficult. However, there is a common special case that is manageable. This case is where the source and destination hosts are on the same type of network, but there is a different network in between. As an example, think of an international bank with a TCP/IP based Ethernet in Paris, a TCP/IP based Ethernet in London, and a PTT WAN in between, as shown in Fig. 5-38.

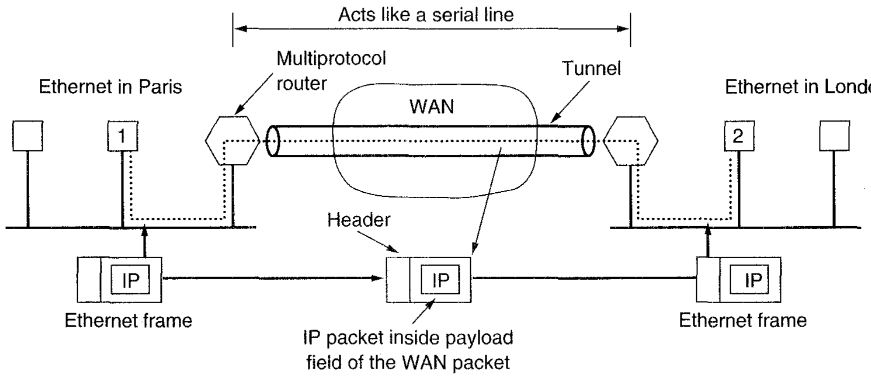

Fig. 5-38. Tunneling a packet from Paris to London.

The solution to this problem is a technique called **tunneling**. To send an IP packet to host 2, host 1 constructs the packet containing the IP address of host 2, inserts it into an Ethernet frame addressed to the Paris multiprotocol router, and puts it on the Ethernet. When the multiprotocol router gets the frame, it removes the IP packet, inserts it in the payload field of the WAN network layer packet, and addresses the latter to the WAN address of the London multiprotocol router. When it gets there, the London router removes the IP packet and sends it to host 2 inside an Ethernet frame.

The WAN can be seen as a big tunnel extending from one multiprotocol router to the other. The IP packet just travels from one end of the tunnel to the other, snug in its nice box. It does not have to worry about dealing with the WAN at all. Neither do the hosts on either Ethernet. Only the multiprotocol router has to understand IP and WAN packets. In effect, the entire distance from the middle of one multiprotocol router to the middle of the other acts like a serial line.

An analogy may make tunneling clearer. Consider a person driving her car from Paris to London. Within France, the car moves under its own power, but when it hits the English Channel, it is loaded into a high-speed train and transported to England through the Chunnel (cars are not permitted to drive through the Chunnel). Effectively, the car is being carried as freight, as depicted in Fig. 5-39.

At the far end, the car is let loose on the English roads and once again continues to move under its own power. Tunneling of packets through a foreign network works the same way.

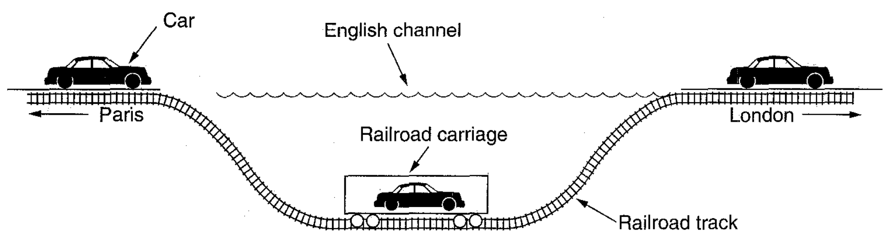

Fig. 5-39. Tunneling a car from France to England.

5.4.5. Internetwork Routing

Routing through an internetwork is similar to routing within a single subnet, but with some added complications. Consider, for example, the internetwork of Fig. 5-40(a) in which five networks are connected by six multiprotocol routers. Making a graph model of this situation is complicated by the fact that every multiprotocol router can directly access (i.e., send packets to) every other router connected to any network to which it is connected. For example, *B* in Fig. 5-40(a) can directly access *A* and *C* via network 2 and also *D* via network 3. This leads to the graph of Fig. 5-40(b).

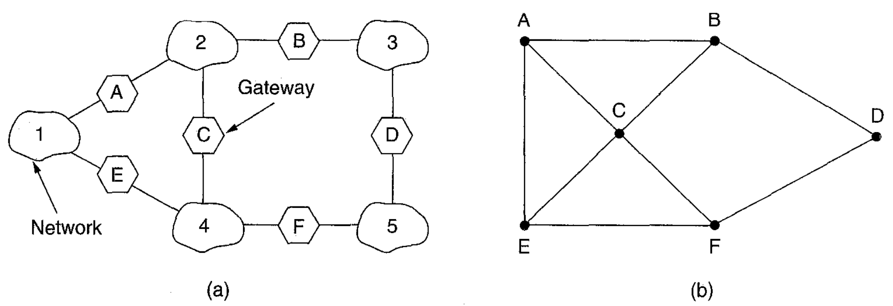

Fig. 5-40. (a) An internetwork. (b) A graph of the internetwork.

Once the graph has been constructed, known routing algorithms, such as the distance vector and link state algorithms, can be applied to the set of multiprotocol routers. This gives a two-level routing algorithm: within each network an **interior gateway protocol** is used, but between the networks, an **exterior gateway protocol** is used ("gateway" is an older term for "router"). In fact, since each network is independent, they may all use different algorithms. Because each network in an internetwork is independent of all the others, it is often referred to as an **Autonomous System** (**AS**).

A typical internet packet starts out on its LAN addressed to the local multiprotocol router (in the MAC layer header). After it gets there, the network layer code decides which multiprotocol router to forward the packet to, using its own routing tables. If that router can be reached using the packet's native network protocol, it is forwarded there directly. Otherwise it is tunneled there, encapsulated in the protocol required by the intervening network. This process is repeated until the packet reaches the destination network.

One of the differences between internetwork routing and intranetwork routing is that internetwork routing often requires crossing international boundaries. Various laws suddenly come into play, such as Sweden's strict privacy laws about exporting personal data about Swedish citizens from Sweden. Another example is the Canadian law saying that data traffic originating in Canada and ending in Canada may not leave the country. This law means that traffic from Windsor, Ontario to Vancouver may not be routed via nearby Detroit.

Another difference between interior and exterior routing is the cost. Within a single network, a single charging algorithm normally applies. However, different networks may be under different managements, and one route may be less expensive than another. Similarly, the quality of service offered by different networks may be different, and this may be a reason to choose one route over another.

In a large internetwork, choosing the best route may be a time-consuming operation. Estrin et al. (1992) have proposed dealing with this problem by precomputing routes for popular (source, destination) pairs and storing them in a database to be consulted at route selection time.

5.4.6. Fragmentation

Each network imposes some maximum size on its packets. These limits have various causes, among them:

1. Hardware (e.g., the width of a TDM transmission slot).
2. Operating system (e.g., all buffers are 512 bytes).
3. Protocols (e.g., the number of bits in the packet length field).
4. Compliance with some (inter)national standard.
5. Desire to reduce error induced retransmissions to some level.
6. Desire to prevent one packet from occupying the channel too long.

The result of all these factors is that the network designers are not free to choose any maximum packet size they wish. Maximum payloads range from 48 bytes

(ATM cells) to 65,515 bytes (IP packets), although the payload size in higher layers is often larger.

An obvious problem appears when a large packet wants to travel through a network whose maximum packet size is too small. One solution is to make sure the problem does not occur in the first place. In other words, the internet should use a routing algorithm that avoids sending packets through networks that cannot handle them. However, this solution is no solution at all. What happens if the original source packet is too large to be handled by the destination network? The routing algorithm can hardly bypass the destination.

Basically, the only solution to the problem is to allow gateways to break packets up into **fragments**, sending each fragment as a separate internet packet. However, as every parent of a small child knows, converting a large object into small fragments is considerably easier than the reverse process. (Physicists have even given this effect a name: the second law of thermodynamics.) Packet-switching networks, too, have trouble putting the fragments back together again.

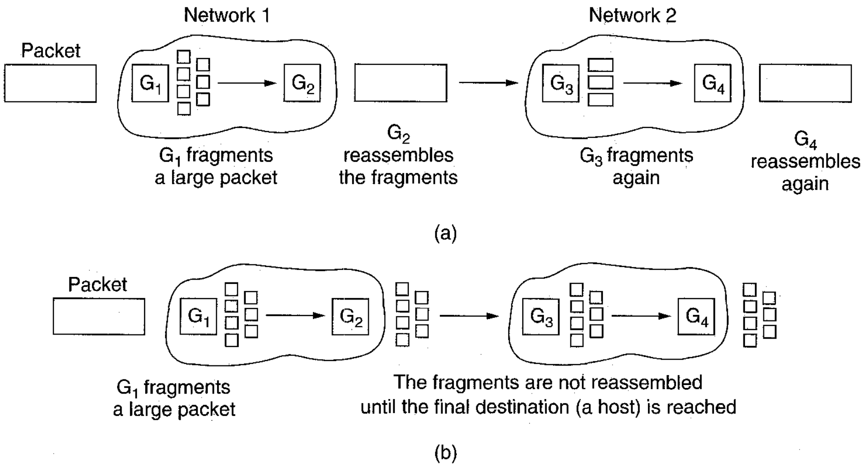

Fig. 5-41. (a) Transparent fragmentation. (b) Nontransparent fragmentation.

Two opposing strategies exist for recombining the fragments back into the original packet. The first strategy is to make fragmentation caused by a "small-packet" network transparent to any subsequent networks through which the packet must pass on its way to the ultimate destination. This option is shown in Fig. 5-41(a). When an oversized packet arrives at a gateway, the gateway breaks it up into fragments. Each fragment is addressed to the same exit gateway, where the pieces are recombined. In this way passage through the small-packet network has been made transparent. Subsequent networks are not even aware that fragmentation has occurred. ATM networks, for example, have special hardware to provide transparent fragmentation of packets into cells and then reassembly of cells into packets. In the ATM world, fragmentation is called segmentation; the concept is the same, but some of the details are different.

Transparent fragmentation is simple but has some problems. For one thing, the exit gateway must know when it has received all the pieces, so that either a count field or an "end of packet" bit must be included in each packet. For another thing, all packets must exit via the same gateway. By not allowing some fragments to follow one route to the ultimate destination, and other fragments a disjoint route, some performance may be lost. A last problem is the overhead required to repeatedly reassemble and then refragment a large packet passing through a series of small-packet networks.

The other fragmentation strategy is to refrain from recombining fragments at any intermediate gateways. Once a packet has been fragmented, each fragment is treated as though it were an original packet. All fragments are passed through the exit gateway (or gateways), as shown in Fig. 5-41(b). Recombination occurs only at the destination host.

Nontransparent fragmentation also has some problems. For example, it requires *every* host to be able to do reassembly. Yet another problem is that when a large packet is fragmented the total overhead increases, because each fragment must have a header. Whereas in the first method this overhead disappears as soon as the small-packet network is exited, in this method the overhead remains for the rest of the journey. An advantage of this method, however, is that multiple exit gateways can now be used and higher performance can be achieved. Of course, if the concatenated virtual circuit model is being used, this advantage is of no use.

When a packet is fragmented, the fragments must be numbered in such a way that the original data stream can be reconstructed. One way of numbering the fragments is to use a tree. If packet 0 must be split up, the pieces are called 0.0, 0.1, 0.2, etc. If these fragments themselves must be fragmented later on, the pieces are numbered 0.0.0, 0.0.1, 0.0.2, . . . , 0.1.0, 0.1.1, 0.1.2, etc. If enough fields have been reserved in the header for the worst case and no duplicates are generated anywhere, this scheme is sufficient to ensure that all the pieces can be correctly reassembled at the destination, no matter what order they arrive in.

However, if even one network loses or discards packets, there is a need for end-to-end retransmissions, with unfortunate effects for the numbering system. Suppose that a 1024-bit packet is initially fragmented into four equal-sized fragments, 0.0, 0.1, 0.2, and 0.3. Fragment 0.1 is lost, but the other parts arrive at the destination. Eventually, the source times out and retransmits the original packet again. Only this time the route taken passes through a network with a 512-bit limit, so two fragments are generated. When the new fragment 0.1 arrives at the destination, the receiver will think that all four pieces are now accounted for and reconstruct the packet incorrectly.

A completely different (and better) numbering system is for the internetwork protocol to define an elementary fragment size small enough that the elementary fragment can pass through every network. When a packet is fragmented, all the pieces are equal to the elementary fragment size except the last one, which may be shorter. An internet packet may contain several fragments, for efficiency reasons. The internet header must provide the original packet number, and the number of the (first) elementary fragment contained in the packet. As usual, there must also be a bit indicating that the last elementary fragment contained within the internet packet is the last one of the original packet.

This approach requires two sequence fields in the internet header: the original packet number, and the fragment number. There is clearly a trade-off between the size of the elementary fragment and the number of bits in the fragment number. Because the elementary fragment size is presumed to be acceptable to every network, subsequent fragmentation of an internet packet containing several fragments causes no problem. The ultimate limit here is to have the elementary fragment be a single bit or byte, with the fragment number then being the bit or byte offset within the original packet, as shown in Fig. 5-42.

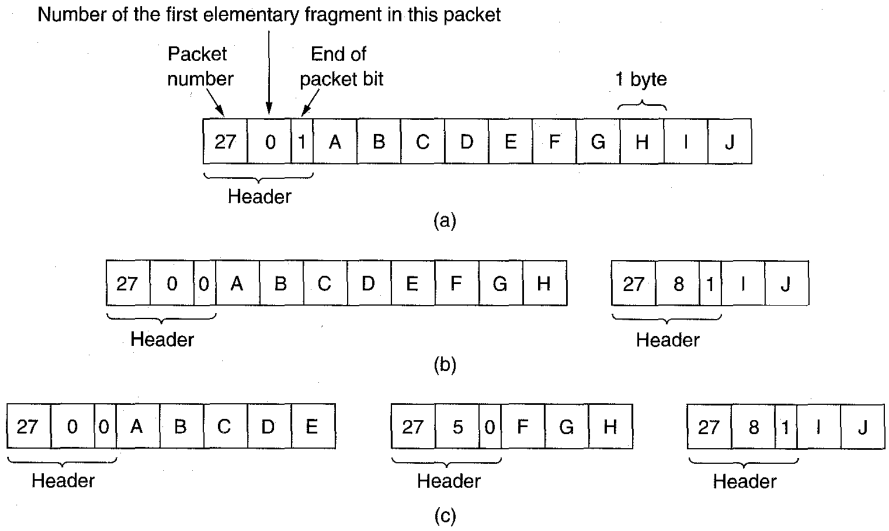

Fig. 5-42. Fragmentation when the elementary data size is 1 byte. (a) Original packet, containing 10 data bytes. (b) Fragments after passing through a network with maximum packet size of 8 bytes. (c) Fragments after passing through a size 5 gateway.

Some internet protocols take this method even further and consider the entire transmission on a virtual circuit to be one giant packet, so that each fragment contains the absolute byte number of the first byte within the fragment. Some other issues relating to fragmentation are discussed in (Kent and Mogul, 1987).

5.4.7. Firewalls

The ability to connect any computer, anywhere, to any other computer, anywhere, is a mixed blessing. For individuals at home, wandering around the Internet is lots of fun. For corporate security managers, it is a nightmare. Most companies have large amounts of confidential information on-line—trade secrets, product development plans, marketing strategies, financial analyses, etc. Disclosure of this information to a competitor could have dire consequences.

In addition to the danger of information leaking out, there is also a danger of information leaking in. In particular, viruses, worms, and other digital pests (Kaufman et al., 1995) can breach security, destroy valuable data, and waste large amounts of administrators' time trying to clean up the mess they leave. Often they are imported by careless employees who want to play some nifty new game.

Consequently, mechanisms are needed to keep "good" bits in and "bad" bits out. One method is to use encryption. This approach protects data in transit between secure sites. We will study it in Chap. 7. However, encryption does nothing to keep digital pests and hackers out. To accomplish this goal, we need to look at firewalls (Chapman and Zwicky, 1995; and Cheswick and Bellovin, 1994).

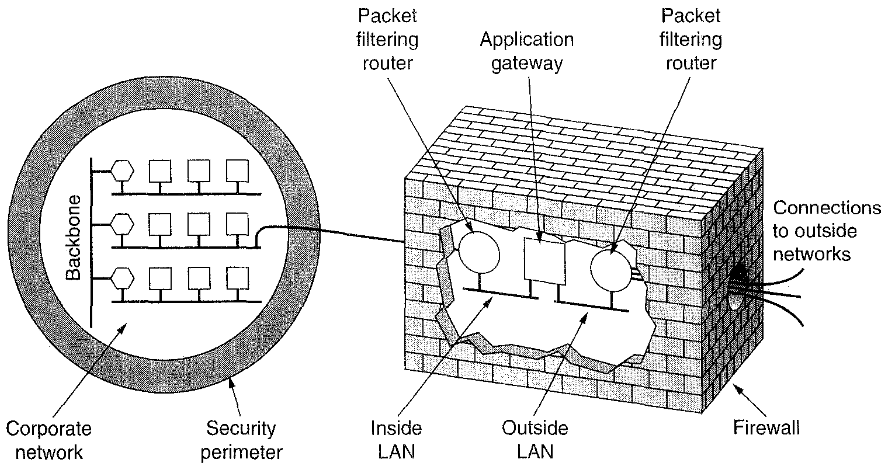

Fig. 5-43. A firewall consisting of two packet filters and an application gateway.

Firewalls are just a modern adaptation of that old medieval security standby: digging a deep moat around your castle. This design forced everyone entering or leaving the castle to pass over a single drawbridge, where they could be inspected by the I/O police. With networks, the same trick is possible: a company can have many LANs connected in arbitrary ways, but all traffic to or from the company is forced through an electronic drawbridge (firewall), as shown in Fig. 5-43.

The firewall in this configuration has two components: two routers that do packet filtering and an application gateway. Simpler configurations also exist, but the advantage of this design is that every packet must transit two filters and an application gateway to go in or out. No other route exists. Readers who think that one security checkpoint is enough clearly have not made an international flight on a scheduled airline recently.

Each **packet filter** is a standard router equipped with some extra functionality. The extra functionality allows every incoming or outgoing packet to be inspected. Packets meeting some criterion are forwarded normally. Those that fail the test are dropped.

In Fig. 5-43, most likely the packet filter on the inside LAN checks outgoing packets and the one on the outside LAN checks incoming packets. Packets crossing the first hurdle go to the application gateway for further examination. The point of putting the two packet filters on different LANs is to ensure that no packet gets in or out without having to pass through the application gateway: there is no path around it.

Packet filters are typically driven by tables configured by the system administrator. These tables list sources and destinations that are acceptable, sources and destinations that are blocked, and default rules about what to do with packets coming from or going to other machines.

In the common case of a UNIX setting, a source or destination consists of an IP address and a port. Ports indicate which service is desired. For example, port 23 is for Telnet, port 79 is for Finger, and port 119 is for USENET news. A company could block incoming packets for all IP addresses combined with one of these ports. In this way, no one outside the company could log in via Telnet, or look up people using the Finger daemon. Furthermore, the company would be spared from having employees spend all day reading USENET news.

Blocking outgoing packets is trickier because although most sites stick to the standard port naming conventions, they are not forced to do so. Furthermore, for some important services, such as FTP (File Transfer Protocol), port numbers are assigned dynamically. In addition, although blocking TCP connections is difficult, blocking UDP packets is even harder because so little is known a priori about what they will do. Many packet filters simply ban UDP traffic altogether.

The second half of the firewall mechanism is the **application gateway**. Rather than just looking at raw packets, the gateway operates at the application level. A mail gateway, for example, can be set up to examine each message going in or coming out. For each one it makes a decision to transmit or discard it based on header fields, message size, or even the content (e.g., at a military installation, the presence of words like "nuclear" or "bomb" might cause some special action to be taken).

Installations are free to set up one or more application gateways for specific applications, but it is not uncommon for suspicious organizations to permit email in and out, and perhaps use of the World Wide Web, but ban everything else as too dicey. Combined with encryption and packet filtering, this arrangement offers a limited amount of security at the cost of some inconvenience.

One final note concerns wireless communication and firewalls. It is easy to design a system that is logically completely secure, but which, in practice, leaks like a sieve. This situation can occur if some of the machines are wireless and use radio communication, which passes right over the firewall in both directions.

5.5. THE NETWORK LAYER IN THE INTERNET

At the network layer, the Internet can be viewed as a collection of subnetworks or **Autonomous Systems** (**ASes**) that are connected together. There is no real structure, but several major backbones exist. These are constructed from high-bandwidth lines and fast routers. Attached to the backbones are regional (midlevel) networks, and attached to these regional networks are the LANs at many universities, companies, and Internet service providers. A sketch of this quasihierarchical organization is given in Fig. 5-44.

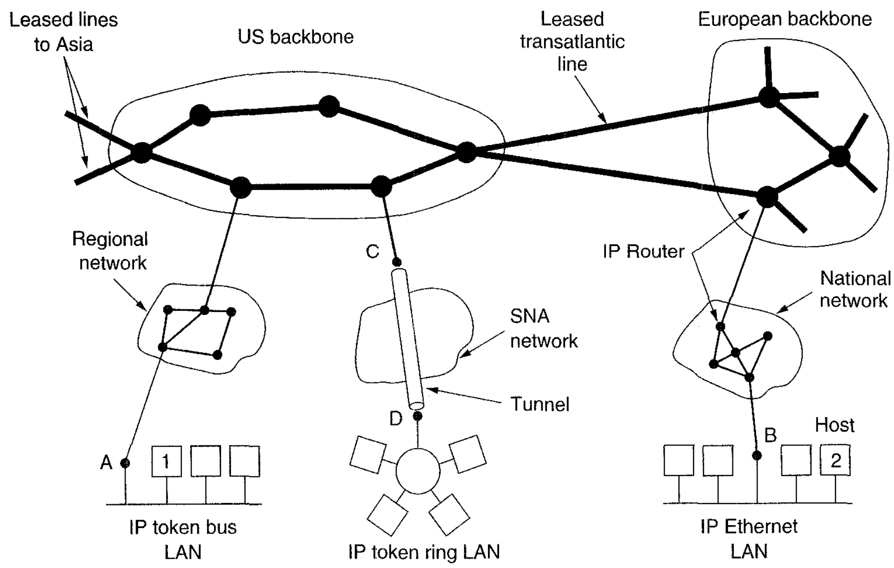

Fig. 5-44. The Internet is an interconnected collection of many networks.

