Computer Networks

Third Edition

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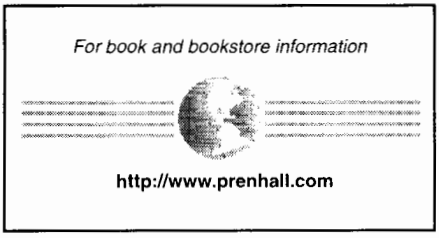

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To Suzanne, Barbara, Marvin, and Little Bram

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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.

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.

ANDREW S. TANENBAUM

1

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

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.

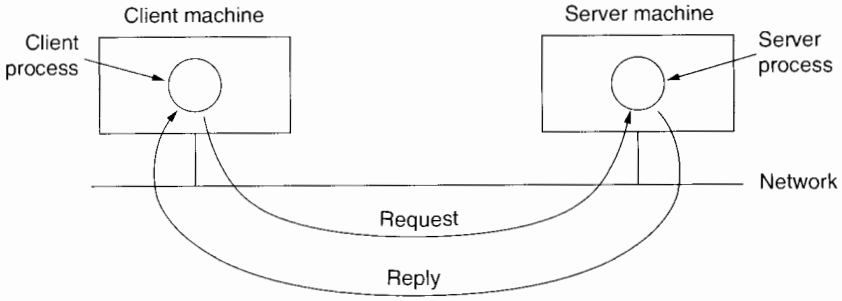

Fig. 1-1. The client-server model.

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	Local area network
100 m	Building	
1 km	Campus	
10 km	City	Metropolitan area network
100 km	Country	Wide area network
1,000 km	Continent	
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.

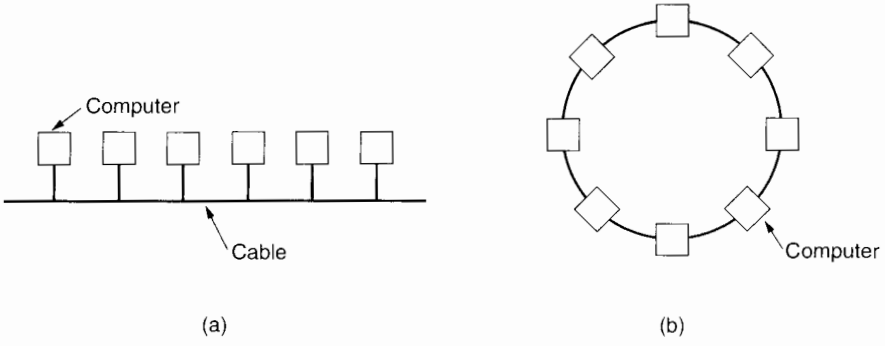

Fig. 1-3. Two broadcast networks. (a) Bus. (b) Ring.

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.

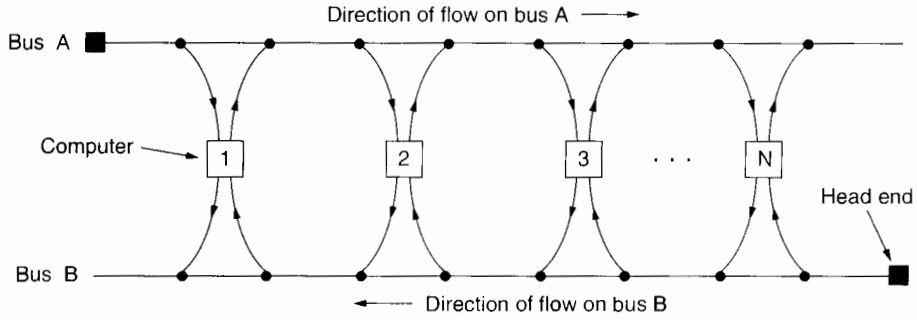

Fig. 1-4. Architecture of the DQDB metropolitan area network.

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.