The glue that holds the Internet together is the network layer protocol, **IP** (**Internet Protocol**). Unlike most older network layer protocols, it was designed from the beginning with internetworking in mind. A good way to think of the network layer is this. Its job is to provide a best-efforts way to transport datagrams from source to destination, without regard to whether or not these machines are on the same network, or whether or not there are other networks in between them.

Communication in the Internet works as follows. The transport layer takes data streams and breaks them up into datagrams. In theory, datagrams can be up to 64 Kbytes each, but in practice they are usually around 1500 bytes. Each datagram is transmitted through the Internet, possibly being fragmented into smaller units as it goes. When all the pieces finally get to the destination machine, they are reassembled by the network layer into the original datagram. This datagram is then handed to the transport layer, which inserts it into the receiving process' input stream.

5.5.1. The IP Protocol

An appropriate place to start our study of the network layer in the Internet is the format of the IP datagrams themselves. An IP datagram consists of a header part and a text part. The header has a 20-byte fixed part and a variable length optional part. The header format is shown in Fig. 5-45. It is transmitted in big endian order: from left to right, with the high-order bit of the *Version* field going first. (The SPARC is big endian; the Pentium is little endian.) On little endian machines, software conversion is required on both transmission and reception.

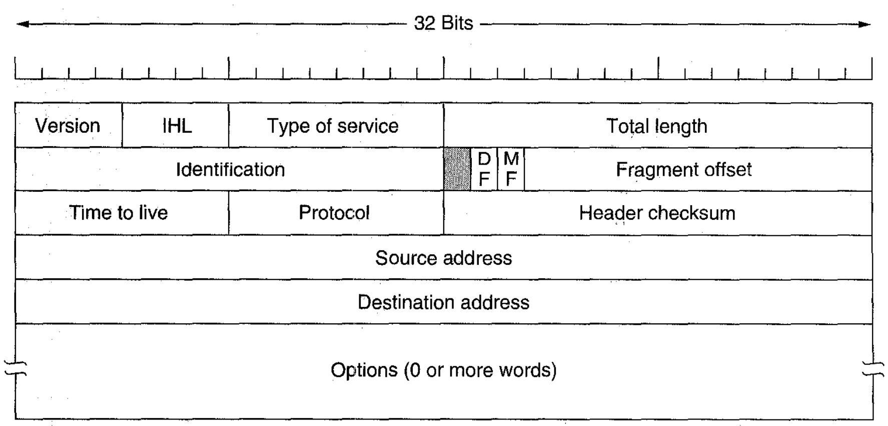

Fig. 5-45. The IP (Internet Protocol) header.

The *Version* field keeps track of which version of the protocol the datagram belongs to. By including the version in each datagram, it becomes possible to have the transition between versions take months, or even years, with some machines running the old version and others running the new one.

Since the header length is not constant, a field in the header, *IHL*, is provided to tell how long the header is, in 32-bit words. The minimum value is 5, which

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applies when no options are present. The maximum value of this 4-bit field is 15, which limits the header to 60 bytes, and thus the options field to 40 bytes. For some options, such as one that records the route a packet has taken, 40 bytes is far too small, making the option useless.

The *Type of service* field allows the host to tell the subnet what kind of service it wants. Various combinations of reliability and speed are possible. For digitized voice, fast delivery beats accurate delivery. For file transfer, error-free transmission is more important than fast transmission.

The field itself contains (from left to right), a three-bit *Precedence* field, three flags, *D*, *T*, and *R*, and 2 unused bits. The *Precedence* field is a priority, from 0 (normal) to 7 (network control packet). The three flag bits allow the host to specify what it cares most about from the set {Delay, Throughput, Reliability}. In theory, these fields allow routers to make choices between, for example, a satellite link with high throughput and high delay or a leased line with low throughput and low delay. In practice, current routers ignore the *Type of Service* field altogether.

The *Total length* includes everything in the datagram—both header and data. The maximum length is 65,535 bytes. At present, this upper limit is tolerable, but with future gigabit networks larger datagrams will be needed.

The *Identification* field is needed to allow the destination host to determine which datagram a newly arrived fragment belongs to. All the fragments of a datagram contain the same *Identification* value.

Next comes an unused bit and then two 1-bit fields. *DF* stands for Don't Fragment. It is an order to the routers not to fragment the datagram because the destination is incapable of putting the pieces back together again. For example, when a computer boots, its ROM might ask for a memory image to be sent to it as a single datagram. By marking the datagram with the *DF* bit, the sender knows it will arrive in one piece, even if this means that the datagram must avoid a small-packet network on the best path and take a suboptimal route. All machines are required to accept fragments of 576 bytes or less.

MF stands for More Fragments. All fragments except the last one have this bit set. It is needed to know when all fragments of a datagram have arrived.

The *Fragment offset* tells where in the current datagram this fragment belongs. All fragments except the last one in a datagram must be a multiple of 8 bytes, the elementary fragment unit. Since 13 bits are provided, there is a maximum of 8192 fragments per datagram, giving a maximum datagram length of 65,536 bytes, one more than the *Total length* field.

The *Time to live* field is a counter used to limit packet lifetimes. It is supposed to count time in seconds, allowing a maximum lifetime of 255 sec. It must be decremented on each hop and is supposed to be decremented multiple times when queued for a long time in a router. In practice, it just counts hops. When it hits zero, the packet is discarded and a warning packet is sent back to the source host. This feature prevents datagrams for wandering around forever, something that otherwise might happen if the routing tables ever become corrupted.

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When the network layer has assembled a complete datagram, it needs to know what to do with it. The *Protocol* field tells it which transport process to give it to. TCP is one possibility, but so are UDP and some others. The numbering of protocols is global across the entire Internet and is defined in RFC 1700.

The *Header checksum* verifies the header only. Such a checksum is useful for detecting errors generated by bad memory words inside a router. The algorithm is to add up all the 16-bit halfwords as they arrive, using one's complement arithmetic and then take the one's complement of the result. For purposes of this algorithm, the *Header checksum* is assumed to be zero upon arrival. This algorithm is more robust than using a normal add. Note that the *Header checksum* must be recomputed at each hop, because at least one field always changes (the *Time to live* field), but tricks can be used to speed up the computation.

The *Source address* and *Destination address* indicate the network number and host number. We will discuss Internet addresses in the next section. The *Options* field was designed to provide an escape to allow subsequent versions of the protocol to include information not present in the original design, to permit experimenters to try out new ideas, and to avoid allocating header bits to information that is rarely needed. The options are variable length. Each begins with a 1-byte code identifying the option. Some options are followed by a 1-byte option length field, and then one or more data bytes. The *Options* field is padded out to a multiple of four bytes. Currently five options are defined, as listed in Fig. 5-46, but not all routers support all of them.

Option	Description
Security	Specifies how secret the datagram is
Strict source routing	Gives the complete path to be followed
Loose source routing	Gives a list of routers not to be missed
Record route	Makes each router append its IP address
Timestamp	Makes each router append its address and timestamp

Fig. 5-46. IP options.

The *Security* option tells how secret the information is. In theory, a military router might use this field to specify not to route through certain countries the military considers to be "bad guys." In practice, all routers ignore it, so its only practical function is to help spies find the good stuff more easily.

The *Strict source routing* option gives the complete path from source to destination as a sequence of IP addresses. The datagram is required to follow that exact route. It is most useful for system managers to send emergency packets when the routing tables are corrupted, or for making timing measurements.

The *Loose source routing* option requires the packet to traverse the list of routers specified, and in the order specified, but it is allowed to pass through other

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routers on the way. Normally, this option would only provide a few routers, to force a particular path. For example, to force a packet from London to Sydney to go west instead of east, this option might specify routers in New York, Los Angeles, and Honolulu. This option is most useful when political or economic considerations dictate passing through or avoiding certain countries.

The *Record route* option tells the routers along the path to append their IP address to the option field. This allows system managers to track down bugs in the routing algorithms ("Why are packets from Houston to Dallas all visiting Tokyo first?") When the ARPANET was first set up, no packet ever passed through more than nine routers, so 40 bytes of option was ample. As mentioned above, now it is too small.

Finally, the *Timestamp* option is like the *Record route* option, except that in addition to recording its 32-bit IP address, each router also records a 32-bit time-stamp. This option, too, is mostly for debugging routing algorithms.

5.5.2. IP Addresses

Every host and router on the Internet has an IP address, which encodes its network number and host number. The combination is unique: no two machines have the same IP address. All IP addresses are 32 bits long and are used in the *Source address* and *Destination address* fields of IP packets. The formats used for IP address are shown in Fig. 5-47. Those machines connected to multiple networks have a different IP address on each network.

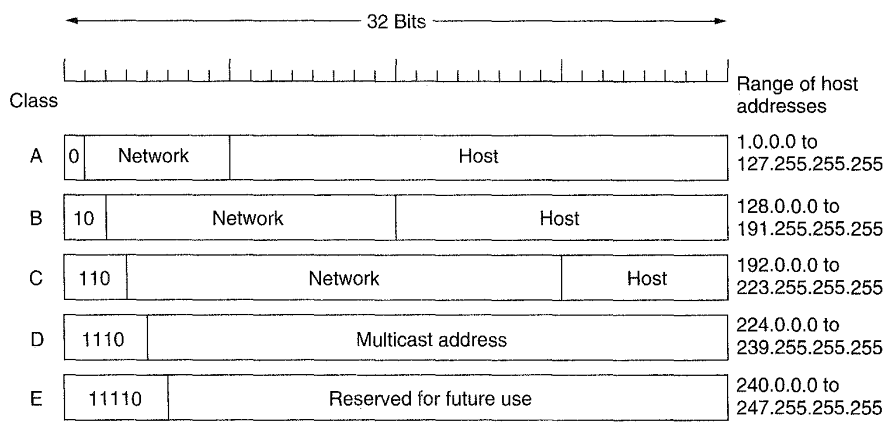

Fig. 5-47. IP address formats.

The class A, B, C, and D formats allow for up to 126 networks with 16 million hosts each, 16,382 networks with up to 64K hosts, 2 million networks, (e.g.,

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LANs), with up to 254 hosts each, and multicast, in which a datagram is directed to multiple hosts. Addresses beginning with 11110 are reserved for future use. Tens of thousands of networks are now connected to the Internet, and the number doubles every year. Network numbers are assigned by the **NIC** (**Network Information Center**) to avoid conflicts.

Network addresses, which are 32-bit numbers, are usually written in **dotted decimal notation**. In this format, each of the 4 bytes is written in decimal, from 0 to 255. For example, the hexadecimal address C0290614 is written as 192.41.6.20. The lowest IP address is 0.0.0.0 and the highest is 255.255.255.255.

The values 0 and −1 have special meanings, as shown in Fig. 5-48. The value 0 means this network or this host. The value of −1 is used as a broadcast address to mean all hosts on the indicated network.

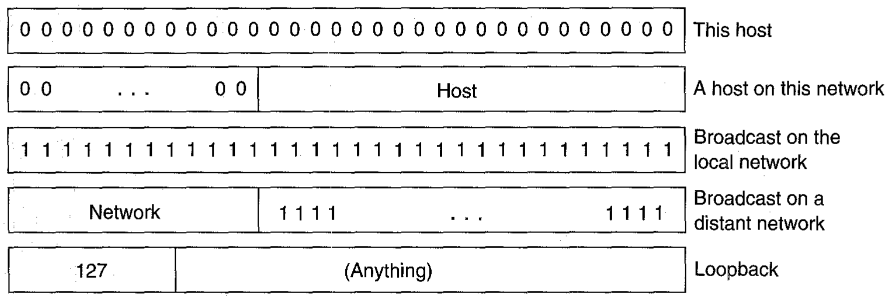


Fig. 5-48. Special IP addresses.

The IP address 0.0.0.0 is used by hosts when they are being booted but is not used afterward. IP addresses with 0 as network number refer to the current network. These addresses allow machines to refer to their own network without knowing its number (but they have to know its class to know how many 0s to include). The address consisting of all 1s allows broadcasting on the local network, typically a LAN. The addresses with a proper network number and all 1s in the host field allow machines to send broadcast packets to distant LANs anywhere in the Internet. Finally, all addresses of the form 127.xx.yy.zz are reserved for loopback testing. Packets sent to that address are not put out onto the wire; they are processed locally and treated as incoming packets. This allows packets to be sent to the local network without the sender knowing its number. This feature is also used for debugging network software.

5.5.3. Subnets

As we have seen, all the hosts in a network must have the same network number. This property of IP addressing can cause problems as networks grow. For example, consider a company that starts out with one class C LAN on the

Internet. As time goes on, it might acquire more than 254 machines, and thus need a second class C address. Alternatively, it might acquire a second LAN of a different type and want a separate IP address for it (the LANs could be bridged to form a single IP network, but bridges have their own problems). Eventually, it might end up with many LANs, each with its own router and each with its own class C network number.

As the number of distinct local networks grows, managing them can become a serious headache. Every time a new network is installed the system administrator has to contact NIC to get a new network number. Then this number must be announced worldwide. Furthermore, moving a machine from one LAN to another requires it to change its IP address, which in turn may mean modifying its configuration files and also announcing the new IP address to the world. If some other machine is given the newly-released IP address, that machine will get email and other data intended for the original machine until the address has propagated all over the world.

The solution to these problems is to allow a network to be split into several parts for internal use but still act like a single network to the outside world. In the Internet literature, these parts are called **subnets**. As we mentioned in Chap. 1, this usage conflicts with "subnet" to mean the set of all routers and communication lines in a network. Hopefully it will be clear from the context which meaning is intended. In this section, the new definition will be the one used. If our growing company started up with a class B address instead of a class C address, it could start out just numbering the hosts from 1 to 254. When the second LAN arrived, it could decide, for example, to split the 16-bit host number into a 6-bit subnet number and a 10-bit host number, as shown in Fig. 5-49. This split allows 62 LANs (0 and −1 are reserved), each with up to 1022 hosts.

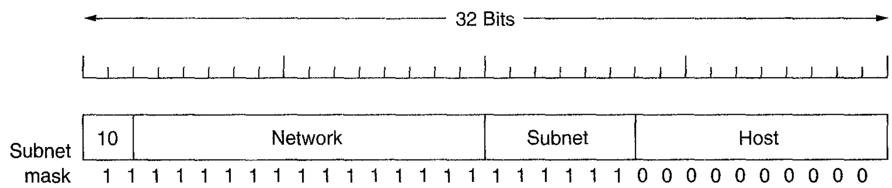


Fig. 5-49. One of the ways to subnet a class B network.

Outside the network, the subnetting is not visible, so allocating a new subnet does not require contacting NIC or changing any external databases. In this example, the first subnet might use IP addresses starting at 130.50.4.1, the second subnet might start at 130.50.8.1, and so on.

To see how subnets work, it is necessary to explain how IP packets are processed at a router. Each router has a table listing some number of (network, 0) IP addresses and some number of (this-network, host) IP addresses. The first kind tells how to get to distant networks. The second kind tells how to get to local hosts. Associated with each table is the network interface to use to reach the destination, and certain other information.

When an IP packet arrives, its destination address is looked up in the routing table. If the packet is for a distant network, it is forwarded to the next router on the interface given in the table. If it is a local host (e.g., on the router's LAN), it is sent directly to the destination. If the network is not present, the packet is forwarded to a default router with more extensive tables. This algorithm means that each router only has to keep track of other networks and local hosts, not (network, host) pairs, greatly reducing the size of the routing table.

When subnetting is introduced, the routing tables are changed, adding entries of the form (this-network, subnet, 0) and (this-network, this-subnet, host). Thus a router on subnet k knows how to get to all the other subnets and also how to get to all the hosts on subnet k. It does not have to know the details about hosts on other subnets. In fact, all that needs to be changed is to have each router do a Boolean AND with the network's **subnet mask** (see Fig. 5-49) to get rid of the host number and look up the resulting address in its tables (after determining which network class it is). For example, a packet addressed to 130.50.15.6 and arriving at a router on subnet 5 is ANDed with the subnet mask of Fig. 5-49 to give the address 130.50.12.0. This address is looked up in the routing tables to find out how to get to hosts on subnet 3. The router on subnet 5 is thus spared the work of keeping track of the data link addresses of hosts other than those on subnet 5. Subnetting thus reduces router table space by creating a three-level hierarchy.

5.5.4. Internet Control Protocols

In addition to IP, which is used for data transfer, the Internet has several control protocols used in the network layer, including ICMP, ARP, RARP, and BOOTP. In this section we will look at each of these in turn.

The Internet Control Message Protocol

The operation of the Internet is monitored closely by the routers. When something unexpected occurs, the event is reported by the **ICMP** (**Internet Control Message Protocol**), which is also used to test the Internet. About a dozen types of ICMP messages are defined. The most important ones are listed in Fig. 5-50. Each ICMP message type is encapsulated in an IP packet.

The DESTINATION UNREACHABLE message is used when the subnet or a router cannot locate the destination, or a packet with the *DF* bit cannot be delivered because a "small-packet" network stands in the way.

The TIME EXCEEDED message is sent when a packet is dropped due to its counter reaching zero. This event is a symptom that packets are looping, that there is enormous congestion, or that the timer values are being set too low.

Message type	Description
Destination unreachable	Packet could not be delivered
Time exceeded	Time to live field hit 0
Parameter problem	Invalid header field
Source quench	Choke packet
Redirect	Teach a router about geography
Echo request	Ask a machine if it is alive
Echo reply	Yes, I am alive
Timestamp request	Same as Echo request, but with timestamp
Timestamp reply	Same as Echo reply, but with timestamp

Fig. 5-50. The principal ICMP message types.

The PARAMETER PROBLEM message indicates that an illegal value has been detected in a header field. This problem indicates a bug in the sending host's IP software, or possibly in the software of a router transited.

The SOURCE QUENCH message was formerly used to throttle hosts that were sending too many packets. When a host received this message, it was expected to slow down. It is rarely used any more because when congestion occurs, these packets tend to add more fuel to the fire. Congestion control in the Internet is now done largely in the transport layer and will be studied in detail in Chap. 6.

The REDIRECT message is used when a router notices that a packet seems to be routed wrong. It is used by the router to tell the sending host about the probable error.

The ECHO REQUEST and ECHO REPLY messages are used to see if a given destination is reachable and alive. Upon receiving the ECHO message, the destination is expected to send an ECHO REPLY message back. The TIMESTAMP REQUEST and TIMESTAMP REPLY messages are similar, except that the arrival time of the message and the departure time of the reply are recorded in the reply. This facility is used to measure network performance.

In addition to these messages, there are four others that deal with Internet addressing, to allow hosts to discover their network numbers and to handle the case of multiple LANs sharing a single IP address. ICMP is defined in RFC 792.

The Address Resolution Protocol

Although every machine on the Internet has one (or more) IP addresses, these cannot actually be used for sending packets because the data link layer hardware does not understand Internet addresses. Nowadays, most hosts are attached to a

LAN by an interface board that only understands LAN addresses. For example, every Ethernet board ever manufactured comes equipped with a 48-bit Ethernet address. Manufacturers of Ethernet boards request a block of addresses from a central authority to ensure that no two boards have the same address (to avoid conflicts should the two boards ever appear on the same LAN). The boards send and receive frames based on 48-bit Ethernet addresses. They know nothing at all about 32-bit IP addresses.

The question now arises: How do IP addresses get mapped onto data link layer addresses, such as Ethernet? To explain how this works, let us use the example of Fig. 5-51, in which a small university with several class C networks is illustrated. Here we have two Ethernets, one in the Computer Science department, with IP address 192.31.65.0 and one in Electrical Engineering, with IP address 192.31.63.0. These are connected by a campus FDDI ring with IP address 192.31.60.0. Each machine on an Ethernet has a unique Ethernet address, labeled *E1* through *E6*, and each machine on the FDDI ring has an FDDI address, labeled *F1* through *F3*.

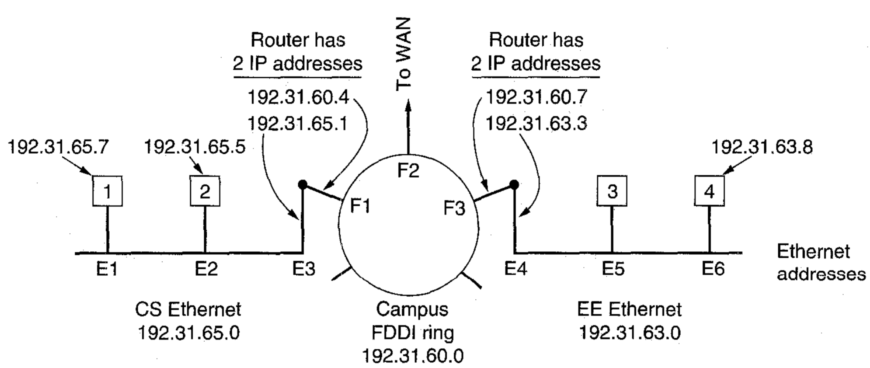


Fig. 5-51. Three interconnected class C networks: two Ethernets and an FDDI ring.

Let us start out by seeing how a user on host 1 sends a packet to a user on host 2. Let us assume the sender knows the name of the intended receiver, possibly something like *mary@eagle.cs.uni.edu*. The first step is to find the IP address for host 2, known as *eagle.cs.uni.edu*. This lookup is performed by the Domain Name System, which we will study in Chap. 7. For the moment, we will just assume that DNS returns the IP address for host 2 (192.31.65.5).