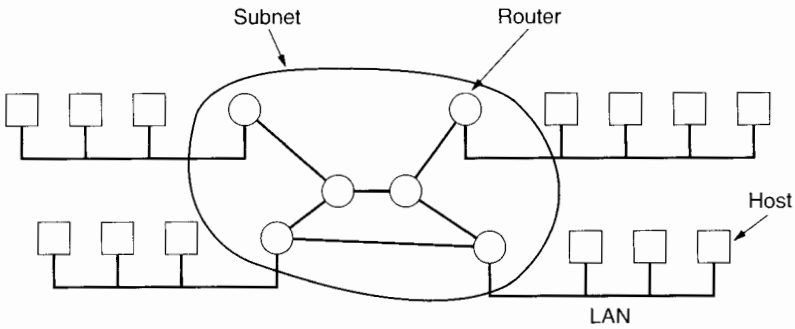

Fig. 1-5. Relation between hosts and the subnet.

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.

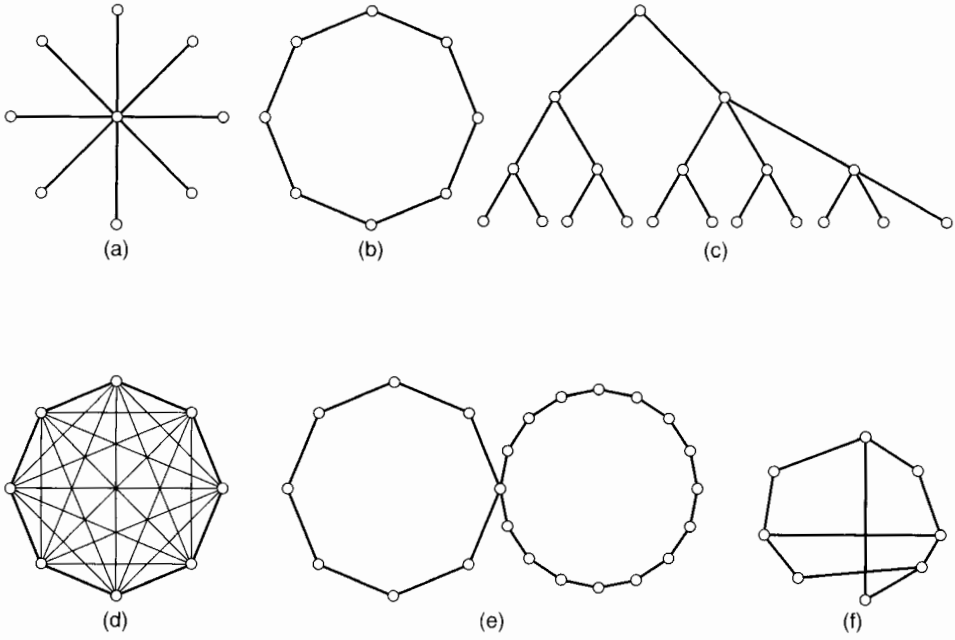

Fig. 1-6. Some possible topologies for a point-to-point subnet. (a) Star. (b) Ring. (c) Tree. (d) Complete. (e) Intersecting rings. (f) Irregular.

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

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

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.

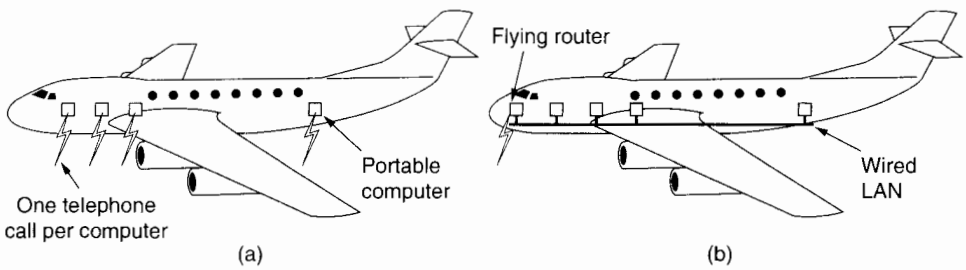

Fig. 1-8. (a) Individual mobile computers. (b) A flying LAN.

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.

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.

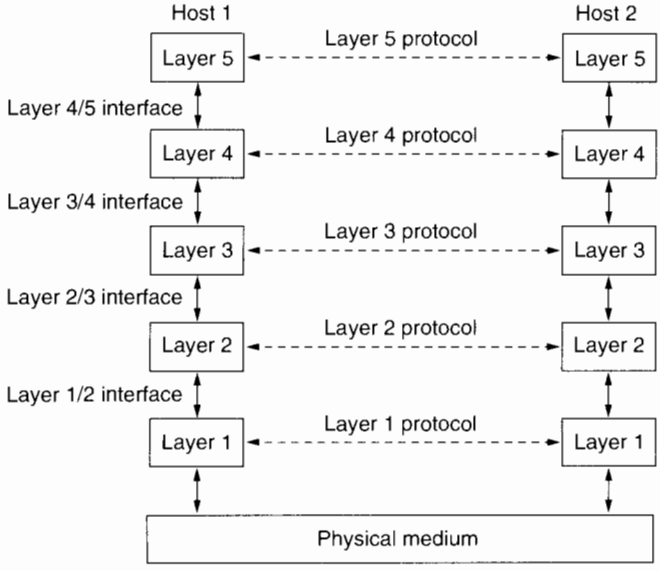

Fig. 1-9. Layers, protocols, and interfaces.

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

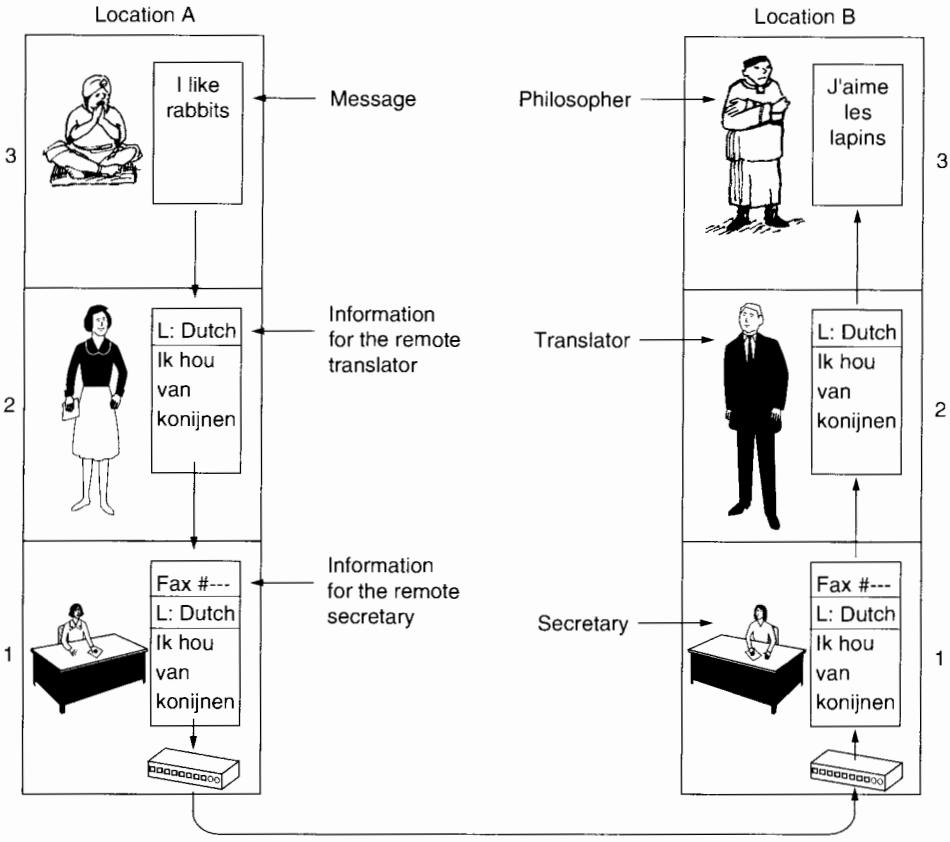

Fig. 1-10. The philosopher-translator-secretary architecture.