The upper layer software on host 1 now builds a packet with 192.31.65.5 in the *Destination address* field and gives it to the IP software to transmit. The IP software can look at the address and see that the destination is on its own network, but it needs a way to find the destination's Ethernet address. One solution is to have a configuration file somewhere in the system that maps IP addresses onto

Ethernet addresses. This solution is certainly possible, but for organizations with thousands of machines, keeping these files up to date is an error-prone, time-consuming job.

A better solution is for host 1 to output a broadcast packet onto the Ethernet asking: "Who owns IP address 192.31.65.5?" The broadcast will arrive at every machine on Ethernet 192.31.65.0, and each one will check its IP address. Host 2 alone will respond with its Ethernet address (*E2*). In this way host 1 learns that IP address 192.31.65.5 is on the host with Ethernet address *E2*. The protocol for asking this question and getting the reply is called **ARP** (**Address Resolution Protocol**). Almost every machine on the Internet runs it. It is defined in RFC 826.

The advantage of using ARP over configuration files is the simplicity. The system manager does not have to do much except assign each machine an IP address and decide about subnet masks. ARP does the rest.

At this point, the IP software on host 1 builds an Ethernet frame addressed to *E2*, puts the IP packet (addressed to 192.31.65.5) in the payload field, and dumps it onto the Ethernet. The Ethernet board of host 2 detects this frame, recognizes it as a frame for itself, scoops it up, and causes an interrupt. The Ethernet driver extracts the IP packet from the payload and passes it to the IP software, which sees that it is correctly addressed, and processes it.

Various optimizations are possible to make ARP more efficient. To start with, once a machine has run ARP, it caches the result in case it needs to contact the same machine shortly. Next time it will find the mapping in its own cache, thus eliminating the need for a second broadcast. In many cases host 2 will need to send back a reply, forcing it, too, to run ARP to determine the sender's Ethernet address. This ARP broadcast can be avoided by having host 1 include its IP to Ethernet mapping in the ARP packet. When ARP broadcast arrives at host 2, the pair (192.31.65.7, E1) is entered into host 2's ARP cache for future use. In fact, all machines on the Ethernet can enter this mapping into their ARP caches.

Yet another optimization is to have every machine broadcast its mapping when it boots. This broadcast is generally done in the form of an ARP looking for its own IP address. There should not be a response, but a side effect of the broadcast is to make any entry in everyone's ARP cache. If a response does arrive, two machines have been assigned the same IP address. The new one should inform the system manager and not boot.

To allow mappings to change, for example, when an Ethernet board breaks and is replaced with a new one (and thus a new Ethernet address), entries in the ARP cache should time out after a few minutes.

Now let us look at Fig. 5-51 again, only this time host 1 wants to send a packet to host 6 (192.31.63.8). Using ARP will fail because host 4 will not see the broadcast (routers do not forward Ethernet-level broadcasts). There are two solutions. First, the CS router could be configured to respond to ARP requests for network 192.31.63.0 (and possibly other local networks). In this case, host 1 will make an ARP cache entry of (192.31.63.8, E3) and happily send all traffic for host

4 to the local router. This solution is called **proxy ARP**. The second solution is to have host 1 immediately see that the destination is on a remote network and just send all such traffic to a default Ethernet address that handles all remote traffic, in this case *E3*. This solution does not require having the CS router know which remote networks it is serving.

Either way, what happens is that host 1 packs the IP packet into the payload field of an Ethernet frame addressed to *E3*. When the CS router gets the Ethernet frame, it removes the IP packet from the payload field and looks up the IP address in its routing tables. It discovers that packets for network 192.31.63.0 are supposed to go to router 192.31.60.7. If it does not already know the FDDI address of 192.31.60.7, it broadcasts an ARP packet onto the ring and learns that its ring address is *F3*. It then inserts the packet into the payload field of an FDDI frame addressed to *F3* and puts it on the ring.

At the EE router, the FDDI driver removes the packet from the payload field and gives it to the IP software, which sees that it needs to send the packet to 192.31.63.8. If this IP address is not in its ARP cache, it broadcasts an ARP request on the EE Ethernet and learns that the destination address is *E6* so it builds an Ethernet frame addressed to *E6*, puts the packet in the payload field, and sends it over the Ethernet. When the Ethernet frame arrives at host 4, the packet is extracted from the frame and passed to the IP software for processing.

Going from host 1 to a distant network over a WAN works essentially the same way, except that this time the CS router's tables tell it to use the WAN router whose FDDI address is *F2*.

The Reverse Address Resolution Protocol

ARP solves the problem of finding out which Ethernet address corresponds to a given IP address. Sometimes the reverse problem has to solved: Given an Ethernet address, what is the corresponding IP address? In particular, this problem occurs when booting a diskless workstation. Such a machine will normally get the binary image of its operating system from a remote file server. But how does it learn its IP address?

The solution is to use the **RARP** (**Reverse Address Resolution Protocol**) (defined in RFC 903). This protocol allows a newly-booted workstation to broadcast its Ethernet address and say: "My 48-bit Ethernet address is 14.04.05.18.01.25. Does anyone out there know my IP address?" The RARP server sees this request, looks up the Ethernet address in its configuration files, and sends back the corresponding IP address.

Using RARP is better than embedding an IP address in the memory image because it allows the same image to be used on all machines. If the IP address were buried inside the image, each workstation would need its own image.

A disadvantage of RARP is that it uses a destination address of all 1s (limited broadcasting) to reach the RARP server. However, such broadcasts are not forwarded by routers, so a RARP server is needed on each network. To get around this problem, an alternative bootstrap protocol called **BOOTP** has been invented (see RFCs 951, 1048, and 1084). Unlike RARP, it uses UDP messages, which are forwarded over routers. It also provides a diskless workstation with additional information, including the IP address of the file server holding the memory image, the IP address of the default router, and the subnet mask to use. BOOTP is described in RFC 951.

5.5.5. The Interior Gateway Routing Protocol: OSPF

As we mentioned earlier, the Internet is made up of a large number of autonomous systems. Each AS is operated by a different organization and can use its own routing algorithm inside. For example, the internal networks of companies *X*, *Y*, and *Z* would usually be seen as three ASes if all three were on the Internet. All three may use different routing algorithms internally. Nevertheless, having standards, even for internal routing, simplifies the implementation at the boundaries between ASes and allows reuse of code. In this section we will study routing within an AS. In the next one, we will look at routing between ASes. A routing algorithm within an AS is called an **interior gateway protocol**; an algorithm for routing between ASes is called an **exterior gateway protocol**.

The original Internet interior gateway protocol was a distance vector protocol (RIP) based on the Bellman-Ford algorithm. It worked well in small systems, but less well as ASes got larger. It also suffered from the count-to-infinity problem and generally slow convergence, so it was replaced in May 1979 by a link state protocol. In 1988, the Internet Engineering Task Force began work on a successor. That successor, called **OSPF** (**Open Shortest Path First**) became a standard in 1990. Many router vendors are now supporting it, and it will become the main interior gateway protocol in the near future. Below we will give a sketch of how OSPF works. For the complete story, see RFC 1247.

Given the long experience with other routing protocols, the group designing the new protocol had a long list of requirements that had to be met. First, the algorithm had to be published in the open literature, hence the "O" in OSPF. A proprietary solution owned by one company would not do. Second, the new protocol had to support a variety of distance metrics, including physical distance, delay, and so on. Third, it had to be a dynamic algorithm, one that adapted to changes in the topology automatically and quickly.

Fourth, and new for OSPF, it had to support routing based on type of service. The new protocol had to be able to route real-time traffic one way and other traffic a different way. The IP protocol has a *Type of Service* field, but no existing routing protocol used it.

Fifth, and related to the above, the new protocol had to do load balancing, splitting the load over multiple lines. Most previous protocols sent all packets over the best route. The second-best route was not used at all. In many cases, splitting the load over multiple lines gives better performance.

Sixth, support for hierarchical systems was needed. By 1988, the Internet had grown so large that no router could be expected to know the entire topology. The new routing protocol had to be designed so that no router would have to.

Seventh, some modicum of security was required to prevent fun-loving students from spoofing routers by sending them false routing information. Finally, provision was needed for dealing with routers that were connected to the Internet via a tunnel. Previous protocols did not handle this well.

OSPF supports three kinds of connections and networks:

1. Point-to-point lines between exactly two routers.
2. Multiaccess networks with broadcasting (e.g., most LANs).
3. Multiaccess networks without broadcasting (e.g., most packet-switched WANs).

A **multiaccess** network is one that can have multiple routers on it, each of which can directly communicate with all the others. All LANs and WANs have this property. Figure 5-52(a) shows an AS containing all three kinds of networks. Note that hosts do not generally play a role in OSPF.

OSPF works by abstracting the collection of actual networks, routers, and lines into a directed graph in which each arc is assigned a cost (distance, delay, etc.). It then computes the shortest path based on the weights on the arcs. A serial connection between two routers is represented by a pair of arcs, one in each direction. Their weights may be different. A multiaccess network is represented by a node for the network itself plus a node for each router. The arcs from the network node to the routers have weight 0 and are omitted from the graph.

Figure 5-52(b) shows the graph representation of the network of Fig. 5-52(a). What OSPF fundamentally does is represent the actual network as a graph like this and then compute the shortest path from every router to every other router.

Many of the ASes in the Internet are themselves large and nontrivial to manage. OSPF allows them to be divided up into numbered **areas**, where an area is a network or a set of contiguous networks. Areas do not overlap but need not be exhaustive, that is, some routers may belong to no area. An area is a generalization of a subnet. Outside an area, its topology and details are not visible.

Every AS has a **backbone** area, called area 0. All areas are connected to the backbone, possibly by tunnels, so it is possible to go from any area in the AS to any other area in the AS via the backbone. A tunnel is represented in the graph as an arc and has a cost. Each router that is connected to two or more areas is part of the backbone. As with other areas, the topology of the backbone is not visible outside the backbone.

Within an area, each router has the same link state database and runs the same shortest path algorithm. Its main job is to calculate the shortest path from itself to

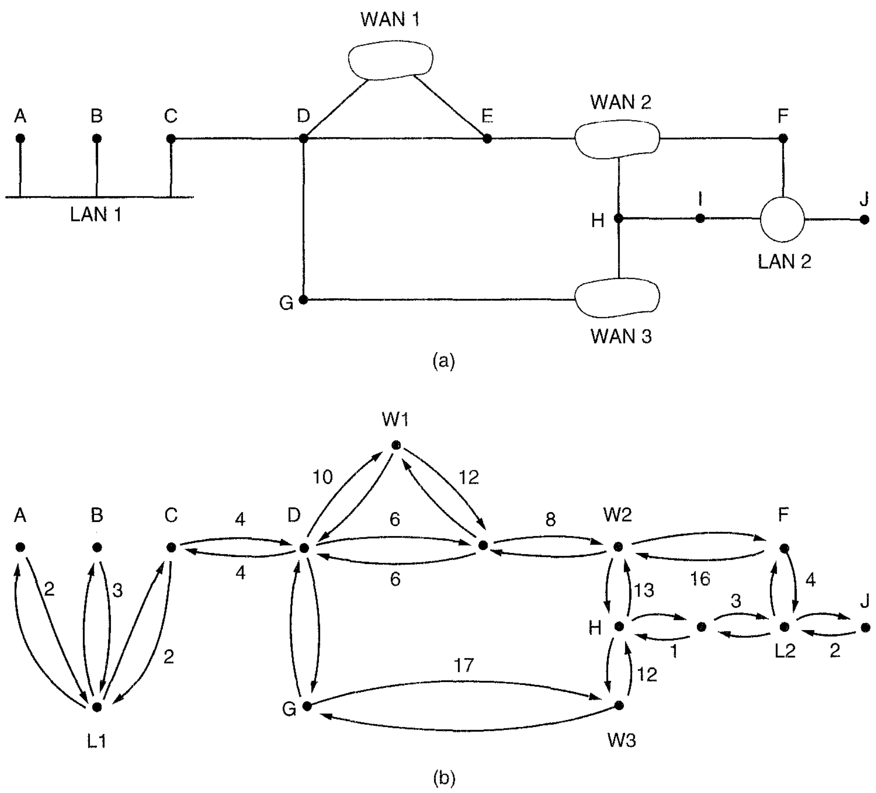


Fig. 5-52. (a) An autonomous system. (b) A graph representation of (a).

every other router in the area, including the router that is connected to the backbone, of which there must be at least one. A router that connects to two areas needs the databases for both areas and must run the shortest path algorithm for each one separately.

The way OSPF handles type of service routing is to have multiple graphs, one labeled with the costs when delay is the metric, one labeled with the costs when throughput is the metric, and one labeled with the costs when reliability is the metric. Although this triples the computation needed, it allows separate routes for optimizing delay, throughput, and reliability.

During normal operation, three kinds of routes may be needed: intra-area, interarea, and interAS. Intra-area routes are the easiest, since the source router already knows the shortest path to the destination router. Interarea routing always proceeds in three steps: go from the source to the backbone; go across the backbone to the destination area; go to the destination. This algorithm forces a star configuration on OSPF with the backbone being the hub and the other areas being spokes. Packets are routed from source to destination "as is." They are not encapsulated or tunneled, unless going to an area whose only connection to the backbone is a tunnel. Figure 5-53 shows part of the Internet with ASes and areas.

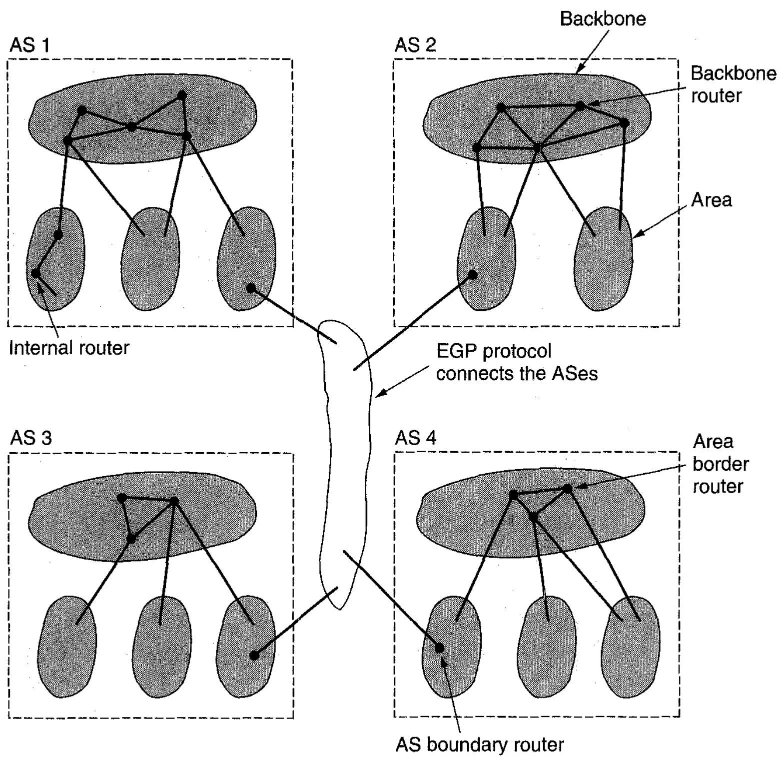


Fig. 5-53. The relation between ASes, backbones, and areas in OSPF.

OSPF distinguishes four classes of routers:

1. Internal routers are wholly within one area.
2. Area border routers connect two or more areas.
3. Backbone routers are on the backbone.
4. AS boundary routers talk to routers in other ASes.

These classes are allowed to overlap. For example, all the border routers are automatically part of the backbone. In addition, a router that is in the backbone but not part of any other area is also an internal router. Examples of all four classes of routers are illustrated in Fig. 5-53.

When a router boots, it sends HELLO messages on all of its point-to-point lines and multicasts them on LANs to the group consisting of all the other routers. On WANs, it needs some configuration information to know who to contact. From the responses, each router learns who its neighbors are.

OSPF works by exchanging information between **adjacent** routers, which is not the same as between neighboring routers. In particular, it is inefficient to have every router on a LAN talk to every other router on the LAN. To avoid this situation, one router is elected as the **designated router**. It is said to be adjacent to all the other routers, and exchanges information with them. Neighboring routers that are not adjacent do not exchange information with each other. A backup designated router is always kept up to date to ease the transition should the primary designated router crash.

During normal operation, each router periodically floods LINK STATE UPDATE messages to each of its adjacent routers. This message gives its state and provides the costs used in the topological database. The flooding messages are acknowledged, to make them reliable. Each message has a sequence number, so a router can see whether an incoming LINK STATE UPDATE is older or newer than what it currently has. Routers also send these messages when a line goes up or down or its cost changes.

DATABASE DESCRIPTION messages give the sequence numbers of all the link state entries currently held by the sender. By comparing its own values with those of the sender, the receiver can determine who has the most recent values. These messages are used when a line is brought up.

Either partner can request link state information from the other one using LINK STATE REQUEST messages. The net result of this algorithm is that each pair of adjacent routers checks to see who has the most recent data, and new information is spread throughout the area this way. All these messages are sent as raw IP packets. The five kinds of messages are summarized in Fig. 5-54.

Message type	Description
Hello	Used to discover who the neighbors are
Link state update	Provides the sender's costs to its neighbors
Link state ack	Acknowledges link state update
Database description	Announces which updates the sender has
Link state request	Requests information from the partner

Fig. 5-54. The five types of OSPF messages.

Finally, we can put all the pieces together. Using flooding, each router informs all the other routers in its area of its neighbors and costs. This information allows each router to construct the graph for its area(s) and compute the shortest path. The backbone area does this too. In addition, the backbone routers accept information from the area border routers in order to compute the best route from each backbone router to every other router. This information is propagated back to the area border routers, which advertise it within their areas. Using this information, a router about to send an interarea packet can select the best exit router to the backbone.

5.5.6. The Exterior Gateway Routing Protocol: BGP

Within a single AS, the recommended routing protocol on the Internet is OSPF (although it is certainly not the only one in use). Between ASes, a different protocol, **BGP** (**Border Gateway Protocol**), is used. A different protocol is needed between ASes because the goals of an interior gateway protocol and an exterior gateway protocol are not the same. All an interior gateway protocol has to do is move packets as efficiently as possible from the source to the destination. It does not have to worry about politics.

Exterior gateway protocol routers have to worry about politics a great deal. For example, a corporate AS might want the ability to send packets to any Internet site and receive packets from any Internet site. However, it might be unwilling to carry transit packets originating in a foreign AS and ending in a different foreign AS, even if its own AS was on the shortest path between the two foreign ASes ("That's their problem, not ours"). On the other hand, it might be willing to carry transit traffic for its neighbors, or even for specific other ASes that paid it for this service. Telephone companies, for example, might be happy to act as a carrier for their customers, but not for others. Exterior gateway protocols in general, and BGP in particular, have been designed to allow many kinds of routing policies to be enforced in the interAS traffic.

Typical policies involve political, security, or economic considerations. A few examples of routing constraints are

1. No transit traffic through certain ASes.
2. Never put Iraq on a route starting at the Pentagon.
3. Do not use the United States to get from British Columbia to Ontario.
4. Only transit Albania if there is no alternative to the destination.
5. Traffic starting or ending at IBM® should not transit Microsoft®.

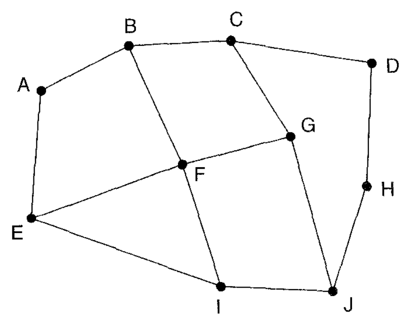
Policies are manually configured into each BGP router. They are not part of the protocol itself.

From the point of view of a BGP router, the world consists of other BGP routers and the lines connecting them. Two BGP routers are considered connected if they share a common network. Given BGP's special interest in transit traffic, networks are grouped into one of three categories. The first category is the **stub networks**, which have only one connection to the BGP graph. These cannot be used for transit traffic because there is no one on the other side. Then come the **multiconnected networks**. These could be used for transit traffic, except that they refuse. Finally, there are the **transit networks**, such as backbones, which are willing to handle third-party packets, possibly with some restrictions.

Pairs of BGP routers communicate with each other by establishing TCP connections. Operating this way provides reliable communication and hides all the details of the network being passed through.

BGP is fundamentally a distance vector protocol, but quite different from most others such as RIP. Instead of maintaining just the cost to each destination, each BGP router keeps track of the exact path used. Similarly, instead of periodically giving each neighbor its estimated cost to each possible destination, each BGP router tells its neighbors the exact path it is using.

As an example, consider the BGP routers shown in Fig. 5-55(a). In particular, consider *F*'s routing table. Suppose that it uses the path *FGCD* to get to *D*. When the neighbors give it routing information, they provide their complete paths, as shown in Fig. 5-55(b) (for simplicity, only destination *D* is shown here).



(a)

Information F receives
from its neighbors about D

From B: "I use BCD"
From G: "I use GCD"
From I: "I use IFGCD"
From E: "I use EFGCD"

(b)

Fig. 5-55. (a) A set of BGP routers. (b) Information sent to *F*.

After all the paths come in from the neighbors, *F* examines them to see which is the best. It quickly discards the paths from *I* and *E*, since these paths pass through *F* itself. The choice is then between using *B* and *G*. Every BGP router contains a module that examines routes to a given destination and scores them, returning a number for the "distance" to that destination for each route. Any route violating a policy constraint automatically gets a score of infinity. The router then adopts the route with the shortest distance. The scoring function is not part of the BGP protocol and can be any function the system managers want.

BGP easily solves the count-to-infinity problem that plagues other distance vector routing algorithms. For example, suppose *G* crashes or the line *FG* goes down. *F* then receives routes from its three remaining neighbors. These routes are *BCD*, *IFGCD*, and *EFGCD*. It can immediately see that the two latter routes are pointless, since they pass through *F* itself, so it chooses *FBCD* as its new route. Other distance vector algorithms often make the wrong choice because they cannot tell which of their neighbors have independent routes to the destination, and which do not. The current definition of BGP is in RFC 1654. Additional useful information can be found in RFC 1268.