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

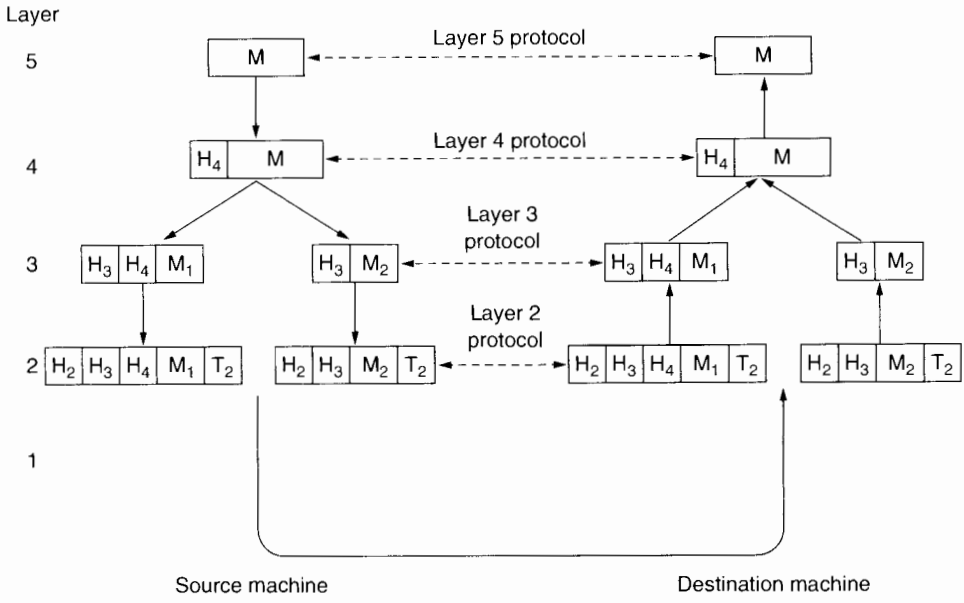

Fig. 1-11. Example information flow supporting virtual communication in layer 5.

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.

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

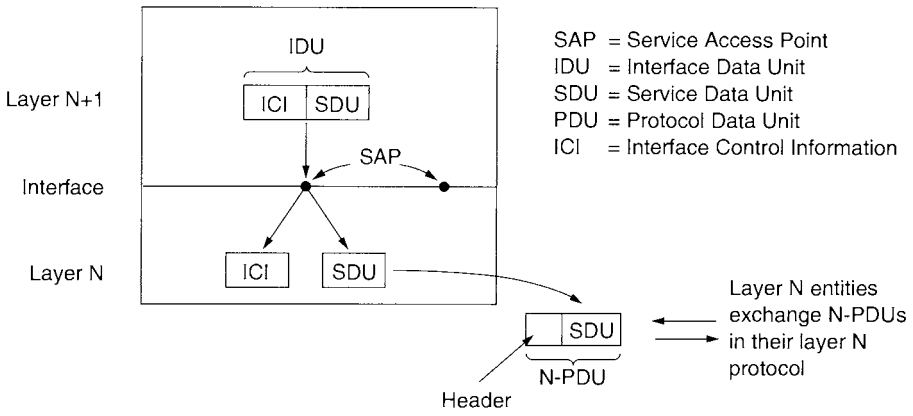

Fig. 1-12. Relation between layers at an interface.

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
Connectionless	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

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.

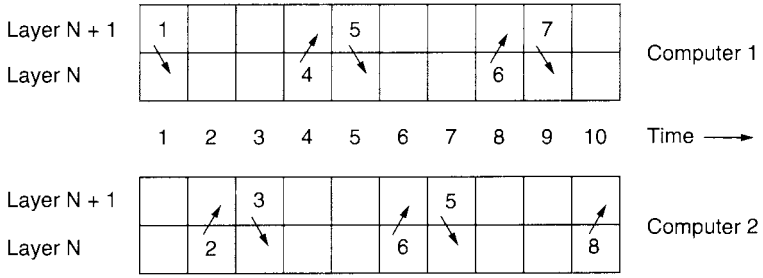

Fig. 1-15. How a computer would invite its Aunt Millie to tea. The numbers near the tail end of each arrow refer to the eight service primitives discussed in this section.

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

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.

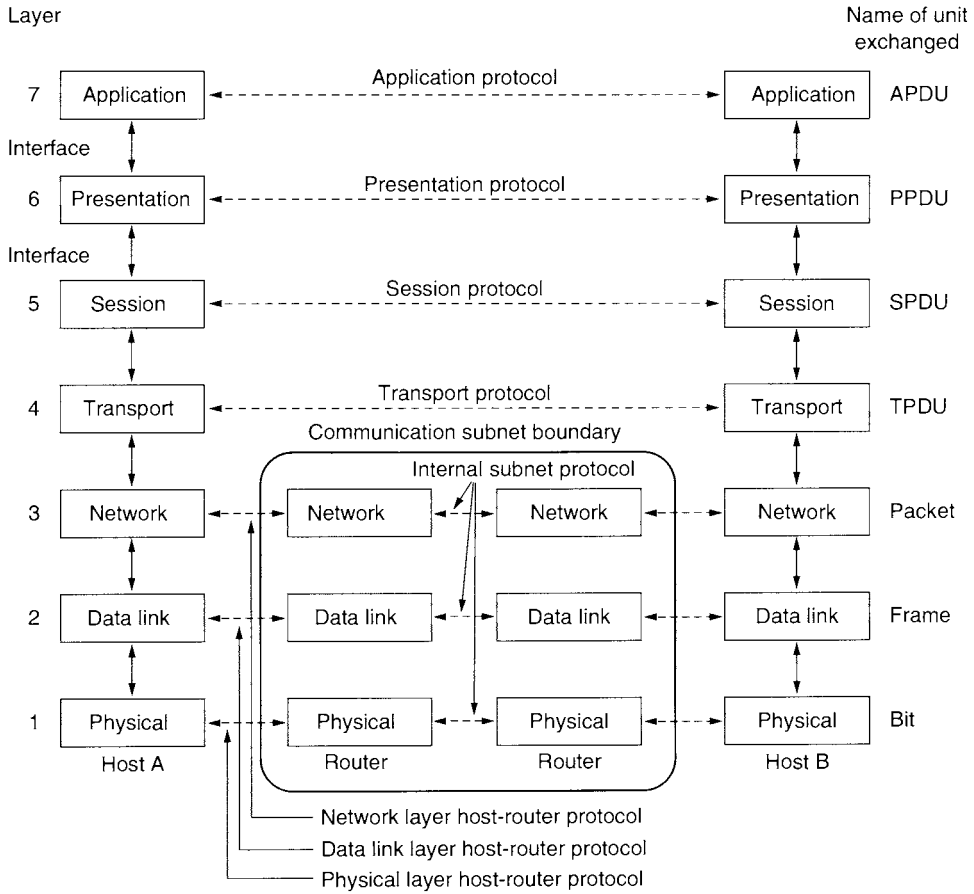

Fig. 1-16. The OSI reference model.

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.

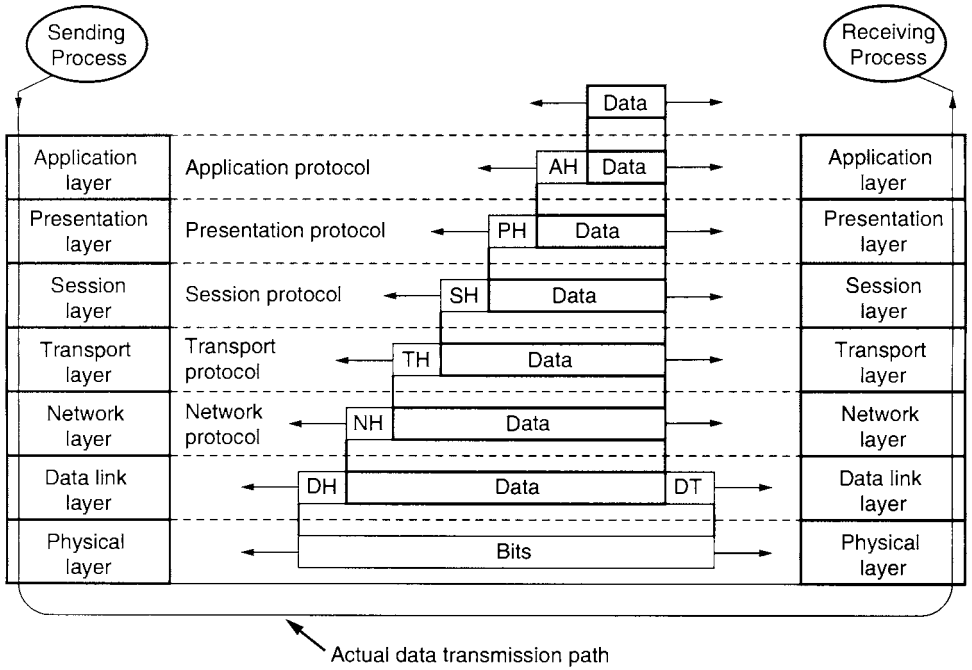

Fig. 1-17. An example of how the OSI model is used. Some of the headers may be null. (Source: H.C. Folts. Used with permission.)