5.5.7. Internet Multicasting

Normal IP communication is between one sender and one receiver. However, for some applications it is useful for a process to be able to send to a large number of receivers simultaneously. Examples are updating replicated, distributed databases, transmitting stock quotes to multiple brokers, and handling digital conference (i.e., multiparty) telephone calls.

IP supports multicasting, using class D addresses. Each class D address identifies a group of hosts. Twenty-eight bits are available for identifying groups, so over 250 million groups can exist at the same time. When a process sends a packet to a class D address, a best-efforts attempt is made to deliver it to all the members of the group addressed, but no guarantees are given. Some members may not get the packet.

Two kinds of group addresses are supported: permanent addresses and temporary ones. A permanent group is always there and does not have to be set up. Each permanent group has a permanent group address. Some examples of permanent group addresses are

224.0.0.1 All systems on a LAN
224.0.0.2 All routers on a LAN
224.0.0.5 All OSPF routers on a LAN
224.0.0.6 All designated OSPF routers on a LAN

Temporary groups must be created before they can be used. A process can ask its host to join a specific group. It can also ask its host to leave the group. When the last process on a host leaves a group, that group is no longer present on the host. Each host keeps track of which groups its processes currently belong to.

Multicasting is implemented by special multicast routers, which may or may not be colocated with the standard routers. About once a minute, each multicast router sends a hardware (i.e., data link layer) multicast to the hosts on its LAN (address 224.0.0.1) asking them to report back on the groups their processes currently belong to. Each host sends back responses for all the class D addresses it is interested in.

These query and response packets use a protocol called **IGMP (Internet Group Management Protocol)**, which is vaguely analogous to ICMP. It has only two kinds of packets: query and response, each with a simple fixed format containing some control information in the first word of the payload field and a class D address in the second word. It is described in RFC 1112.

Multicast routing is done using spanning trees. Each multicast router exchanges information with its neighbors using a modified distance vector protocol in order for each one to construct a spanning tree per group covering all group members. Various optimizations are used to prune the tree to eliminate routers and networks not interested in particular groups. The protocol makes heavy use of tunneling to avoid bothering nodes not in a spanning tree.

5.5.8. Mobile IP

Many users of the Internet have portable computers and want to stay connected to the Internet when they visit a distant Internet site and even on the road in between. Unfortunately, the IP addressing system makes working far from home easier said than done. In this section we will examine the problem and the solution. A more detailed description is given in (Johnson, 1995).

The real villain is the addressing scheme itself. Every IP address contains three fields: the class, the network number, and the host number. For example, consider the machine with IP address 160.80.40.20. The 160.80 gives the class (B) and network number (8272); the 40.20 is the host number (10260). Routers all over the world have routing tables telling which line to use to get to network 160.80. Whenever a packet comes in with a destination IP address of the form 160.80.xxx.yyy, it goes out on that line.

If all of a sudden, the machine with that address is carted off to some distant site, the packets for it will continue to be routed to its home LAN (or router). The owner will no longer get email, and so on. Giving the machine a new IP address corresponding to its new location is unattractive because large numbers of people, programs, and databases would have to be informed of the change.

Another approach is to have the routers use complete IP addresses for routing, instead of just the class and network. However, this strategy would require each router to have millions of table entries, at astronomical cost to the Internet.

When people began demanding the ability to have mobile hosts, the IETF set up a Working Group to find a solution. The Working Group quickly formulated a number of goals considered desirable in any solution. The major ones were

1. Each mobile host must be able to use its home IP address anywhere.
2. Software changes to the fixed hosts were not permitted.
3. Changes to the router software and tables were not permitted.
4. Most packets for mobile hosts should not make detours on the way.
5. No overhead should be incurred when a mobile host is at home.

The solution chosen was the one described in Sec. 5.2.8. To review it briefly, every site that wants to allow its users to roam has to create a home agent. Every site that wants to allow visitors has to create a foreign agent. When a mobile host shows up at a foreign site, it contacts the foreign host there and registers. The foreign host then contacts the user's home agent and gives it a **care-of address**, normally the foreign agent's own IP address.

When a packet arrives at the user's home LAN, it comes in at some router attached to the LAN. The router then tries to locate the host in the usual way, by broadcasting an ARP packet asking, for example: "What is the Ethernet address of 160.80.40.20?" The home agent responds to this query by giving its own Ethernet address. The router then sends packets for 160.80.40.20 to the home agent. It, in turn, tunnels them to the care-of address by encapsulating them in the payload field of an IP packet addressed to the foreign agent. The foreign agent then decapsulates and delivers them to the data link address of the mobile host. In addition, the home agent gives the care-of address to the sender, so future packets can be tunnelled directly to the foreign agent. This solution meets all the requirements stated above.

One small detail is probably worth mentioning. At the time the mobile host moves, the router probably has its (soon-to-be-invalid) Ethernet address cached. To replace that Ethernet address with the home agent's, a trick called **gratuitious ARP** is used. This is a special, unsolicited message to the router that causes it to replace a specific cache entry, in this case, that of the mobile host about to leave. When the mobile host returns later, the same trick is used to update the router's cache again.

Nothing in the design prevents a mobile host from being its own foreign agent, but that approach only works if the mobile host (in its capacity as foreign agent) is logically connected to the Internet at its current site. Also, it must be able to acquire a (temporary) care-of IP address to use. That IP address must belong to the LAN to which it is currently attached.

The IETF solution for mobile hosts solves a number of other problems not mentioned so far. For example, how are agents located? The solution is for each agent to periodically broadcast its address and the type of services it is willing to provide (e.g., home, foreign, or both). When a mobile host arrives somewhere, it can just listen for these broadcasts, called **advertisements**. Alternatively, it can broadcast a packet announcing its arrival and hope that the local foreign agent responds to it.

Another problem that had to be solved is what to do about impolite mobile hosts that leave without saying goodbye. The solution is to make registration valid only for a fixed time interval. If it is not refreshed periodically, it times out, so the foreign host can clear its tables.

Yet another issue is security. When a home agent gets a message asking it to please forward all of Nora's packets to some IP address, it had better not comply unless it is convinced that Nora is the source of this request, and not somebody trying to impersonate her. Cryptographic authentication protocols are used for this purpose. We will study such protocols in Chap. 7.

A final point addressed by the Working Group relates to levels of mobility. Imagine an airplane with an on-board Ethernet used by the navigation and avionics computers. On this Ethernet is a standard router that talks to the wired Internet on the ground over a radio link. One fine day, some clever marketing executive gets the idea to install Ethernet connectors in all the arm rests so passengers with mobile computers can also plug in.

Now we have two levels of mobility: the aircraft's own computers, which are stationary with respect to the Ethernet, and the passengers' computers, which are mobile with respect to it. In addition, the on-board router is mobile with respect to routers on the ground. Being mobile with respect to a system that is itself mobile can be handled using recursive tunneling.

5.5.9. CIDR—Classless InterDomain Routing

IP has been in heavy use for over a decade. It has worked extremely well, as demonstrated by the exponential growth of the Internet. Unfortunately, IP is rapidly becoming a victim of its own popularity: it is running out of addresses. This looming disaster has sparked a great deal of discussion and controversy within the Internet community about what to do about it. In this section we will describe both the problem and several proposed solutions. A more complete description is given in (Huitema, 1996).

Back in 1987, a few visionaries predicted that some day the Internet might grow to 100,000 networks. Most experts pooh-poohed this as being decades in the future, if ever. The 100,000th network was connected in 1996. The problem, simply stated, is that the Internet is rapidly running out of IP addresses. In principle, over 2 billion addresses exist, but the practice of organizing the address space by classes (see Fig. 5-47), wastes millions of them. In particular, the real villain is the class B network. For most organizations, a class A network, with 16 million addresses is too big, and a class C network, with 256 addresses is too small. A class B network, with 65,536, is just right. In Internet folklore, this situation is known as the **three bears problem** (as in *Goldilocks and the Three Bears*).

In reality, a class B address is far too large for most organizations. Studies have shown that more than half of all class B networks have fewer than 50 hosts. A class C network would have done the job, but no doubt every organization that asked for a class B address thought that one day it would outgrow the 8-bit host field. In retrospect, it might have been better to have had class C networks use 10 bits instead of eight for the host number, allowing 1022 hosts per network. Had this been the case, most organizations would have probably settled for a class C network, and there would have been half a million of them (versus only 16,384 class B networks).

However, then another problem would have emerged more quickly: the routing table explosion. From the point of view of the routers, the IP address space is a two-level hierarchy, with network numbers and host numbers. Routers do not have to know about all the hosts, but they do have to know about all the networks. If half a million class C networks were in use, every router in the entire Internet would need a table with half a million entries, one per network, telling which line to use to get to that network, as well as other information.

The actual physical storage of half a million entry tables is probably doable, although expensive for critical routers that keep the tables in static RAM on I/O boards. A more serious problem is that the complexity of various algorithms relating to management of the tables grows faster than linear. Worse yet, much of the existing router software and firmware was designed at a time when the Internet had 1000 connected networks and 10,000 networks seemed decades away. Design choices made then often are far from optimal now.

In addition, various routing algorithms require each router to transmit its tables periodically. The larger the tables, the more likely some parts will get lost underway, leading to incomplete data at the other end and possibly routing instabilities.

The routing table problem could have been solved by going to a deeper hierarchy. For example, having each IP address contain a country, state, city, network, and host field might work. Then each router would only need to know how to get to each country, the states or provinces in its own country, the cities in its state or province, and the networks in its city. Unfortunately, this solution would require considerably more than 32 bits for IP addresses and would use addresses inefficiently (Liechtenstein would have as many bits as the United States).

In short, most solutions solve one problem but create a new one. One solution that is now being implemented and which will give the Internet a bit of extra breathing room is **CIDR** (**Classless InterDomain Routing**). The basic idea behind CIDR, which is described in RFC 1519, is to allocate the remaining class C networks, of which there are almost two million, in variable-sized blocks. If a site needs, say, 2000 addresses, it is given a block of 2048 addresses (eight contiguous class C networks), and not a full class B address. Similarly, a site needing 8000 addresses gets 8192 addresses (32 contiguous class C networks).

In addition to using blocks of contiguous class C networks as units, the allocation rules for the class C addresses were also changed in RFC 1519. The world was partitioned into four zones, and each one given a portion of the class C address space. The allocation was as follows:

Addresses 194.0.0.0 to 195.255.255.255 are for Europe
Addresses 198.0.0.0 to 199.255.255.255 are for North America
Addresses 200.0.0.0 to 201.255.255.255 are for Central and South America
Addresses 202.0.0.0 to 203.255.255.255 are for Asia and the Pacific

In this way, each region was given about 32 million addresses to allocate, with another 320 million class C addresses from 204.0.0.0 through 223.255.255.255

held in reserve for the future. The advantage of this allocation is that now any router outside of Europe that gets a packet addressed to 194.xx.yy.zz or 195.xx.yy.zz can just send it to its standard European gateway. In effect 32 million addresses have now been compressed into one routing table entry. Similarly for the other regions.

Of course, once a 194.xx.yy.zz packet gets to Europe, more detailed routing tables are needed. One possibility is to have 131,072 entries for networks 194.0.0.xx through 195.255.255.xx, but this is precisely this routing table explosion that we are trying to avoid. Instead, each routing table entry is extended by giving it a 32-bit mask. When a packet comes in, its destination address is first extracted. Then (conceptually) the routing table is scanned entry by entry, masking the destination address and comparing it to the table entry looking for a match.

To make this comparison process clearer, let us consider an example. Suppose that Cambridge University needs 2048 addresses and is assigned the addresses 194.24.0.0 through 194.24.7.255, along with mask 255.255.248.0. Next, Oxford University asks for 4096 addresses. Since a block of 4096 addresses must lie on a 4096-byte boundary, they cannot be given addresses starting at 194.8.0.0. Instead they get 194.24.16.0 through 194.24.31.255 along with mask 255.255.240.0. Now the University of Edinburgh asks for 1024 addresses and is assigned addresses 194.24.8.0 through 194.24.11.255 and mask 255.255.252.0.

The routing tables all over Europe are now updated with three entries, each one containing a base address and a mask. These entries (in binary) are:

Address	Mask
11000010 00011000 00000000 00000000	11111111 11111111 11111000 00000000
11000010 00011000 00010000 00000000	11111111 11111111 11110000 00000000
11000010 00011000 00001000 00000000	11111111 11111111 11111100 00000000

Now consider what happens when a packet comes in addressed to 194.24.17.4, which in binary is

11000010 00011000 00010001 00000100

First it is Boolean ANDed with the Cambridge mask to get

11000010 00011000 00010000 00000000

This value does not match the Cambridge base address, so the original address is next ANDed with the Oxford mask to get

11000010 00011000 00010000 00000000

This value does match the Oxford mask, so the packet is sent to the Oxford router. In practice, the router entries are not tried sequentially; indexing tricks are used to speed up the search. Also, it is possible for two entries to match, in which case the one whose mask has the most 1 bits wins. Finally, the same idea can be applied to all addresses, not just the new class C addresses, so with CIDR, the old class A, B, and C networks are no longer used for routing. This is why CIDR is called classless routing. CIDR is described in more detail in (Ford et al., 1993; and Huitema, 1995).

5.5.10. IPv6

While CIDR may buy a few more years' time, everyone realizes that the days of IP in its current form (IPv4) are numbered. In addition to these technical problems, there is another issue looming in the background. Up until recently, the Internet has been used largely by universities, high-tech industry, and the government (especially the Dept. of Defense). With the explosion of interest in the Internet starting in the mid 1990s, it is likely that in the next millenium, it will be used by a much larger group of people, especially people with different requirements. For one thing, millions of people with wireless portables may use it to keep in contact with their home bases. For another, with the impending convergence of the computer, communication, and entertainment industries, it may not be long before every television set in the world is an Internet node, producing a billion machines being used for video on demand. Under these circumstances, it became apparent that IP had to evolve and become more flexible.

Seeing these problems on the horizon, in 1990, IETF started work on a new version of IP, one which would never run out of addresses, would solve a variety of other problems, and be more flexible and efficient as well. Its major goals were to

1. Support billions of hosts, even with inefficient address space allocation.
2. Reduce the size of the routing tables.
3. Simplify the protocol, to allow routers to process packets faster.
4. Provide better security (authentication and privacy) than current IP.
5. Pay more attention to type of service, particularly for real-time data.
6. Aid multicasting by allowing scopes to be specified.
7. Make it possible for a host to roam without changing its address.
8. Allow the protocol to evolve in the future.
9. Permit the old and new protocols to coexist for years.

To find a protocol that met all these requirements, IETF issued a call for proposals and discussion in RFC 1550. Twenty-one responses were received, not all of them full proposals. By December 1992, seven serious proposals were on the table. They ranged from making minor patches to IP, to throwing it out altogether and replacing with a completely different protocol.

One proposal was to run TCP over CLNP, which, with its 160-bit addresses would have provided enough address space forever and would have unified two major network layer protocols. However, many people felt that this would have been an admission that something in the OSI world was actually done right, a statement considered Politically Incorrect in Internet circles. CLNP was patterned closely on IP, so the two are not really that different. In fact, the protocol ultimately chosen differs from IP far more than CLNP does. Another strike against CLNP was its poor support for service types, something required to transmit multimedia efficiently.

Three of the better proposals were published in *IEEE Network* (Deering, 1993; Francis, 1993; and Katz and Ford, 1993). After much discussion, revision, and jockeying for position, a modified combined version of the Deering and Francis proposals, by now called **SIPP** (**Simple Internet Protocol Plus**) was selected and given the designation **IPv6** (IPv5 was already in use for an experimental real-time stream protocol).

IPv6 meets the goals fairly well. It maintains the good features of IP, discards or deemphasizes the bad ones, and adds new ones where needed. In general, IPv6 is not compatible with IPv4, but it is compatible with all the other Internet protocols, including TCP, UDP, ICMP, IGMP, OSPF, BGP, and DNS, sometimes with small modifications being required (mostly to deal with longer addresses). The main features of IPv6 are discussed below. More information about it can be found in RFC 1883 through RFC 1887.

First and foremost, IPv6 has longer addresses than IPv4. They are 16 bytes long, which solves the problem that IPv6 was set out to solve: provide an effectively unlimited supply of Internet addresses. We will have more to say about addresses shortly.

The second major improvement of IPv6 is the simplification of the header. It contains only 7 fields (versus 13 in IPv4). This change allows routers to process packets faster and thus improve throughout. We will discuss the header shortly, too.

The third major improvement was better support for options. This change was essential with the new header because fields that previously were required are now optional. In addition, the way options are represented is different, making it simple for routers to skip over options not intended for them. This feature speeds up packet processing time.

A fourth area in which IPv6 represents a big advance is in security. IETF had its fill of newspaper stories about precocious 12-year-olds using their personal computers to break into banks and military bases all over the Internet. There was a strong feeling that something had to be done to improve security. Authentication and privacy are key features of the new IP.

Finally, more attention has been paid to type of service than in the past. IPv4 actually has an 8-bit field devoted to this matter, but with the expected growth in multimedia traffic in the future, much more is needed.

The Main IPv6 Header

The IPv6 header is shown in Fig. 5-56. The *Version* field is always 6 for IPv6 (and 4 for IPv4). During the transition period from IPv4, which will probably take a decade, routers will be able to examine this field to tell what kind of packet they have. As an aside, making this test wastes a few instructions in the critical path, so many implementations are likely to try to avoid it by using some field in the data link header to distinguish IPv4 packets from IPv6 packets. In this way, packets can be passed to the correct network layer handler directly. However, having the data link layer be aware of network packet types completely violates the design principle that each layer should not be aware of the meaning of the bits given to it from the layer above. The discussions between the "Do it right" and "Make it fast" camps will no doubt be lengthy and vigorous.

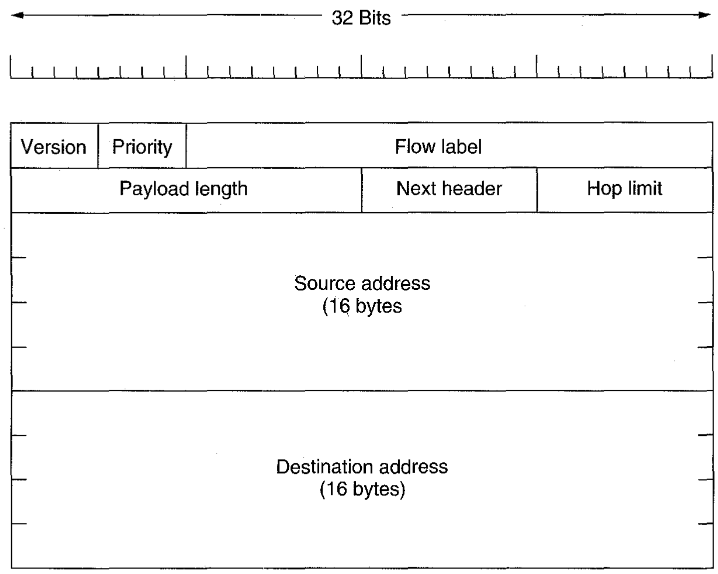


Fig. 5-56. The IPv6 fixed header (required).

The *Priority* field is used to distinguish between packets whose sources can be flow controlled and those that cannot. Values 0 through 7 are for transmissions that are capable of slowing down in the event of congestion. Values 8 through 15 are for real-time traffic whose sending rate is constant, even if all the packets are being lost. Audio and video fall into the latter category. This distinction allows routers to deal with packets better in the event of congestion. Within each group, lower-numbered packets are less important than higher-numbered ones. The IPv6 standard suggests, for example, to use 1 for news, 4 for FTP, and 6 for Telnet

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connections, since delaying a news packet for a few seconds is not noticeable, but delaying a Telnet packet certainly is.

The *Flow label* field is still experimental but will be used to allow a source and destination to set up a pseudoconnection with particular properties and requirements. For example, a stream of packets from one process on a certain source host to a certain process on a certain destination host might have stringent delay requirements and thus need reserved bandwidth. The flow can be set up in advance and given an identifier. When a packet with a nonzero *Flow label* shows up, all the routers can look it up in internal tables to see what kind of special treatment it requires. In effect, flows are an attempt to have it both ways: the flexibility of a datagram subnet and the guarantees of a virtual circuit subnet.

Each flow is designated by the source address, destination address, and flow number, so many flows may be active at the same time between a given pair of IP addresses. Also, in this way, even if two flows coming from different hosts but with the same flow number pass through the same router, the router will be able to tell them apart using the source and destination addresses. It is expected that flow numbers will be chosen randomly, rather than assigned sequentially starting at 1, to make it easy for routers to hash them.

The *Payload length* field tells how many bytes follow the 40-byte header of Fig. 5-56. The name was changed from the IPv4 *Total length* field because the meaning was changed slightly: the 40 header bytes are no longer counted as part of the length as they used to be.

The *Next header* field lets the cat out of the bag. The reason the header could be simplified is that there can be additional (optional) extension headers. This field tells which of the (currently) six extension headers, if any, follows this one. If this header is the last IP header, the *Next header* field tells which transport protocol handler (e.g., TCP, UDP) to pass the packet to.

The *Hop limit* field is used to keep packets from living forever. It is, in practice, the same as the *Time to live* field in IPv4, namely, a field that is decremented on each hop. In theory, in IPv4 it was a time in seconds, but no router used it that way, so the name was changed to reflect the way it is actually used.

Next come the *Source address* and *Destination address* fields. Deering's original proposal, SIP, used 8-byte addresses, but during the review process many people felt that with 8-byte addresses IPv6 would run out of addresses within a few decades, whereas with 16-byte addresses it would never run out. Other people argued that 16 bytes was overkill, whereas still others favored using 20-byte addresses to be compatible with the OSI datagram protocol. Another faction wanted variable-sized addresses. After much discussion, it was decided that fixed-length 16-byte addresses were the best compromise.

The IPv6 address space is divided up as shown in Fig. 5-57. Addresses beginning with 80 zeros are reserved for IPv4 addresses. Two variants are supported, distinguished by the next 16 bits. These variants relate to how IPv6 packets will be tunneled over the existing IPv4 infrastructure.