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.

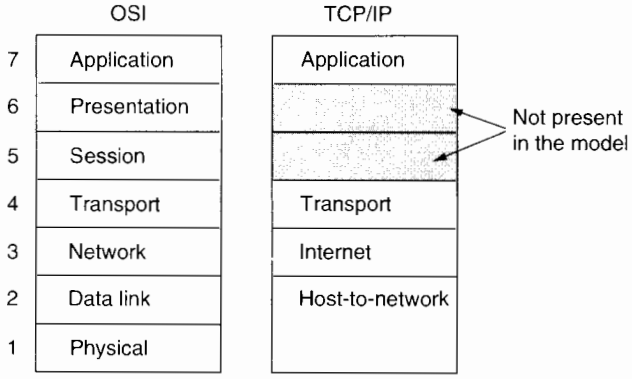

Fig. 1-18. The TCP/IP reference model.

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.

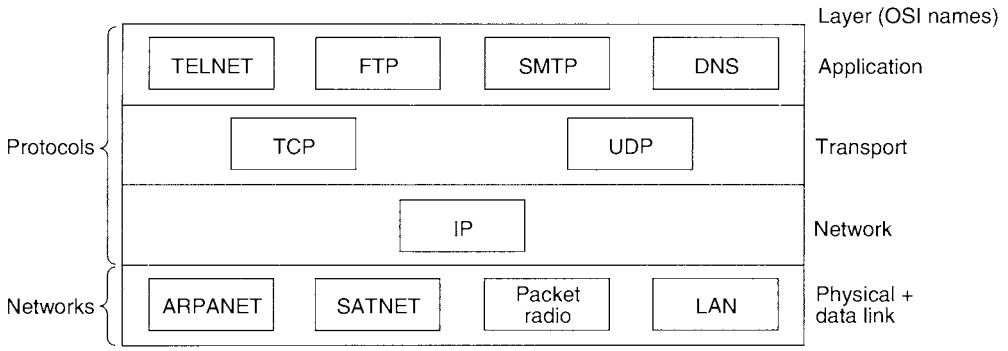

Fig. 1-19. Protocols and networks in the TCP/IP model initially.

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 Web, and many others.

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.

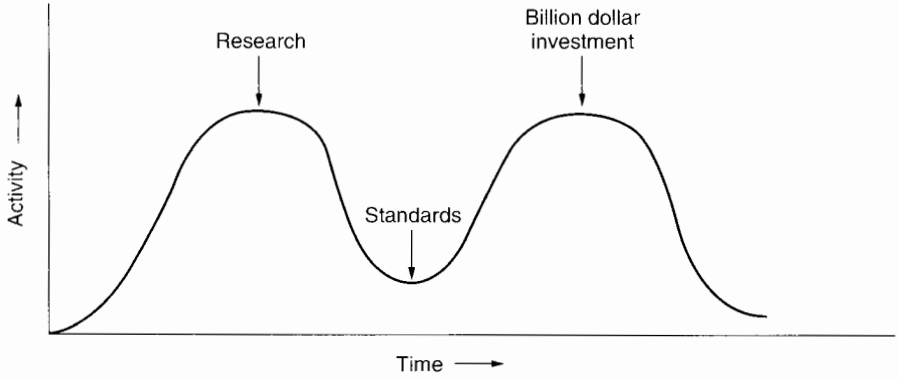

Fig. 1-20. The apocalypse of the two elephants.

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.

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.

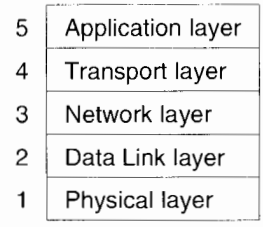

Fig. 1-21. The hybrid reference model to be used in this book.

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

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.