Prefix (binary)	Usage	Fraction
0000 0000	Reserved (including IPv4)	1/256
0000 0001	Unassigned	1/256
0000 001	OSI NSAP addresses	1/128
0000 010	Novell NetWare IPX addresses	1/128
0000 011	Unassigned	1/128
0000 1	Unassigned	1/32
0001	Unassigned	1/16
001	Unassigned	1/8
010	Provider-based addresses	1/8
011	Unassigned	1/8
100	Geographic-based addresses	1/8
101	Unassigned	1/8
110	Unassigned	1/8
1110	Unassigned	1/16
1111 0	Unassigned	1/32
1111 10	Unassigned	1/64
1111 110	Unassigned	1/128
1111 1110 0	Unassigned	1/512
1111 1110 10	Link local use addresses	1/1024
1111 1110 11	Site local use addresses	1/1024
1111 1111	Multicast	1/256

Fig. 5-57. IPv6 addresses

The use of separate prefixes for provider-based and geographic-based addresses is a compromise between two different visions of the future of the Internet. Provider-based addresses make sense if you think that in the future there will be some number of companies providing Internet service to customers, analogous to AT&T, MCI, Sprint, British Telecom, and so on providing telephone service now. Each of these companies will be given some fraction of the address space. The first 5 bits following the 010 prefix are used to indicate which registry to look the provider up in. Currently three registries are operating, for North America, Europe, and Asia. Up to 29 new registries can be added later.

Each registry is free to divide up the remaining 15 bytes as it sees fit. It is expected that many of them will use a 3-byte provider number, giving about 16 million providers, in order to allow large companies to act as their own provider. Another possibility is to use 1 byte to indicate national providers and let them do further allocation. In this manner, additional levels of hierarchy can be introduced as needed.

The geographic model is the same as the current Internet, in which providers do not play a large role. In this way, IPv6 can handle both kinds of addresses.

The link and site local addresses have only a local significance. They can be reused at each organization without conflict. They cannot be propagated outside organizational boundaries, making them well suited to organizations that currently use firewalls to wall themselves off from the rest of the Internet.

Multicast addresses have a 4-bit flag field and a 4-bit scope field following the prefix, then a 112-bit group identifier. One of the flag bits distinguishes permanent from transient groups. The scope field allows a multicast to be limited to the current link, site, organization, or planet. These four scopes are spread out over the 16 values to allow new scopes to be added later. For example, the planetary scope is 14, so code 15 is available to allow future expansion of the Internet to other planets, solar systems, and galaxies.

In addition to supporting the standard unicast (point-to-point) and multicast addresses, IPv6 also supports a new kind of addressing: anycast. **Anycasting** is like multicasting in that the destination is a group of addresses, but instead of trying to deliver the packet to all of them, it tries to deliver it to just one, usually the nearest one. For example, when contacting a group of cooperating file servers, a client can use anycast to reach the nearest one, without having to know which one that is. Anycasting uses regular unicast addresses. It is up to the routing system to choose the lucky host that gets the packet.

A new notation has been devised for writing 16-byte addresses. They are written as eight groups of four hexadecimal digits with colons between the groups, like this:

8000:0000:0000:0000:0123:4567:89AB:CDEF

Since many addresses will have many zeros inside them, three optimizations have been authorized. First, leading zeros within a group can be omitted, so 0123 can be written as 123. Second, one or more groups of 16 zeros can be replaced by a pair of colons. Thus the above address now becomes

8000::123:4567:89AB:CDEF

Finally, IPv4 addresses can be written as a pair of colons and an old dotted decimal number, for example

::192.31.20.46

Perhaps it is unnecessary to be so explicit about it, but there are a lot of 16-byte addresses. Specifically, there are 2^{128} of them, which is approximately

3×10^{38}. If the entire earth, land and water, were covered with computers, IPv6 would allow 7×10^{23} IP addresses per square meter. Students of chemistry will notice that this number is larger than Avogadro's number. While it was not the intention to give every molecule on the surface of the earth its own IP address, we are not that far off.

In practice, the address space will not be used efficiently, just as the telephone number address space is not (the area code for Manhattan, 212, is nearly full, but that for Wyoming, 307, is nearly empty). In RFC 1715, Huitema calculated that using the allocation of telephone numbers as a guide, even in the most pessimistic scenario, there will still be well over 1000 IP addresses per square meter of the earth's surface (land and water). In any likely scenario, there will be trillions of them per square meter. In short, it seems unlikely that we will run out in the foreseeable future. It is also worth noting that only 28 percent of the address space has been allocated so far. The other 72 percent is available for future purposes not yet thought of.

It is instructive to compare the IPv4 header (Fig. 5-45) with the IPv6 header (Fig. 5-56) to see what has been left out in IPv6. The *IHL* field is gone because the IPv6 header has a fixed length. The *Protocol* field was taken out because the *Next header* field tells what follows the last IP header (e.g., a UDP or TCP segment).

All the fields relating to fragmentation were removed because IPv6 takes a different approach to fragmentation. To start with, all IPv6 conformant hosts and routers must support packets of 576 bytes. This rule makes fragmentation less likely to occur in the first place. In addition, when a host sends an IPv6 packet that is too large, instead of fragmenting it, the router that is unable to forward it sends back an error message. This message tells the host to break up all future packets to that destination. Having the host send packets that are the right size in the first place is ultimately much more efficient than having the routers fragment them on the fly.

Finally, the *Checksum* field is gone because calculating it greatly reduces performance. With the reliable networks now used, combined with the fact that the data link layer and transport layers normally have their own checksums, the value of yet another checksum was not worth the performance price it extracted. Removing all these features has resulted in a lean and mean network layer protocol. Thus the goal of IPv6—a fast, yet flexible, protocol with plenty of address space—has been met by this design.

Extension Headers

Nevertheless, some of the missing fields are occasionally still needed so IPv6 has introduced the concept of an (optional) **extension header**. These headers can be supplied to provide extra information, but encoded in an efficient way. Six kinds of extension headers are defined at present, as listed in Fig. 5-58. Each one is optional, but if more than one is present, they must appear directly after the fixed header, and preferably in the order listed.

Extension header	Description
Hop-by-hop options	Miscellaneous information for routers
Routing	Full or partial route to follow
Fragmentation	Management of datagram fragments
Authentication	Verification of the sender's identity
Encrypted security payload	Information about the encrypted contents
Destination options	Additional information for the destination

Fig. 5-58. IPv6 extension headers.

Some of the headers have a fixed format; others contain a variable number of variable-length fields. For these, each item is encoded as a (Type, Length, Value) tuple. The *Type* is a 1-byte field telling which option this is. The *Type* values have been chosen so that the first 2 bits tell routers that do not know how to process the option what to do. The choices are: skip the option, discard the packet, discard the packet and send back an ICMP packet, and the same as the previous one, except do not send ICMP packets for multicast addresses (to prevent one bad multicast packet from generating millions of ICMP reports).

The *Length* is also a 1-byte field. It tells how long the value is (0 to 255 bytes). The *Value* is any information required, up to 255 bytes.

The hop-by-hop header is used for information that all routers along the path must examine. So far, one option has been defined: support of datagrams exceeding 64K. The format of this header is shown in Fig. 5-59.

<table>
<tr><td>Next header</td><td>0</td><td>194</td><td>0</td></tr>
<tr><td colspan="4">Jumbo payload length</td></tr>
</table>

Fig. 5-59. The hop-by-hop extension header for large datagrams (jumbograms).

As with all extension headers, this one starts out with a byte telling what kind of header comes next. This byte is followed by one telling how long the hop-by-hop header is in bytes, excluding the first 8 bytes, which are mandatory. The next 2 bytes indicate that this option defines the datagram size (code 194) as a 4-byte number. The last 4 bytes give the size of the datagram. Sizes less than 65,536 are not permitted and will result in the first router discarding the packet and sending back an ICMP error message. Datagrams using this header extension are called

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jumbograms. The use of jumbograms is important for supercomputer applications that must transfer gigabytes of data efficiently across the Internet.

The routing header lists one or more routers that must be visited on the way to the destination. Both strict routing (the full path is supplied) and loose routing (only selected routers are supplied) are available, but they are combined. The format of the routing header is shown in Fig. 5-60.

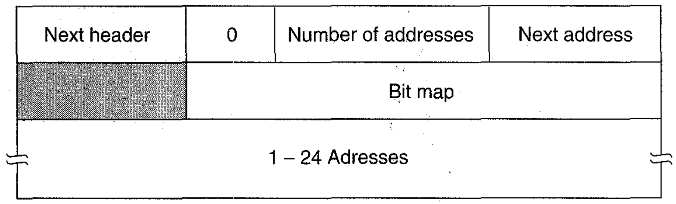

Fig. 5-60. The extension header for routing.

The first 4 bytes of the routing extension header contain four 1-byte integers: the next header type, the routing type (currently 0), the number of addresses present in this header (1 to 24), and the index of the next address to visit. The latter field starts at 0 and is incremented as each address is visited.

Then comes a reserved byte followed by a bit map with bits for each of the 24 potential IPv6 addresses following it. These bits tell whether each address must be visited directly after the one before it (strict source routing), or whether other routers may come in between (loose source routing).

The fragment header deals with fragmentation similarly to the way IPv4 does. The header holds the datagram identifier, fragment number, and a bit telling whether more fragments will follow. In IPv6, unlike in IPv4, only the source host can fragment a packet. Routers along the way may not do this. Although this change is a major philosophical break with the past, it simplifies the routers' work and makes routing go faster. As mentioned above, if a router is confronted with a packet that is too big, it discards the packet and sends an ICMP packet back to the source. This information allows the source host to fragment the packet into smaller pieces using this header and try again.

The authentication header provides a mechanism by which the receiver of a packet can be sure of who sent it. With IPv4, no such guarantee is present. The encrypted security payload makes it possible to encrypt the contents of a packet so that only the intended recipient can read it. These headers use cryptographic techniques to accomplish their missions. We will give brief descriptions below, but readers not already familiar with modern cryptography may not understand the full description now. They should come back after having read Chap. 7 (in particular, Sec. 7.1), which treats cryptographic protocols.

When a sender and receiver wish to communicate securely, they must first agree on one or more secret keys that only they know. How they do this is outside

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the scope of IPv6. Each of these keys is assigned a unique 32-bit key number. The key numbers are global, so that if Alice is using key 4 to talk to Bob, she cannot also have a key 4 to talk to Carol. Associated with each key number are other parameters, such as key lifetime, and so on.

To send an authenticated message, the sender first constructs a packet consisting of all the IP headers and the payload and then replaces the fields that change underway (e.g., *Hop limit*) with zeros. The packet is then padded out with zeros to a multiple of 16 bytes. Similarly, the secret key to be used is also padded out with zeros to a multiple of 16 bytes. Now a cryptographic checksum is computed on the concatenation of the padded secret key, the padded packet, and the padded secret key again. Users may define their own cryptographic checksum algorithms, but cryptographically unsophisticated users should use the default algorithm, MD5.

Now we come to the role of the authentication header. Basically, it contains three parts. The first part consists of 4 bytes holding the next header number, the length of the authentication header, and 16 zero bits. Then comes the 32-bit key number. Finally, the MD5 (or other) checksum is included.

The receiver then uses the key number to find the secret key. The padded version of it is then prepended and appended to the padded payload, the variable header fields are zeroed out, and the checksum computed. If it agrees with the checksum included in the authentication header, the receiver can be sure that the packet came from the sender with whom the secret key is shared and also be sure that the packet was not tampered with underway. The properties of MD5 make it computationally infeasible for an intruder to forge the sender's identity or modify the packet in a way that escapes detection.

It is important to note that the payload of an authenticated packet is sent unencrypted. Any router along the way can read what it says. For many applications, secrecy is not really important, just authentication. For example, if a user instructs his bank to pay his telephone bill, there is probably no real need for secrecy, but there is a very real need for the bank to be absolutely sure it knows who sent the packet containing the payment order.

For packets that must be sent secretly, the encrypted security payload extension header is used. It starts out with a 32-bit key number, followed by the encrypted payload. The encryption algorithm is up to the sender and receiver, but DES in cipher block chaining mode is the default. When DES-CBC is used, the payload field starts out with the initialization vector (a multiple of 4 bytes), then the payload, then padding out to multiple of 8 bytes. If both encryption and authentication are desired, both headers are needed.

The destination options header is intended for fields that need only be interpreted at the destination host. In the initial version of IPv6, the only options defined are null options for padding this header out to a multiple of 8 bytes, so initially it will not be used. It was included to make sure that new routing and host software can handle it, in case someone thinks of a destination option some day.

Controversies

Given the open design process and the strongly-held opinions of many of the people involved, it should come as no surprise that many choices made for IPv6 were highly controversial. We will summarize a few of these below. For all the gory details, see (Huitema, 1996).

We have already mentioned the argument about the address length. The result was a compromise: 16-byte fixed-length addresses.

Another fight developed over the length of the *Hop limit* field. One camp felt strongly that limiting the maximum number of hops to 255 (implicit in using an 8-bit field) was a gross mistake. After all, paths of 32 hops are common now, and 10 years from now much longer paths may be common. These people argued that using a huge address size was farsighted but using a tiny hop count was shortsighted. In their view, the greatest sin a computer scientist can commit is to provide too few bits somewhere.

The response was that arguments could be made to increase every field, leading to a bloated header. Also, the function of the *Hop limit* field is to keep packets from wandering around for a long time and 65,535 hops is far too long. Finally, as the Internet grows, more and more long-distance links will be built, making it possible to get from any country to any other country in half a dozen hops at most. If it takes more than 125 hops to get from the source and destination to their respective international gateways, something is wrong with the national backbones. The 8-bitters won this one.

Another hot potato was the maximum packet size. The supercomputer community wanted packets in excess of 64 KB. When a supercomputer gets started transferring, it really means business and does not want to be interrupted every 64 KB. The argument against large packets is that if a 1-MB packet hits a 1.5-Mbps T1 line, that packet will tie the line up for over 5 seconds, producing a very noticeable delay for interactive users sharing the line. A compromise was reached here: normal packets are limited to 64 KB, but the hop-by-hop extension header can be used to permit jumbograms.

A third hot topic was removing the IPv4 checksum. Some people likened this move to removing the brakes from a car. Doing so makes the car lighter so it can go faster, but if an unexpected event happens, you have a problem.

The argument against checksums was that any application that really cares about data integrity has to have a transport layer checksum anyway, so having another one in IP (in addition to the data link layer checksum) is overkill. Furthermore, experience showed that computing the IP checksum was a major expense in IPv4. The antichecksum camp won this one, and IPv6 does not have a checksum.

Mobile hosts were also a point of contention. If a portable computer flies halfway around the world, can it continue operating at the destination with the same IPv6 address, or does it have to use a scheme with home agents and foreign agents? Mobile hosts also introduce asymmetries into the routing system. It may well be the case that a small mobile computer can easily hear the powerful signal put out by a large stationary router, but the stationary router cannot hear the feeble signal put out by the mobile host. Consequently, some people wanted to build explicit support for mobile hosts into IPv6. That effort failed when no consensus could be found for any specific proposal.

Probably the biggest battle was about security. Everyone agreed it was needed. The war was about where and how. First where. The argument for putting it in the network layer is that it then becomes a standard service that all applications can use without any advance planning. The argument against it is that really secure applications generally want nothing less than end-to-end encryption, where the source application does the encryption and the destination application undoes it. With anything less, the user is at the mercy of potentially buggy network layer implementations over which he has no control. The response to this argument is that these applications can just refrain from using the IP security features and do the job themselves. The rejoinder to that is that the people who do not trust the network to do it right, do not want to have to pay the price of slow, bulky IP implementations that have this capability, even if it is disabled.

Another aspect of where to put security relates to the fact that many (but not all) countries have stringent export laws concerning cryptography. Some, notably France and Iraq, also greatly restrict its use domestically, so that people cannot have secrets from the police. As a result, any IP implementation that used a cryptographic system strong enough to be of much value could not be exported from the United States (and many other countries) to customers worldwide. Having to maintain two sets of software, one for domestic use and one for export, is something most computer vendors vigorously oppose.

One potential solution is for all vendors to move their cryptography shops to a country that does not regulate cryptography, such as Finland or Switzerland. Strong cryptographic software could be designed and manufactured there and then shipped legally to all countries except France and Iraq. The problem with this approach is that designing part of the router software in one country and part in another can lead to integration problems.

The final controversy concerning security relates to the choice of the default algorithms that all implementations must support. While MD5 was thought to be relatively secure, recent advances in cryptography may weaken it. No serious cryptographer believes that DES is secure against attacks by major governments, but it is probably good enough to foil even the most precocious 12-year-olds for the time being. The compromise was thus to mandate security in IPv6, use a state-of-the-art checksum algorithm for good authentication and a weakish algorithm for secrecy but give users the option of replacing these algorithms with their own.

One point on which there was no controversy is that no one expects the IPv4 Internet to be turned off on a Sunday morning and come back up as an IPv6

Internet Monday morning. Instead, isolated "islands" of IPv6 will be converted, initially communicating via tunnels. As the IPv6 islands grow, they will merge into bigger islands. Eventually, all the islands will merge, and the Internet will be fully converted. Given the massive investment in IPv4 routers currently deployed, the conversion process will probably take a decade. For this reason, an enormous amount of effort has gone into making sure that this transition will be as painless as possible.

5.6. THE NETWORK LAYER IN ATM NETWORKS

The layers of the ATM model (see Fig. 1-30) do not map onto the OSI layers especially well, which leads to ambiguities. The OSI data link layer deals with framing and transfer protocols between two machines on the same physical wire (or fiber). Data link layer protocols are single-hop protocols. They do not deal with end-to-end connections because switching and routing do not occur in the data link layer. About this there is no doubt.

The lowest layer that goes from source to destination, and thus involves routing and switching (i.e., is multihop), is the network layer. The ATM layer deals with moving cells from source to destination and definitely involves routing algorithms and protocols within the ATM switches. It also deals with global addressing. Thus functionally, the ATM layer performs the work expected of the network layer. The ATM layer is not guaranteed to be 100 percent reliable, but that is not required for a network layer protocol.

Also, the ATM layer resembles layer 3 of X.25, which everyone agrees is a network layer protocol. Depending on bit settings, the X.25 network layer protocol may or may not be reliable, but most implementations treat it as unreliable. Because the ATM layer has the functionality expected of the network layer, does not have the functionality expected of the data link layer, and is quite similar to existing network layer protocols, we will discuss the ATM layer in this chapter.

Confusion arises because many people in the ATM community regard the ATM layer as a data link layer, or when doing LAN emulation, even a physical layer. Many people in the Internet community also regard it as a data link layer because they want to put IP on top of it, and making the ATM layer a data link layer fits well with this idea. (Although following through with this line of reasoning, to the Internet community, *all* networks operate at the data link layer, no matter what their physical characteristics are.)

The only problem is that the ATM layer does not have the characteristics of a data link layer protocol: a single-hop protocol used by machines at the opposite ends of a wire, such as protocols 1 through 6 in Chap. 3. It has the characteristics of a network layer protocol: end-to-end virtual circuits, switching, and routing.

The author is reminded of an old riddle:

Q: How many legs would a mule have if you called the tail a leg?
A: Four. *Calling* the tail a leg does not *make* it a leg.

Suffice it to say, the reader is warned of controversy here, combined with a major dose of raw emotion.

The ATM layer is connection oriented, both in terms of the service it offers and the way it operates internally. The basic element of the ATM layer is the virtual circuit (officially called a **virtual channel**). A virtual circuit is normally a connection from one source to one destination, although multicast connections are also permitted. Virtual circuits are unidirectional, but a pair of circuits can be created at the same time. Both parts of the pair are addressed by the same identifier, so effectively a virtual circuit is full duplex. However, the channel capacity and other properties may be different in the two directions and may even be zero for one of them.

The ATM layer is unusual for a connection-oriented protocol in that it does not provide any acknowledgements. The reason for this design is that ATM was designed for use on fiber optic networks, which are highly reliable. It was thought adequate to leave error control to higher layers. After all, sending acknowledgements in the data link or network layer is really only an optimization. It is always sufficient for the transport layer to send a message and then send the entire message again if it is not acknowledged on time.

Furthermore, ATM networks are often used for real-time traffic, such as audio and video. For this kind of traffic, retransmitting an occasional bad cell is worse than just ignoring it.

Despite its lack of acknowledgements, the ATM layer does provide one hard guarantee: cells sent along a virtual circuit will never arrive out of order. The ATM subnet is permitted to discard cells if congestion occurs but under no conditions may it reorder the cells sent on a single virtual circuit. No ordering guarantees are given about cells sent on *different* virtual circuits, however. For example, if a host sends a cell on virtual circuit 10 and later sends a cell on virtual circuit 20 to the same destination, the second cell may arrive first. If the two cells had been sent on the same virtual circuit, the first one sent would always be the first one to arrive.

The ATM layer supports a two-level connection hierarchy that is visible to the transport layer. Along any transmission path from a given source to a given destination, a group of virtual circuits can be grouped together into what is called a **virtual path**, as depicted in Fig. 5-61. Conceptually, a virtual path is like a bundle of twisted copper pairs: when it is rerouted, all the pairs (virtual circuits) are rerouted together. We will consider the implications of this two-level design in detail later.

5.6.1. Cell Formats

In the ATM layer, two interfaces are distinguished: the **UNI** (**User-Network Interface**) and the **NNI** (**Network-Network Interface**). The former defines the boundary between a host and an ATM network (in many cases, between the

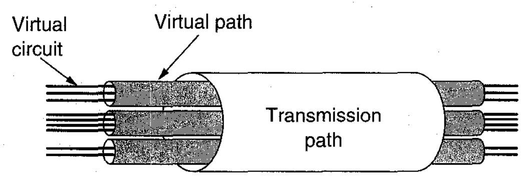

Fig. 5-61. A transmission path can hold multiple virtual paths, each of which can hold multiple virtual circuits.

customer and the carrier). The latter applies to the line between two ATM switches (the ATM term for routers).