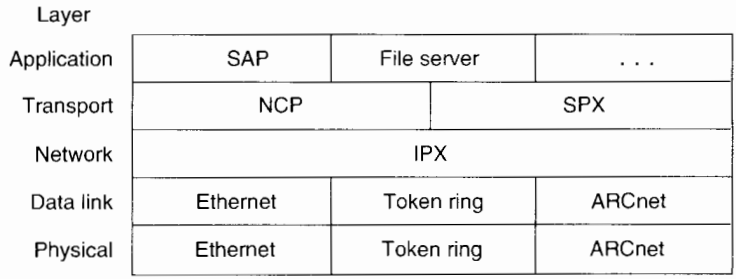

Fig. 1-22. The Novell NetWare reference model.

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

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.

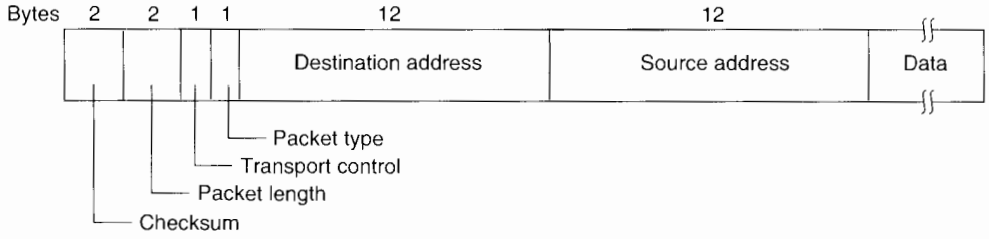

Fig. 1-23. A Novell NetWare IPX packet.

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.

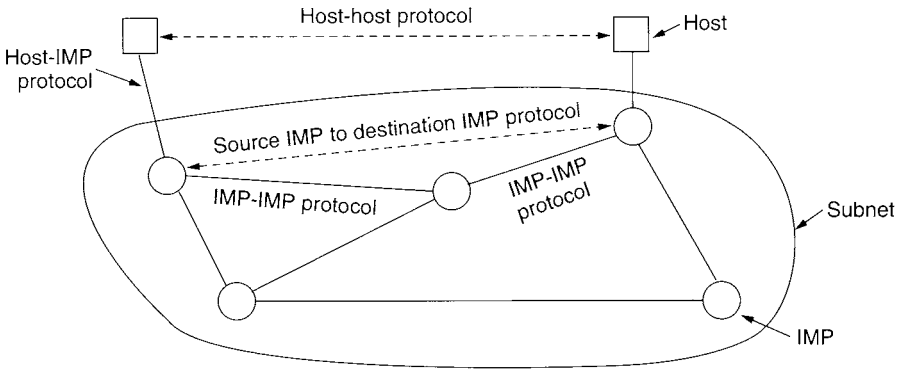

Fig. 1-24. The original ARPANET design.