In both cases the cells consist of a 5-byte header followed by a 48-byte payload, but the two headers are slightly different. The headers, as defined by the ATM Forum, are illustrated in Fig. 5-62. Cells are transmitted leftmost byte first and leftmost bit within a byte first.

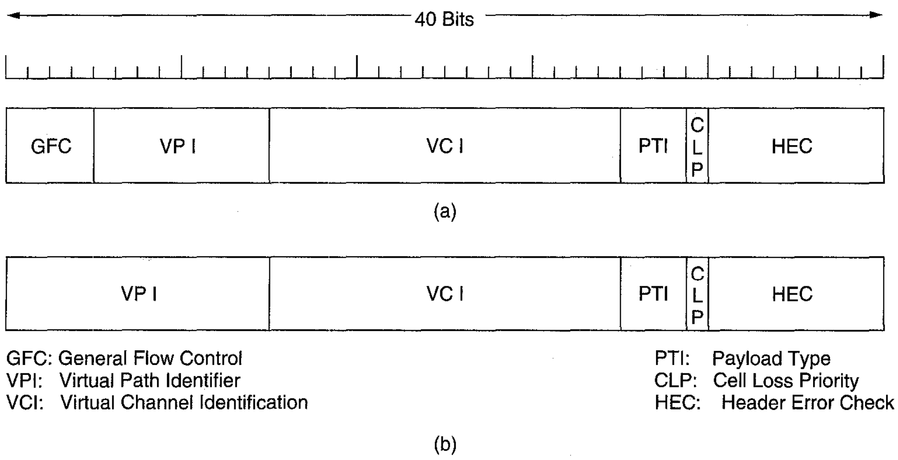

Fig. 5-62. (a) The ATM layer header at the UNI. (b) The ATM layer header at the NNI.

The *GFC* field is present only in cells between a host and the network. It is overwritten by the first switch it reaches, so it does not have end-to-end significance and is not delivered to the destination. It was originally conceived as perhaps having some utility for flow control or priority between hosts and the networks, but no values are defined for it and the network ignores it. The best way to think of it is as a bug in the standard.

The *VPI* field is a small integer selecting a particular virtual path (see Fig. 5-61). Similarly, the *VCI* field selects a particular virtual circuit within the chosen virtual path. Since the *VPI* field has 8 bits (at the UNI) and the *VCI* field has 16 bits, theoretically, a host could have up to 256 VC bundles, each containing up to 65,536 virtual circuits. Actually, slightly fewer of each are available because some *VCI*s are reserved for control functions, such as setting up virtual circuits.

The *PTI* field defines the type of payload the cell contains in accordance with the values given in Fig. 5-63. Here the cell types are user supplied, but the congestion information is network supplied. In other words, a cell sent with *PTI* 000 might arrive with 010 to warn the destination of problems underway.

Payload type	Meaning
000	User data cell, no congestion, cell type 0
001	User data cell, no congestion, cell type 1
010	User data cell, congestion experienced, cell type 0
011	User data cell, congestion experienced, cell type 1
100	Maintenance information between adjacent switches
101	Maintenance information between source and destination switches
110	Resource Management cell (used for ABR congestion control)
111	Reserved for future function

Fig. 5-63. Values of the *PTI* field.

The *CLP* bit can be set by a host to differentiate between high-priority traffic and low-priority traffic. If congestion occurs and cells must be discarded, switches first attempt to discard cells with *CLP* set to 1 before throwing out any set to 0.

Finally, the *HEC* field is a checksum over the header. It does not check the payload. A Hamming code on a 40-bit number only requires 5 bits, so with 8 bits a more sophisticated code can be used. The one chosen can correct all single-bit errors and can detect about 90 percent of all multibit errors. Various studies have shown that the vast majority of errors on optical links are single-bit errors.

Following the header comes 48 bytes of payload. Not all 48 bytes are available to the user, however, since some of the AAL protocols put their headers and trailers inside the payload.

The NNI format is the same as the UNI format, except that the *GFC* field is not present and those 4 bits are used to make the *VPI* field 12 bits instead of 8.

5.6.2. Connection Setup

ATM supports both permanent virtual circuits and switched virtual circuits. The former are always present and can be used at will, like leased lines. The latter have to be established each time they are used, like making phone calls. In this section we will describe how switched virtual circuits are established.

Technically, connection setup is not part of the ATM layer but is handled by the control plane (see Fig. 1-30) using a highly-complex ITU protocol called **Q.2931** (Stiller, 1995). Nevertheless, the logical place to handle setting up a network layer connection is in the network layer, and similar network layer protocols do connection setup here, so we will discuss it here.

Several ways are provided for setting up a connection. The normal way is to first acquire a virtual circuit for signaling and use it. To establish such a circuit, cells containing a request are sent on virtual path 0, virtual circuit 5. If successful, a new virtual circuit is opened on which connection setup requests and replies can be sent and received.

The reason for this two-step setup procedure is that this way the bandwidth reserved for virtual circuit 5 (which is barely used at all) can be kept extremely low. Also, an alternative way is provided to set up virtual circuits. Some carriers may allow users to have permanent virtual paths between predefined destinations, or allow them to set these up dynamically. Once a host has a virtual path to some other host, it can allocate virtual circuits on it itself, without the switches being involved.

Virtual circuit establishment uses the six message types listed in Fig. 5-64. Each message occupies one or more cells and contains the message type, length, and parameters. The messages can be sent by a host to the network or can be sent by the network (usually in response to a message from another host) to a host. Various other status and error reporting messages also exist but are not shown here.

Message	Meaning when sent by host	Meaning when sent by network
SETUP	Please establish a circuit	Incoming call
CALL PROCEEDING	I saw the incoming call	Your call request will be attempted
CONNECT	I accept the incoming call	Your call request was accepted
CONNECT ACK	Thanks for accepting	Thanks for making the call
RELEASE	Please terminate the call	The other side has had enough
RELEASE COMPLETE	Ack for RELEASE	Ack for RELEASE

Fig. 5-64. Messages used for connection establishment and release.

The normal procedure for establishing a call is for a host to send a SETUP message on a special virtual circuit. The network then responds with CALL PROCEEDING to acknowledge receipt of the request. As the SETUP message propagates toward the destination, it is acknowledged at each hop by CALL PROCEEDING.

When the SETUP message finally arrives, the destination host can respond with CONNECT to accept the call. The network then sends a CONNECT ACK message to indicate that it has received the CONNECT message. As the CONNECT message propagates back toward the originator, each switch receiving it acknowledges it with a CONNECT ACK message. This sequence of events is shown in Fig. 5-65(a).

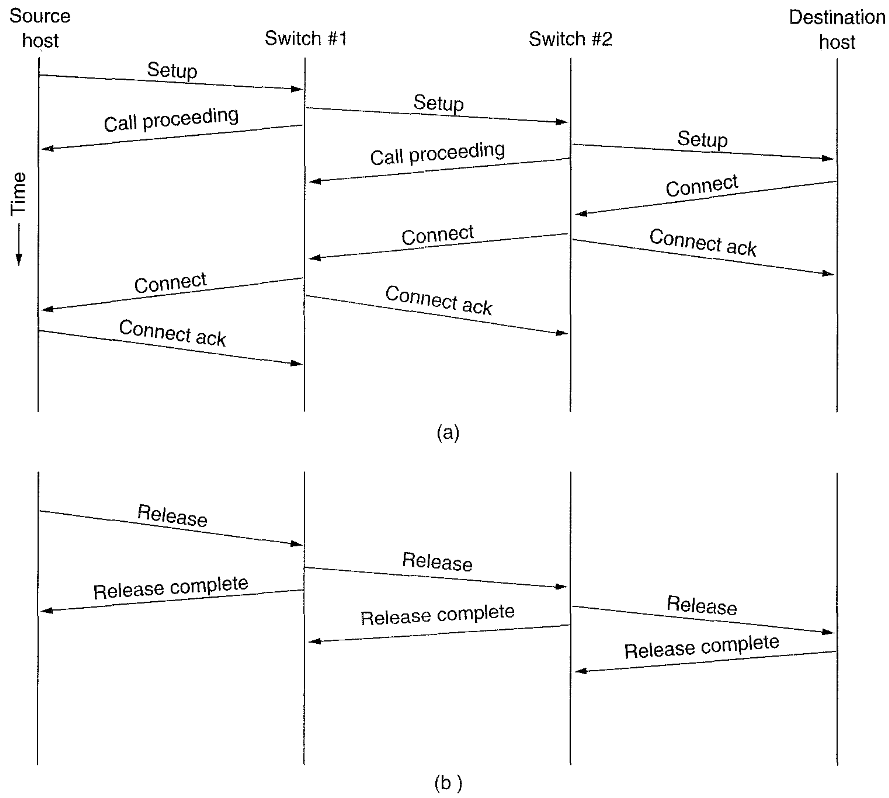

Fig. 5-65. (a) Connection setup in an ATM network. (b) Connection release.

The sequence for terminating a virtual circuit is simple. The host wishing to hang up just sends a RELEASE message, which propagates to the other end and causes the circuit to be released. Each hop along the way, the message is acknowledged, as shown in Fig. 5-65(b).

ATM networks allow multicast channels to be set up. A multicast channel has one sender and more than one receiver. These are constructed by setting up a connection to one of the destinations in the usual way. Then the ADD PARTY message is sent to attach a second destination to the virtual circuit returned by the previous call. Additional ADD PARTY messages can be sent afterward to increase the size of the multicast group.

In order to set up a connection to a destination, it is necessary to specify which destination, by including its address in the SETUP message. ATM addresses come in three forms. The first is 20 bytes long and is based on OSI addresses. The first byte indicates which of three formats the address is in. In the first format, bytes 2 and 3 specify a country, and byte 4 gives the format of the rest of the address, which contains a 3-byte authority, a 2-byte domain, a 2-byte area, and a 6-byte address, plus some other items. In the second format, bytes 2 and 3 designate an international organization instead of a country. The rest of the address is the same as in format 1. Alternatively, a older form of addressing (CCITT E.164) using 15-digit decimal ISDN telephone numbers is also permitted.

5.6.3. Routing and Switching

When a virtual circuit is set up, the SETUP message wends its way through the network from source to destination. The routing algorithm determines the path taken by this message, and thus by the virtual circuit. The ATM standard does not specify any particular routing algorithm, so the carrier is free to choose among the algorithms discussed earlier in this chapter, or to use a different one.

Experience with previous connection-oriented networks, such as X.25, have shown that a considerable amount of computing power in the switches can be expended determining how to convert the virtual circuit information in each cell to the choice of output line. The ATM designers wanted to avoid this fate, so the ATM layer has been designed to make efficient routing possible. In particular, the idea was to route on the *VPI* field, but not the *VCI* field, except at the final hop in each direction, when cells are sent between a switch and a host. Between two switches, only the virtual path was to be used.

Using only the *VPI*s between interior switches has several advantages. To start with, once a virtual path has been established from a source to a destination, any additional virtual circuits along that path can just follow the existing path. No new routing decisions have to be made. It is as though a bundle of twisted pairs has already been pulled from the source to the destination. Setting up a new connection merely requires allocating one of the unused pairs.

Second, routing of individual cells is easier when all virtual circuits for a given path are always in the same bundle. The routing decision only involves looking at a 12-bit number, not a 12-bit number and a 16-bit number. We will describe how cell switching is done below, but even without going into the details, it should be clear that indexing into a table of 2^{12} entries is feasible whereas indexing into a table of 2^{28} entries is not.

Third, basing all routing on virtual paths makes it easier to switch a whole group of virtual circuits. Consider, for example, the hypothetical U.S. ATM backbone illustrated in Fig. 5-66. Normally, virtual circuits from NY to SF pass through Omaha and Denver. However, suppose a disturbance occurs on the Omaha-Denver line. By rerouting the Omaha-Denver virtual path to LA and then SF, all the virtual circuits (potentially up to 65,535 of them) can be switched in one operation instead of potentially thousands of operations.

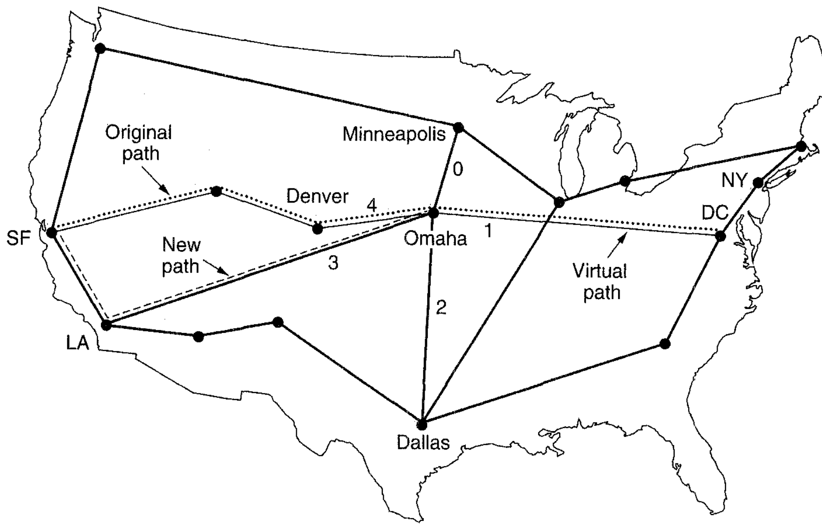

Fig. 5-66. Rerouting a virtual path reroutes all of its virtual circuits.

Finally, virtual paths make it easier for carriers to offer closed user groups (private networks) to corporate customers. A company can set up a network of permanent virtual paths among its various offices, and then allocate virtual circuits within these paths on demand. No calls can come into the private network from outside and no calls can leave the private network, except via special gateways. Many companies like this kind of security.

Whether switches will actually use the *VPI* field for routing as planned, or will, in fact, use the combination of the *VPI* and *VCI* fields (thus negating all the advantages just discussed) remains to be seen. Initial evidence from the field is not encouraging.

Let us now see how cells could be routed within an interior switch (one that is attached only to other switches and not to hosts). To make matters concrete, let us consider the Omaha switch of Fig. 5-66. For each of its five incoming lines, it has a table, *vpi_table*, indexed by incoming *VPI* that tells which of the five outgoing lines to use and what VPI to put in outgoing cells. Let us assume the five lines are numbered from 0 to 4, clockwise starting at Minneapolis. For each outgoing line, the switch maintains a bit map telling which *VPIs* are currently in use on that line.

When the switch is booted, all the entries in all the *vpi_table* structures are marked as not in use. Similarly, all the bit maps are marked to indicate that all *VPIs* are available (except the reserved ones). Now suppose calls come as shown in Fig. 5-67.

As each virtual path (and virtual circuit) is set up, entries are made in the tables. We will assume the virtual circuits are full duplex, so that each one set up

Source	Incoming line	Incoming VPI	Destination	Outgoing line	Outgoing VPI	Path:
NY	1	1	SF	4	1	New
NY	1	2	Denver	4	2	New
LA	3	1	Minneapolis	0	1	New
DC	1	3	LA	3	2	New
NY	1	1	SF	4	1	Old
SF	4	3	DC	1	4	New
DC	1	5	SF	4	4	New
NY	1	2	Denver	4	2	Old
SF	4	5	Minneapolis	0	2	New
NY	1	1	SF	4	1	Old

Fig. 5-67. Some routes through the Omaha switch of Fig. 5-66.

results in two entries, one for the forward traffic from the source and one for the reverse traffic from the destination.

The tables corresponding to the routes of Fig. 5-67 are shown in Fig. 5-68. For example, the first call generates the (4, 1) entry for *VPI* 1 in the DC table because it refers to cells coming in on line 1 with *VPI* 1 and going to SF. However, an entry is also made in the Denver table for *VPI* 1 showing that cells coming in from Denver with *VPI* 1 should go out on line 1 with *VPI* 1. These are cells traveling the other way (from SF to NY) on this virtual path. Note that in some cases two or three virtual circuits are sharing a common path. No new table entries are needed for additional virtual circuits connecting a source and destination that already have a path assigned.

Now we can explain how cells are processed inside a switch. Suppose that a cell arrives on line 1 (DC) with *VPI* 3. The switch hardware or software uses the 3 as an index into the table for line 1 and sees that the cell should go out on line 3 (LA) with *VPI* 2. It overwrites the *VPI* field with a 2 and puts the outgoing line number, 3, somewhere in the cell, for example, in the *HEC* field, since that has to be recomputed later anyway.

Now the question is how to get the cell from its current input buffer to line 3. However, this issue (routing within a switch) was discussed in detail in Chap. 2, and we saw how it was done in knockout and Batcher-banyan switches.

At this point it is straightforward to see how an entire bundle of virtual circuits can be rerouted, as is done in Fig. 5-66. By changing the entry for *VPI* 1 in the DC table from (4, 1) to (3, 3), cells from NY headed for SF will be diverted to LA. Of course, the LA switch has to be informed of this event, so the switch has

Incoming VPI	VPI_table for Minn. (Line 0) Outgoing Line	Outgoing VPI	VPI_table for DC (Line 1) Outgoing Line	Outgoing VPI	VPI_table for Dallas (Line 2) Outgoing Line	Outgoing VPI	VPI_table for LA (Line 3) Outgoing Line	Outgoing VPI	VPI_table for Denver (Line 4) Outgoing Line	Outgoing VPI
0										
1	3	1	4	1			0	1	1	1
2	4	5	4	2			1	3	1	2
3			3	2					1	4
4			4	3					1	5
5			4	4					0	2
6										
7										
8										
4095										

Fig. 5-68. The table entries for the routes of Fig. 5-67.

to generate and send a SETUP message to LA to establish the new path with *VPI* 3. Once this path has been set up, all the virtual circuits from NY to SF are now rerouted via LA, even if there are thousands of them. If virtual paths did not exist, each virtual circuit would have its own table entry and would have to be rerouted separately.

It is worth pointing out explicitly that the discussion above is about ATM in WANs. In a LAN, matters are much simpler. For example, a single virtual path can be used for all virtual circuits.

5.6.4. Service Categories

After a fair amount of trial and error, by version 4.0 of the ATM specification, it was becoming clear what kinds of traffic ATM networks were carrying and what kind of services their customers wanted. Consequently, the standard was modified to explicitly list the service categories commonly used, in order to allow equipment vendors to optimize their adaptor boards and switches for some or all of these categories. The service categories chosen as being important are listed in Fig. 5-69.

The **CBR** (**Constant Bit Rate**) class is intended to emulate a copper wire or optical fiber (only at much greater expense). Bits are put on one end and they come off the other end. No error checking, flow control, or other processing is done. Nevertheless, this class is essential to making a smooth transition between

Class	Description	Example
CBR	Constant bit rate	T1 circuit
RT-VBR	Variable bit rate: real time	Real-time videoconferencing
NRT-VBR	Variable bit rate: non-real time	Multimedia email
ABR	Available bit rate	Browsing the Web
UBR	Unspecified bit rate	Background file transfer

Fig. 5-69. The ATM service categories.

the current telephone system and future B-ISDN systems, since voice-grade PCM channels, T1 circuits, and most of the rest of the telephone system use constant-rate, synchronous bit transmission. With the CBR class, all of this traffic can be carried directly by an ATM system. CBR is also suited to all other interactive (i.e., real-time) audio and video streams.

The next class, **VBR (Variable Bit Rate)**, is divided into two subclasses, for real time and non-real time, respectively. RT-VBR is intended for services that have variable bit rates combined with stringent real-time requirements, such as interactive compressed video (e.g., videoconferencing). Due to the way MPEG and other compression schemes work, with a complete base frame followed by a series of differences between the current frame and the base frame, the transmission rate varies strongly in time (Pancha and El Zarki, 1994). Despite this variation, it is essential that the ATM network not introduce any jitter in the cell arrival pattern, as this will cause the display to appear jerky. In other words, both the average cell delay and the variation in cell delay must be tightly controlled. On the other hand, an occasional lost bit or cell here is tolerable and is best just ignored.

The other VBR subclass is for traffic where timely delivery is important but a certain amount of jitter can be tolerated by the application. For example, multimedia email is typically spooled to the receiver's local disk before being displayed, so any variation in cell delivery times will be eliminated before the email is viewed.

The **ABR (Available Bit Rate)** service category is designed for bursty traffic whose bandwidth range is known roughly. A typical example might be for use in a company that currently connects its offices by a collection of leased lines. Typically, the company has a choice of putting in enough capacity to handle the peak load, which means that some lines are idle part of the day, or putting in just enough capacity for the minimum load, which leads to congestion during the busiest part of the day.

Using ABR service avoids having to make a long term commitment to a fixed bandwidth. With ABR it is possible to say, for example, that the capacity between two points must always be 5 Mbps, but might have peaks up to 10 Mbps.

The system will then guarantee 5 Mbps all the time, and do its best to provide 10 Mbps when needed, but with no promises.

ABR is the only service category in which the network provides rate feedback to the sender, asking it to slow down when congestion occurs. Assuming that the sender complies with such requests, cell loss for ABR traffic is expected to be low. Traveling ABR is a little like flying standby: if there are seats left over (excess capacity), standby passengers are transported without delay. If there is insufficient capacity, they have to wait (unless some of the minimum bandwidth is available).

Finally, we come to **UBR** (**Unspecified Bit Rate**), which makes no promises and gives no feedback about congestion. This category is well suited to sending IP packets, since IP also makes no promises about delivery. All UBR cells are accepted, and if there is capacity left over, they will also be delivered. If congestion occurs, UBR cells will be discarded, with no feedback to the sender and no expectation that the sender slows down.

To continue our standby analogy, with UBR, all standby passengers get to board, but if halfway to the destination the pilot sees that fuel is running low, standby passengers are unceremoniously pushed through the emergency exit. To make UBR attractive, carriers are likely to make it cheaper than the other classes. For applications that have no delivery constraints and want to do their own error control and flow control anyway, UBR is a perfectly reasonably choice. File transfer, email, and USENET news are all potential candidates for UBR service because none of these applications have real-time characteristics.

The properties of the various service categories are summarized in Fig. 5-70.

Service characteristic	CBR	RT-VBR	NRT-VBR	ABR	UBR
Bandwidth guarantee	Yes	Yes	Yes	Optional	No
Suitable for real-time traffic	Yes	Yes	No	No	No
Suitable for bursty traffic	No	No	Yes	Yes	Yes
Feedback about congestion	No	No	No	Yes	No

Fig. 5-70. Characteristics of the ATM service categories.

5.6.5. Quality of Service

Quality of service is an important issue for ATM networks, in part because they are used for real-time traffic, such as audio and video. When a virtual circuit is established, both the transport layer (typically a process in the host machine, the "customer") and the ATM network layer (e.g., a network operator, the "carrier") must agree on a contract defining the service. In the case of a public network, this contract may have legal implications. For example, if the carrier agrees not to lose more than one cell per billion and it loses two cells per billion, the customer's legal staff may get all excited and start running around yelling "breach of contract."

The contract between the customer and the network has three parts:

1. The traffic to be offered.
2. The service agreed upon.
3. The compliance requirements.

It is worth noting that the contract may be different for each direction. For a video-on-demand application, the required bandwidth from the user's remote control to the video server might be 1200 bps. In the other direction it might be 5 Mbps. It should be noted that if the customer and the carrier cannot agree on terms, or the carrier is unable to provide the service desired, the virtual circuit will not be set up.

The first part of the contract is the **traffic descriptor**. It characterizes the load to be offered. The second part of the contract specifies the quality of service desired by the customer and accepted by the carrier. Both the load and the service must be formulated in terms of measurable quantities, so compliance can be objectively determined. Merely saying "moderate load" or "good service" will not do.

To make it possible to have concrete traffic contracts, the ATM standard defines a number of **QoS** (**Quality of Service**) parameters whose values the customer and carrier can negotiate. For each quality of service parameter, the worst case performance for each parameter is specified, and the carrier is required to meet or exceed it. In some cases, the parameter is a minimum; in others it is a maximum. Again here, the quality of service is specified separately for each direction. Some of the more important ones are listed in Fig. 5-71, but not all of them are applicable to all service categories.

The first three parameters specify how fast the user wants to send. **PCR** (**Peak Cell Rate**) is the maximum rate at which the sender is planning to send cells. This parameter may be lower than what the bandwidth of the line permits. If the sender is planning to push out a cell every 4 μsec, its *PCR* is 250,000 cells/sec, even though the actual cell transmission time may be 2.7 μsec.

SCR (**Sustained Cell Rate**) is the expected or required cell rate averaged over a long time interval. For CBR traffic, *SCR* will be equal to *PCR*, but for all the other service categories, it will be substantially lower. The *PCR/SCR* ratio is one measure of the burstiness of the traffic.

MCR (**Minimum Cell Rate**) is the minimum number of cells/sec that the customer considers acceptable. If the carrier is unable to guarantee to provide this much bandwidth it must reject the connection. When ABR service is requested, then the actual bandwidth used must lie between *MCR* and *PCR*, but it may vary

Parameter	Acronym	Meaning
Peak cell rate	PCR	Maximum rate at which cells will be sent
Sustained cell rate	SCR	The long-term average cell rate
Minimum cell rate	MCR	The minimum acceptable cell rate
Cell delay variation tolerance	CDVT	The maximum acceptable cell jitter
Cell loss ratio	CLR	Fraction of cells lost or delivered too late
Cell transfer delay	CTD	How long delivery takes (mean and maximum)
Cell delay variation	CDV	The variance in cell delivery times
Cell error rate	CER	Fraction of cells delivered without error
Severely-errored cell block ratio	SECBR	Fraction of blocks garbled
Cell misinsertion rate	CMR	Fraction of cells delivered to wrong destination

Fig. 5-71. Some of the quality of service parameters.

dynamically during the lifetime of the connection. If the customer and carrier agree to setting MCR to 0, then ABR service becomes similar to UBR service.

CVDT (**Cell Variation Delay Tolerance**) tells how much variation will be present in cell transmission times. It is specified independently for *PCR* and *SCR*. For a perfect source operating at *PCR*, every cell will appear *exactly* $1/PCR$ after the previous one. No cell will ever be early and no cell will ever be late, not even by a picosecond. For a real source operating at *PCR*, some variation will occur in cell transmission times. The question is: How much variation is acceptable? Can a cell be 1 nsec early? How about 30 seconds? *CDVT* controls the amount of variability acceptable using a leaky bucket algorithm to be described shortly.

The next three parameters describe characteristics of the network and are measured at the receiver. All three are negotiable. **CLR** (**Cell Loss Ratio**) is straightforward. It measures the fraction of the transmitted cells that are not delivered at all or are delivered so late as to be useless (e.g., for real-time traffic). **CTD** (**Cell Transfer Delay**) is the average transit time from source to destination. **CDV** (**Cell Delay Variation**) measures how uniformly the cells are delivered.

The model for *CDT* and *CDV* is shown in Fig. 5-72. Here we see the probability of a cell taking time t to arrive, as a function of t. For a given source, destination, and route through the intermediate switches, some minimum delay always exists due to propagation and switching time. However, not all cells make it in the minimum time; the probability density function usually has a long tail. By choosing a value of *CDT*, the customer and the carrier are, in effect, agreeing, on how late a cell can be delivered and still count as a correctly delivered cell. Normally, *CDV* will be chosen so that, α, the fraction of cells that are rejected for being too late will be on the order of 10^{-10} or less. *CDV* measures the spread in arrival times. For real-time traffic, this parameter is often more important than *CDT*.

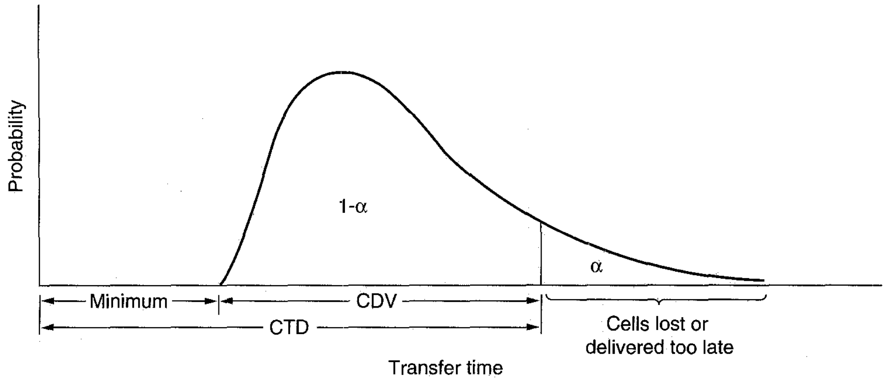

Fig. 5-72. The probability density function for cell arrival times.

The last three QoS parameters specify characteristics of the network. They are generally not negotiable. **CER (Cell Error Ratio)** is the fraction of cells that are delivered with one or more bits wrong. **SECBR (Severely-Errored Cell Block Ratio)** is the fraction of *N*-cell blocks of which *M* or more cells contain an error. Finally, **CMR (Cell Misinsertion Rate)** is the number of cells/sec that are delivered to the wrong destination on account of an undetected error in the header.

The third part of the traffic contract tells what constitutes obeying the rules. If the customer sends one cell too early, does this void the contract? If the carrier fails to meet one of its quality targets for a period of 1 msec, can the customer sue? Effectively, this part of the contract is negotiated between the parties and says how strictly the first two parts will be enforced.

The ATM and Internet quality of service models differ somewhat, which impacts their respective implementations. The ATM model is based strictly on connections, whereas the Internet model uses datagrams plus flows (e.g., RSVP). A comparison of these two models is given in (Crowcroft et al., 1995).

5.6.6. Traffic Shaping and Policing

The mechanism for using and enforcing the quality of service parameters is based (in part) on a specific algorithm, the **Generic Cell Rate Algorithm (GCRA)**, and is illustrated in Fig. 5-73. It works by checking every cell to see if it conforms to the parameters for its virtual circuit.

GCRA has two parameters. These specify the maximum allowed arrival rate (*PCR*) and the amount of variation herein that is tolerable (*CDVT*). The

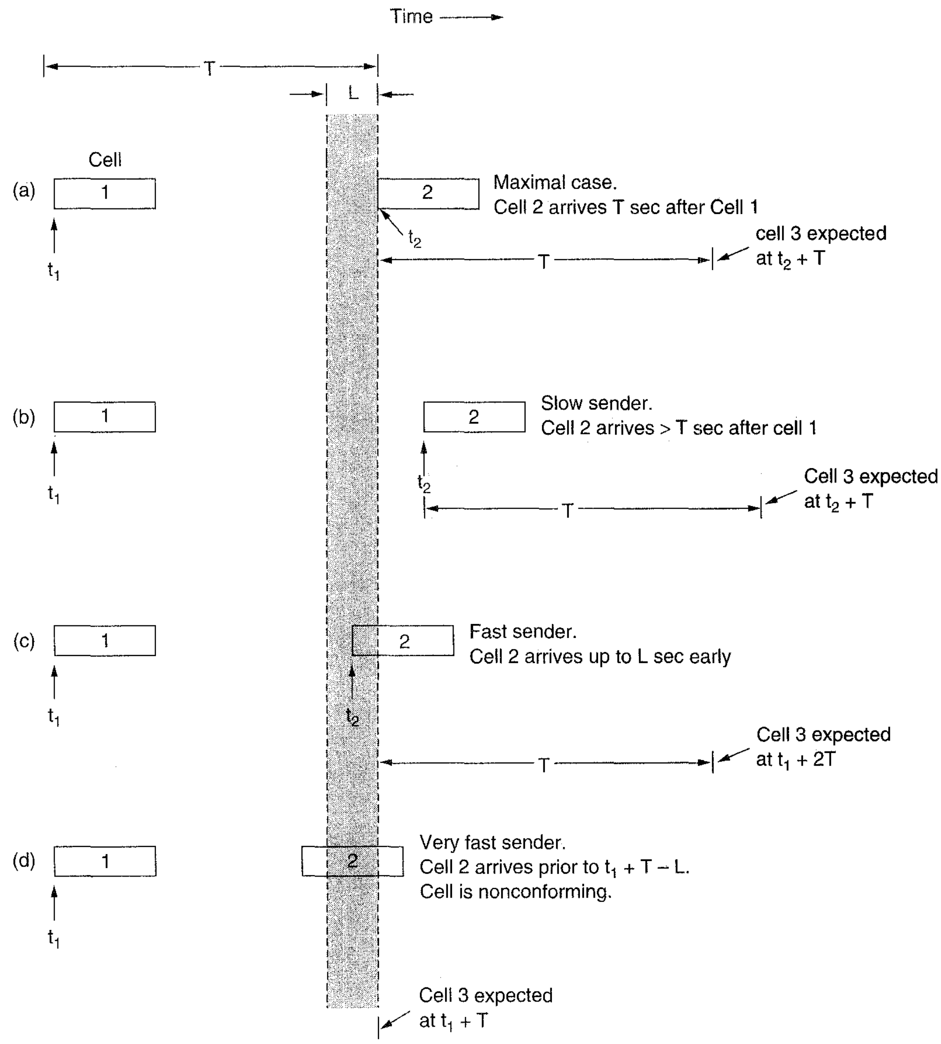

Fig. 5-73. The generic cell rate algorithm.

reciprocal of *PCR*, $T = 1/PCR$, is the minimum cell interarrival time, as shown in Fig. 5-73(a). If the customer agrees not to send more than 100,000 cells/sec, then $T = 10$ μsec. In the maximal case, one cell arrives promptly every 10 μsec. To avoid tiny numbers, we will work in microseconds, but since all the parameters are real numbers, the unit of time does not matter.

A sender is always permitted to space consecutive cells more widely than T, as shown in Fig. 5-73(b). Any cell arriving more than T μsec after the previous one is conforming.

The problem arises with senders that tend to jump the gun, as in Fig. 5-73(c) and (d). If a cell arrives a little early (at or later than $t_1 + T - L$), it is conforming, but the next cell is still expected at $t_1 + 2T$, (not at $t_2 + T$), to prevent the sender from transmitting every cell L μsec early, and thus increasing the peak cell rate.

If a cell arrives more than L μsec early, it is declared as nonconforming. The treatment of nonconforming cells is up to the carrier. Some carriers may simply discard them; others may keep them, but set the *CLP* bit, to mark them as low priority to allow switches to drop nonconforming cells first in the event of congestion. The use of the *CLP* bit may also be different for the different service categories of Fig. 5-69.

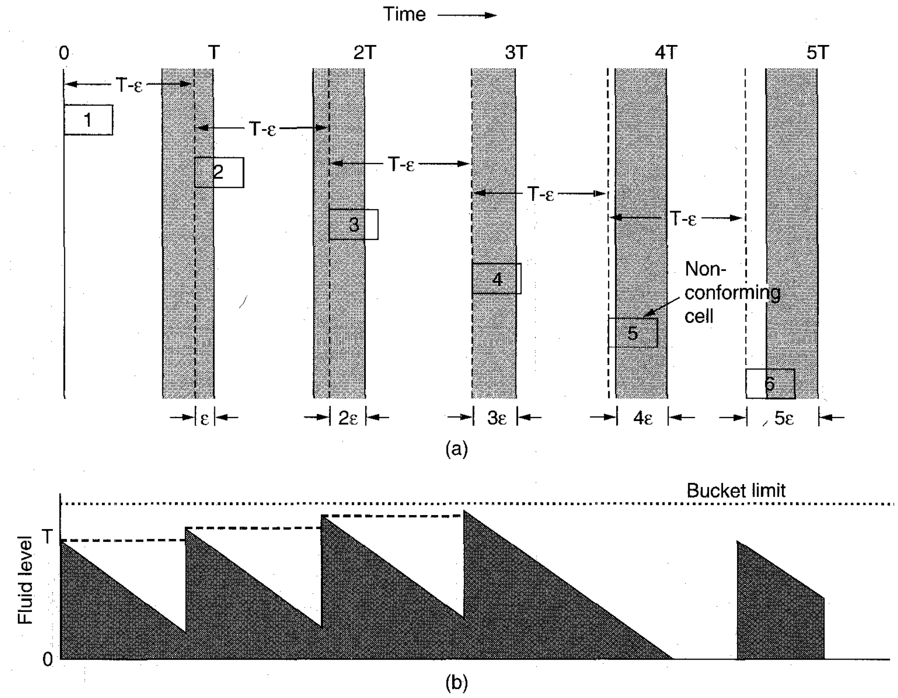

Fig. 5-74. (a) A sender trying to cheat. (b) The same cell arrival pattern, but now viewed in terms of a leaky bucket.

Now let us consider what happens if a sender tries to cheat a little bit, as shown in Fig. 5-74(a). Instead of waiting until time T to send cell 2, the sender transmits it a wee bit early, at $T - \varepsilon$, where, say, $\varepsilon = 0.3L$. This cell is accepted without problems.

Now the sender transmits cell 3, again at $T - \varepsilon$ after the previous cell, that is, at $T - 2\varepsilon$. Again it is accepted. However, every successive cell inches closer and closer to the fatal $T - L$ boundary. In this case, cell 5 arrives at $T - 4\varepsilon$ ($T - 1.2L$) which is too early, so cell 5 is declared nonconforming and is discarded by the network interface.

When viewed in these terms, the GCRA algorithm is called a **virtual scheduling algorithm**. However, viewed differently, it is equivalent to a leaky bucket algorithm, as depicted in Fig. 5-74(b). Imagine that each conforming cell that arrives pours T units of fluid into a leaky bucket. The bucket leaks fluid at a rate of 1 unit/μsec, so that after T μsec it is all gone. If cells arrive precisely every T μsec, each arriving cell will find the bucket (just) emptied, and will refill it with T units of fluid. Thus the fluid level is raised to T when a cell arrives and is reduced linearly until it gets to zero. This situation is illustrated in Fig. 5-74(b) between 0 and T.

Since fluid drains out linearly in time, at a time t after a cell arrives, the amount of its fluid left is $T - t$. At the time cell 2 arrives, at $T - \varepsilon$, there are still ε units of fluid in the bucket. The addition of the new cell raises this value to $T + \varepsilon$. Similarly, at the time cell 3 arrives, 2ε units are left in the bucket so the new cell raises the fluid level to $T + 2\varepsilon$. When cell 4 arrives, it is raised to $T + 3\varepsilon$.

If this goes on indefinitely, some cell is going to raise the level to above the bucket capacity and thus be rejected. To see which one it is, let us now compute what the bucket capacity is. We want the leaky bucket algorithm to give the same result as Fig. 5-74(a), so we want overflow to occur when a cell arrives L μsec early. If the fluid left requires L μsec to drain out, the amount of fluid must be L, since the drain rate is 1 unit/μsec. Thus we want the bucket capacity to be $L + T$ so that any cell arriving more than L μsec early will be rejected due to bucket overflow. In Fig. 5-74(b), when cell 5 arrives, the addition of T units to the 4ε units of fluid already present raises the bucket level to $T + 4\varepsilon$. Since we are using $\varepsilon = 0.3L$ in this example, the bucket would be raised to $T + 1.2L$ by the addition of cell 5, so the cell is rejected, no new fluid is added, and the bucket eventually empties.

For a given T, if we set L very small, the capacity of the bucket will be hardly more than T, so all cells will have to be sent with a very uniform spacing. However, if we now raise L to a value much greater than T, the bucket can hold multiple cells because $T + L \gg T$. This means that the sender can transmit a burst of cells back-to-back at the peak rate and have them still accepted.

We can easily compute the number of conforming cells, N, that can be transmitted back-to-back at the peak cell rate ($PCR = 1/T$). During a burst of N cells, the total amount of fluid added to the bucket is NT because each cell adds T. However, during the maximum burst, fluid drains out of the bucket at a rate of 1 unit per time interval. Let us call the cell transmission time δ time units. Note that $\delta \le T$ because it is entirely possible for a sender on a 155.52 Mbps line to agree to send no more than 100,000 cells/sec, in which case $\delta = 2.73$ μsec and $T = 10$ μsec. During the burst of N cells, the amount of drainage is $(N - 1)\delta$ because drainage does not start until the first cell has been entirely transmitted.

From these observations, we set that the net increase in fluid in the bucket during the maximum burst is $NT - (N - 1)\delta$. The bucket capacity is $T + L$. Equating these two quantities we get

$$NT - (N - 1)\delta = T + L$$

Solving this equation for N we get

$$N = 1 + \frac{L}{T - \delta}$$

However, if this number is not an integer, it must be rounded downward to an integer to prevent bucket overflow. For example, with $PCR = 100{,}000$ cells/sec, $\delta = 2.73$ μsec, and $L = 50$ μsec, seven cells may be sent back-to-back at the 155.52 Mbps rate without filling the bucket. An eighth cell would be nonconforming.

The GCRA is normally specified by giving the parameters T and L. T is just the reciprocal of *PCR*; L is *CDVT*. The GCRA is also used to make sure the mean cell rate does not exceed *SCR* for any substantial period.

In this example we assumed that cells arrive uniformly. In reality, they do not. Nevertheless, the leaky bucket algorithm can also be used here, too. At every cell arrival, a check is made to see if there is room in the bucket for an additional T units of fluid. If there is, the cell is conforming; otherwise it is not.

In addition to providing a rule about which cells are conforming and which ones are not, the GCRA also shapes the traffic to remove some of the burstiness. The smaller *CDVT* is, the greater the smoothing effect, but the greater the chance that cells will be discarded as nonconforming. Some implementations combine the GCRA leaky bucket with a token bucket, to provide additional smoothing.

5.6.7. Congestion Control

Even with traffic shaping, ATM networks do not automatically meet the performance requirements set forth in the traffic contract. For example, congestion at intermediate switches is always a potential problem, especially when over 350,000 cells/sec are pouring in on each line, and a switch can have 100 lines. Consequently, a great deal of thought has gone into the subject of performance and congestion in ATM networks. In this section, we will discuss some of the approaches used. For additional information, see (Eckberg, 1992; Eckberg et al., 1991; Hong and Suda, 1991; Jain, 1995; and Newman, 1994).

ATM networks must deal with both long-term congestion, caused by more traffic coming in than the system can handle, and short-term congestion, caused by burstiness in the traffic. As a result, several different strategies are used together. The most important of these fall into three categories:

1. Admission control.
2. Resource reservation.
3. Rate-based congestion control.

We will now discuss each of these strategies in turn.

Admission Control

In low-speed networks, it is usually adequate to wait for congestion to occur and then react to it by telling the source of the packets to slow down. In high-speed networks, this approach often works poorly, because in the interval between sending the notification and notification arriving at the source, thousands of additional packets may arrive.

Furthermore, many ATM networks have real-time traffic sources that produce data at an intrinsic rate. Telling such a source to slow down may not work (imagine a new digital telephone with a red light on it; when congestion is signaled, the red line comes on and the speaker is required to talk 25 percent slower).

Consequently, ATM networks emphasize preventing congestion from occurring in the first place. However, for CBR, VBR, and UBR traffic, no dynamic congestion control is present at all, so here an ounce of prevention is worth a pound (actually, more like a metric ton) of cure. A major tool for preventing congestion is admission control. When a host wants a new virtual circuit, it must describe the traffic to be offered and the service expected. The network can then check to see if it is possible to handle this connection without adversely affecting existing connections. Multiple potential routes may have to be examined to find one which can do the job. If no route can be located, the call is rejected.

Denying admission should be done fairly. Is it fair that one couch potato zapping through dozens of television programs can wipe out 100 busy beavers trying to read their email? If no controls are applied, a small number of high-bandwidth users can severely affect many low-bandwidth users. To prevent this, users should be divided into classes based on usage. The probability of service denial should be roughly the same for all classes (possibly by giving each class its own resource pool).

Resource Reservation

Closely related to admission control is the technique of reserving resources in advance, usually at call setup time. Since the traffic descriptor gives the peak cell rate, the network has the possibility of reserving enough bandwidth along the path

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to handle that rate. Bandwidth can be reserved by having the SETUP message earmark bandwidth along each line it traverses, making sure, of course, that the total bandwidth earmarked along a line is less than the capacity of that line. If the SETUP message hits a line that is full, it must backtrack and look for an alternative path.

The traffic descriptor can contain not only the peak bandwidth, but also the average bandwidth. If a host wants, for example, a peak bandwidth of 100,000 cells/sec, but an average bandwidth of only 20,000 cells/sec, in principle, five such circuits could be multiplexed onto the same physical trunk. The trouble is that all five connections could be idle for half an hour, then start blasting away at the peak rate, causing massive cell loss. Since VBR traffic can be statistically multiplexed, problems can occur with this service category. Possible solutions are being studied.

Rate-Based Congestion Control

With CBR and VBR traffic, it is generally not possible for the sender to slow down, even in the event of congestion, due to the inherent real-time or semi-real-time nature of the information source. With UBR, nobody cares; if there are too many cells, the extra ones are just dropped.

However, with ABR traffic, it is possible and reasonable for the network to signal one or more senders and ask them to slow down temporarily until the network can recover. It is in the interest of a sender to comply, since the network can always punish it by throwing out its (excess) cells.

How congestion should be detected, signaled, and controlled for ABR traffic was a hot topic during the development of the ATM standard, with vigorous arguments for various proposed solutions. Let us now briefly look at some of the solutions that were quickly rejected before examining the winner.

In one proposal, whenever a sender wished to send a burst of data, it first had to send a special cell reserving the necessary bandwidth. After the acknowledgement came back, the burst could begin. The advantage here is that congestion never occurs because the required bandwidth is always there when it is needed. The ATM Forum rejected this solution due to the potentially long delay before a host may begin to send.

A second proposal had switches sending back choke cells whenever congestion began to occur. Upon receipt of such a cell, a sender was expected to cut back to half its current cell transmission rate. Various schemes were proposed for getting the rate back up again later when the congestion cleared up. This scheme was rejected because choke cells might get lost in the congestion, and because the scheme seemed unfair to small users. For example, consider a switch getting 100-Mbps streams from each of five users, and one 100-kbps stream from another user. Many committee members felt it was inappropriate to tell the 100 kbps user to give up 50 kbps because he was causing too much congestion.

A third proposal used the fact that packet boundaries are marked by a bit in the last cell. The idea here was to discard cells to relieve the congestion but to do this highly selectively. The switch was to scan the incoming cell stream for the end of a packet and then throw out all the cells in the next packet. Of course, this one packet would be transmitted later, but dropping all k cells in one packet ultimately leads to one packet retransmission, which is far better than dropping k random cells, which might lead to k packet retransmissions. This scheme was rejected on fairness grounds because the next end-of-packet mark seen might not belong to the sender overloading the switch. Also the scheme did not need to be standardized. Any switch vendor is free to pick which cells to discard when congestion occurs.

After much discussion, the battle focused on two contenders, a credit-based solution (Kung and Morris, 1995) and rate-based solution (Bonomi and Fendick, 1995). The credit-based solution was essentially a dynamic sliding window protocol. It required each switch to maintain, per virtual circuit, a credit—effectively the number of buffers reserved for that circuit. As long as each transmitted cell had a buffer waiting for it, congestion could never arise.

The argument against it came from the switch vendors. They did not want to do all the accounting to keep track of the credits and did not want to reserve so many buffers in advance. The amount of overhead and waste required was thought to be too much, so ultimately, the rate-based congestion control scheme was adopted. It works like this.

The basic model is that after every k data cells, each sender transmits a special **RM** (**Resource Management**) cell. This cell travels along the same path as the data cells, but is treated specially by the switches along the way. When it gets to the destination, it is examined, updated, and sent back to the sender. The full path for RM cells is shown in Fig. 5-75.

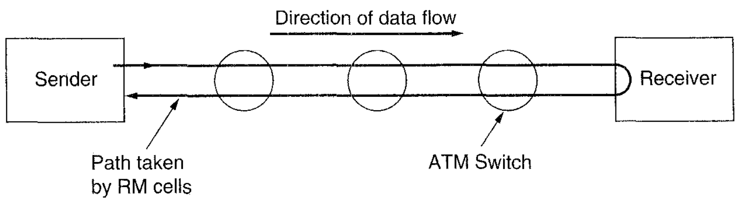


Fig. 5-75. The path taken by RM cells.

In addition, two other congestion control mechanisms are provided. First, overloaded switches can spontaneously generate RM cells and ship them back to the sender. Second, overloaded switches can set the middle *PTI* bit on data cells traveling from the sender to the receiver. Neither of these methods are fully reliable however, since these cells may be lost in the congestion without anyone noticing. In contrast, a lost RM cell will be noticed by the sender when it fails to return within the expected time interval. As an aside, the *CLP* bit is not used for ABR congestion control.

ABR congestion control is based on the idea that each sender has a current rate, **ACR** (**Actual Cell Rate**) that falls between *MCR* and *PCR*. When congestion occurs, *ACR* is reduced (but not below *MCR*). When congestion is absent, *ACR* is increased (but not above *PCR*). Each RM cell sent contains the rate at which the sender would currently like to transmit, possibly *PCR*, possibly lower. This value is called **ER** (**Explicit Rate**). As the RM cell passes through various switches on the way to the receiver, those that are congested may reduce *ER*. No switch may increase it. Reduction can occur either in the forward direction or in the reverse direction. When the sender gets the RM cell back, it can then see what the minimum acceptable rate is according to all the switches along the path. It can then adjust *ACR*, if need be, to bring it into line with what the slowest switch can handle.

The congestion mechanism using the middle *PTI* bit is integrated into the RM cells by having the receiver include this bit (taken from the last data cell) in each RM cell sent back. The bit cannot be taken from the RM cell itself because all RM cells have this bit set all the time, as shown in Fig. 5-63.

The ATM layer is quite complicated. In this chapter, we have highlighted only a portion of the issues. For additional information, see (De Prycker, 1993; McDysan and Spohn, 1995; Minoli and Vitella, 1994; and La Porta et al., 1994). However, the reader should be warned that all these references discuss the ATM 3 standard, not the ATM 4 standard, which was not finalized until 1996.

5.6.8. ATM LANs

As it becomes increasingly obvious that ITU's original goal of replacing the public switched telephone network by an ATM network is going to take a very long time, attention is shifting to the use of ATM technology to connect existing LANs together. In this approach, an ATM network can function either as a LAN, connecting individual hosts, or as a bridge, connecting multiple LANs. Although both concepts are interesting, they raise some challenging issues that we will discuss below. Additional information about ATM LANs can be found in (Chao et al., 1994; Newman, 1994; Truong et al., 1995).

The major problem that must be solved is how to provide connectionless LAN service over a connection-oriented ATM network. One possible solution is to introduce a connectionless server into the network. Every host initially sets up a connection to this server, and sends all packets to it for forwarding. While simple, this solution does not use the full bandwidth of the ATM network, and the connectionless server can easily become a bottleneck.

An alternative approach, proposed by the ATM Forum, is shown in Fig. 5-76. Here every host has a (potential) ATM virtual circuit to every other host. These virtual circuits can be established and released dynamically as needed, or they can be permanent virtual circuits. To send a frame, the source host first encapsulates the packet in the payload field of an ATM AAL message and sends it to the destination, the same way frames are shipped over Ethernets, token rings, and other LANs.

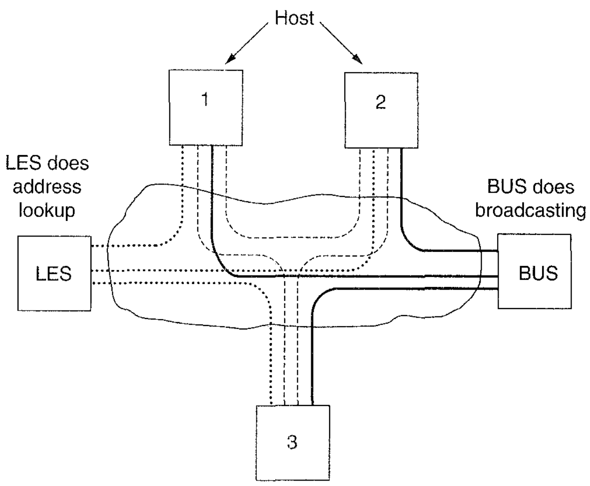


Fig. 5-76. ATM LAN emulation.

The main problem introduced by this scheme is how to tell which IP (or other network layer address) belongs to which virtual circuit. In an 802 LAN, this problem is solved by the ARP protocol, in which a host can broadcast a request such as: "Who has IP address 192.31.20.47?" The host using that address then sends back a point-to-point response, which is cached for later use.

With an ATM LAN, this solution does not work because ATM LANs do not support broadcasting. This problem is solved by introducing a new server, the **LES** (**LAN Emulation Server**). To look up a network layer address (e.g., an IP address), a host sends a packet (e.g., an ARP request) to the LES, which then looks up the corresponding ATM address and returns it to the machine requesting it. This address can then be used to send encapsulated packets to the destination.

However, this solution only solves the host location problem. Some programs use broadcasting or multicasting as an essential part of the application. For these applications, the **BUS** (**Broadcast/Unknown Server**) is introduced. It has connections to all hosts and can simulate broadcasting by sending a packet to all of them, one at a time. Hosts can also speed up delivery of a packet to an unknown host by sending the packet to the BUS for broadcasting and then (in parallel) looking up the address (for future use) using the LES.

A model similar to this one has been adopted by the IETF as the official Internet way to use an ATM network for transporting IP packets. In this model the

LES server is called the **ATMARP** server, but the functionality is essentially the same. Broadcasting and multicasting are not supported in the IETF proposal. The model is described in RFC 1483 and RFC 1577. Another good source of information is (Comer, 1995).

In the IETF method, a set of ATM hosts can be grouped together to form a **logical IP subnet**. Each LIS has its own ATMARP server. In effect, a LIS acts like a virtual LAN. Hosts on the same LIS may exchange IP packets directly, but hosts on different ones are required to go through a router. The reason for having LISes is that every host on a LIS must (potentially) have an open virtual circuit to every other host on its LIS. By restricting the number of hosts per LIS, the number of open virtual circuits can be reduced to a manageable number.

Another use of ATM networks is to use them as bridges to connect existing LANs. In this configuration, only one machine on each LAN needs an ATM connection. Like all transparent bridges, the ATM bridge must listen promiscuously to all LANs to which it is attached, forwarding frames where needed. Since bridges use only MAC addresses (not IP addresses), ATM bridges must build a spanning tree, just as 802 bridges.

In short, while ATM LAN emulation is an interesting idea, there are serious questions about its performance and price, and there is certainly heavy competition from existing LANs and bridges, which are well established and highly optimized. Whether ATM LANs and bridges ever replace 802 LANs and bridges remains to be seen.

5.7. SUMMARY

The network layer provides services to the transport layer. It can be based on either virtual circuits or datagrams. In both cases, its main job is routing packets from the source to the destination. In virtual circuit subnets, a routing decision is made when the virtual circuit is set up. In datagram subnets, it is made on every packet.

Many routing algorithms are used in computer networks. Static algorithms include shortest path routing, flooding, and flow-based routing. Dynamic algorithms include distance vector routing and link state routing. Most actual networks use one of these. Other important routing topics are hierarchical routing, routing for mobile hosts, broadcast routing, and multicast routing.

Subnets can become congested, increasing the delay and lowering the throughput for packets. Network designers attempt to avoid congestion by proper design. Techniques include traffic shaping, flow specifications, and bandwidth reservation. If congestion does occur, it must be dealt with. Choke packets can be sent back, load can be shed, and other methods applied.

Networks differ in various ways, so when multiple networks are connected together problems can occur. Sometimes the problems can be finessed by tunneling a packet through a hostile network, but if the source and destination networks are different, this approach fails. When different networks have different maximum packet sizes, fragmentation may be called for.

The Internet has a rich variety of protocols related to the network layer. These include the data transport protocol, IP, but also the control protocols ICMP, ARP, and RARP, and the routing protocols OSPF and BGP. The Internet is rapidly running out of IP addresses, so a new version of IP, IPv6, has been developed.

Unlike the datagram-based Internet, ATM networks use virtual circuits inside. These must be set up before data can be transferred and torn down after transmission is completed. Quality of service and congestion control are major issues with ATM networks.

PROBLEMS

1. Give two example applications for which connection-oriented service is appropriate. Now give two examples for which connectionless service is best.

2. Are there any circumstances when a virtual circuit service will (or at least should) deliver packets out of order? Explain.

3. Datagram subnets route each packet as a separate unit, independent of all others. Virtual circuit subnets do not have to do this, since each data packet follows a predetermined route. Does this observation mean that virtual circuit subnets do not need the capability to route isolated packets from an arbitrary source to an arbitrary destination? Explain your answer.

4. Give three examples of protocol parameters that might be negotiated when a connection is set up.

5. Consider the following design problem concerning implementation of virtual circuit service. If virtual circuits are used internal to the subnet, each data packet must have a 3-byte header, and each router must tie up 8 bytes of storage for circuit identification. If datagrams are used internally, 15-byte headers are needed, but no router table space is required. Transmission capacity costs 1 cent per 10^6 bytes, per hop. Router memory can be purchased for 1 cent per byte and is depreciated over two years (business hours only). The statistically average session runs for 1000 sec, in which time 200 packets are transmitted. The mean packet requires four hops. Which implementation is cheaper, and by how much?

6. Assuming that all routers and hosts are working properly and that all software in both is free of all errors, is there any chance, however small, that a packet will be delivered to the wrong destination?

7. Give a simple heuristic for finding two paths through a network from a given source to a given destination that can survive the loss of any communication line (assuming two such paths exist). The routers are considered reliable enough, so it is not necessary to worry about the possibility of router crashes.

8. Consider the subnet of Fig. 5-15(a). Distance vector routing is used, and the following vectors have just come in to router *C*: from *B*: (5, 0, 8, 12, 6, 2); from *D*: (16, 12, 6, 0, 9, 10); and from *E*: (7, 6, 3, 9, 0, 4). The measured delays to *B*, *D*, and *E*, are 6, 3, and 5, respectively. What is *C*'s new routing table? Give both the outgoing line to use and the expected delay.

9. If delays are recorded as 8-bit numbers in a 50-router network, and delay vectors are exchanged twice a second, how much bandwidth per (full-duplex) line is chewed up by the distributed routing algorithm? Assume that each router has three lines to other routers.

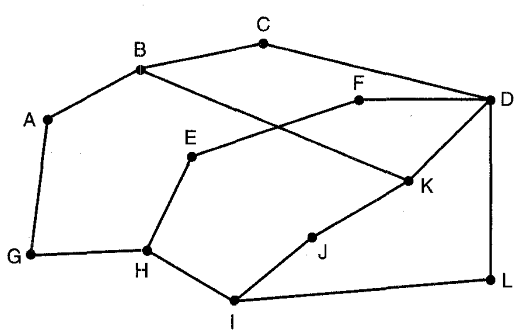
10. In Fig. 5-16 the Boolean OR of the two sets of ACF bits are 111 in every row. Is this just an accident here, or does it hold for all subnets under all circumstances?

11. For hierarchical routing with 4800 routers, what region and cluster sizes should be chosen to minimize the size of the routing table for a three-layer hierarchy?

12. In the text it was stated that when a mobile host is not at home, packets sent to its home LAN are intercepted by its home agent. For an IP network on an 802.3 LAN, how does the home agent accomplish this interception?

13. Looking at the subnet of Fig. 5-5, how many packets are generated by a broadcast from *B*, using
(a) reverse path forwarding?
(b) the sink tree?

14. Compute a multicast spanning tree for router *C* in the subnet below for a group with members at routers *A*, *B*, *C*, *D*, *E*, *F*, *I*, and *K*.



15. As a possible congestion control mechanism in a subnet using virtual circuits internally, a router could refrain from acknowledging a received packet until (1) it knows its last transmission along the virtual circuit was received successfully and (2) it has a free buffer. For simplicity, assume that the routers use a stop-and-wait protocol and that each virtual circuit has one buffer dedicated to it for each direction of traffic. If it

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takes T sec to transmit a packet (data or acknowledgement) and there are n routers on the path, what is the rate at which packets are delivered to the destination host? Assume that transmission errors are rare, and that the host-router connection is infinitely fast.

16. A datagram subnet allows routers to drop packets whenever they need to. The probability of a router discarding a packet is p. Consider the case of a source host connected to the source router, which is connected to the destination router, and then to the destination host. If either of the routers discards a packet, the source host eventually times out and tries again. If both host-router and router-router lines are counted as hops, what is the mean number of
(a) hops a packet makes per transmission?
(b) transmissions a packet makes?
(c) hops required per received packet?

17. Give an argument why the leaky bucket algorithm should allow just one packet per tick, independent of how large the packet is.

18. The byte-counting variant of the leaky bucket algorithm is used in a particular system. The rule is that one 1024-byte packet, two 512-byte packets, etc. may be sent on each tick. Give a serious restriction of this system that was not mentioned in the text.

19. An ATM network uses a token bucket scheme for traffic shaping. A new token is put into the bucket every 5 μsec. What is the maximum sustainable net data rate (i.e., excluding header bits)?

20. A computer on a 6-Mbps network is regulated by a token bucket. The token bucket is filled at a rate of 1 Mbps. It is initially filled to capacity with 8 megabits. How long can the computer transmit at the full 6 Mbps?

21. Figure 5-27 shows four input characteristics for a proposed flow specification. Imagine that the maximum packet size is 1000 bytes, the token bucket rate is 10 million bytes/sec, the token bucket size is 1 million bytes, and the maximum transmission rate is 50 million bytes/sec. How long can a burst at maximum speed last?

22. A device accepts frames from the Ethernet to which it is attached. It removes the packet inside each frame, adds framing information around it, and transmits it over a leased telephone line (its only connection to the outside world) to an identical device at the other end. This device removes the framing, inserts the packet into a token ring frame, and transmits it to a local host over a token ring LAN. What would you call the device?

23. Is fragmentation needed in concatenated virtual circuit internets, or only in datagram systems?

24. Tunneling through a concatenated virtual circuit subnet is straightforward: the multiprotocol router at one end just sets up a virtual circuit to the other end and passes packets through it. Can tunneling also be used in datagram subnets? If so, how?

25. An IP datagram using the *Strict source routing* option has to be fragmented. Do you think the option is copied into each fragment, or is it sufficient to just put it in the first fragment? Explain your answer.

26. Suppose that instead of using 16 bits for the network part of a class B address, 20 bits had been used. How many class B networks would there have been?

27. Convert the IP address whose hexadecimal representation is C22F1582 to dotted decimal notation.

28. A class B network on the Internet has a subnet mask of 255.255.240.0. What is the maximum number of hosts per subnet?

29. You have just explained the ARP protocol to a friend. When you are all done, he says: "I've got it. ARP provides a service to the network layer, so it is part of the data link layer." What do you say to him?

30. ARP and RARP both map addresses from one space to another. In this respect, they are similar. However, their implementations are fundamentally different. In what major way do they differ?

31. Describe a way to do reassembly of IP fragments at the destination.

32. Most IP datagram reassembly algorithms have a timer to avoid having a lost fragment tie up reassembly buffers forever. Suppose a datagram is fragmented into four fragments. The first three fragments arrrive, but the last one is delayed. Eventually the timer goes off and the three fragments in the receiver's memory are discarded. A little later, the last fragment stumbles in. What should be done with it?

33. Most IP routing protocols use number of hops as the metric to be minimized when doing routing computations. For ATM networks, number of hops is not terribly important. Why not? *Hint*: Take a look at Chap. 2. to see how ATM switches work. Do they use store-and-forward?

34. In both IP and ATM, the checksum covers only the header and not the data. Why do you suppose this design was chosen?

35. A person who lives in Boston travels to Minneapolis, taking her portable computer with her. To her surprise, the LAN at her destination in Minneapolis is a wireless IP LAN, so she does not have to plug in. Is it still necessary to go through the entire business with home agents and foreign agents to make email and other traffic arrive correctly?

36. IPv6 uses 16-byte addresses. If a block of 1 million addresses is allocated every picosecond, how long will the addresses last?

37. The *Protocol* field used in the IPv4 header is not present in the fixed IPv6 header. Why not?

38. When the IPv6 protocol is introduced, does the ARP protocol have to be changed? If so, are the changes conceptual or technical?

39. In Chap. 1, we classified interactions between the network and the hosts using four classes of primitives: *request*, *indication*, *response*, and *confirm*. Classify the SETUP and CONNECT messages of Fig. 5-65 into these categories.

40. A new virtual circuit is being set up in an ATM network. Between the source and destination hosts lie three ATM switches. How many messages (including acknowledgements) will be sent to establish the circuit?

41. The logic used to construct the table of Fig. 5-67 is simple: the lowest unused *VPI* is always assigned to a connection. If a new virtual circuit is requested between NY and Denver, which *VPI* will be assigned to it?

42. In Fig. 5-73(c), if a cell arrives early, the next one is still due at $t_1 + 2T$. Suppose that the rule were different, namely that the next cell was expected at $t_2 + T$, and the sender made maximum use of this rule. What maximum peak cell rate could then be achieved? For $T = 10$ μsec and $L = 2$ μsec, give the original and new peak cell rates, respectively.

43. What is the maximum burst length on an 155.52 Mbps ATM ABR connection whose *PCR* value is 200,000 and whose L value is 25 μsec?

44. Write a program to simulate routing using flooding. Each packet should contain a counter that is decremented on each hop. When the counter gets to zero, the packet is discarded. Time is discrete, with each line handling one packet per time interval. Make three versions of the program: all lines are flooded, all lines except the input line are flooded, and only the (statically chosen) best k lines are flooded. Compare flooding with deterministic routing ($k = 1$) in terms of delay and bandwidth used.

45. Write a program that simulates a computer network using discrete time. The first packet on each router queue makes one hop per time interval. Each router has only a finite number of buffers. If a packet arrives and there is no room for it, it is discarded and not retransmitted. Instead, there is an end-to-end protocol, complete with timeouts and acknowledgement packets, that eventually regenerates the packet from the source router. Plot the throughput of the network as a function of the end-to-end timeout interval, parametrized by error rate.

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