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),

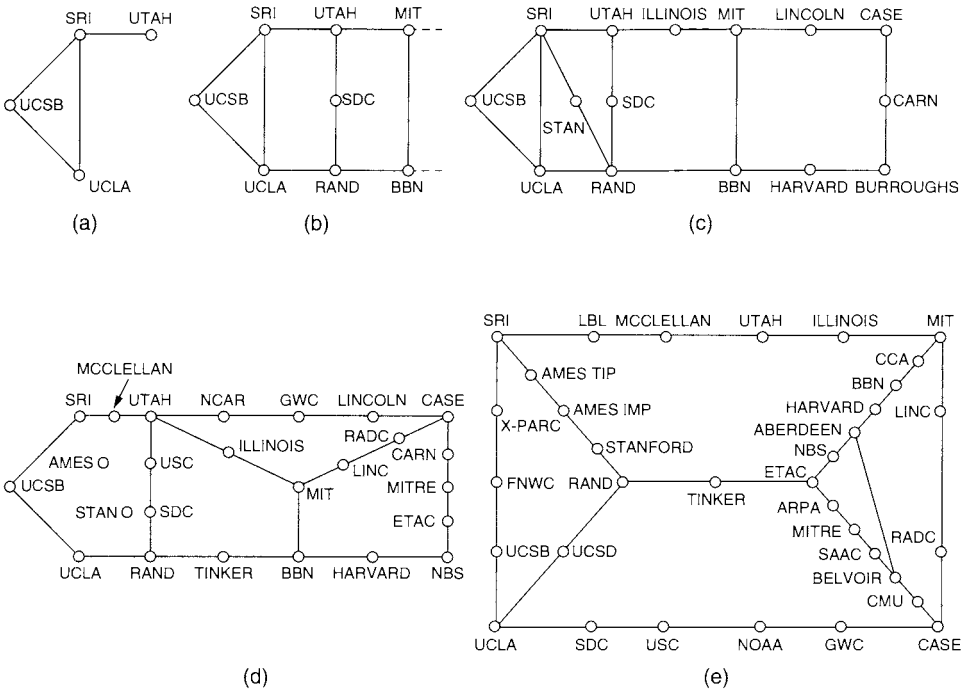

Fig. 1-25. Growth of the ARPANET. (a) Dec. 1969. (b) July 1970. (c) March 1971. (d) April 1972. (e) Sept. 1972.

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

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.

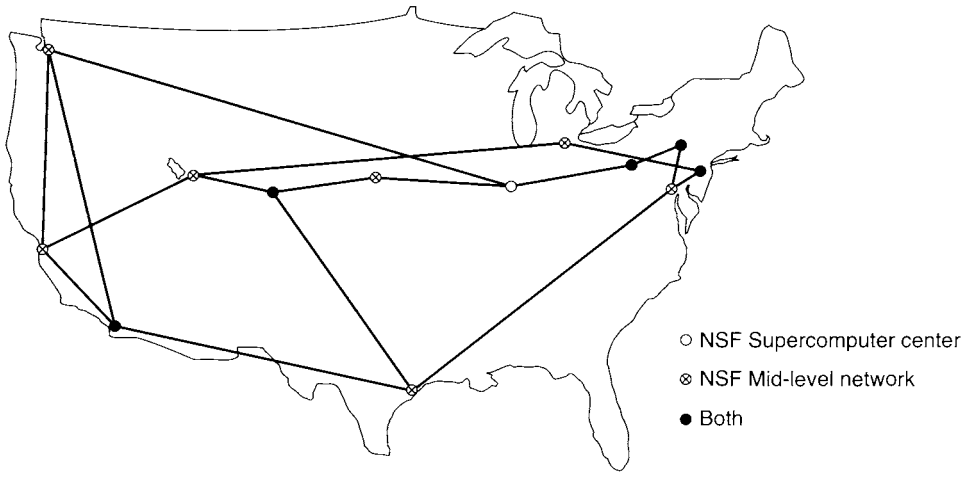

Fig. 1-26. The NSFNET backbone in 1988.

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 PC-RTs 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, EuropaNET is an IP backbone for research organizations and EBONE 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.

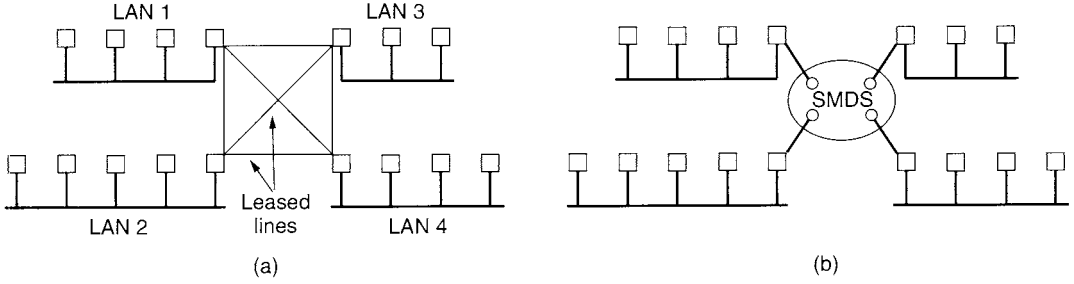

Fig. 1-27. (a) Four LANs interconnected with leased lines. (b) Interconnection using SMDS.

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.

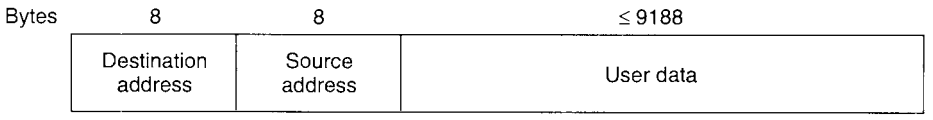

Fig. 1-28. The SMDS packet format.

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.

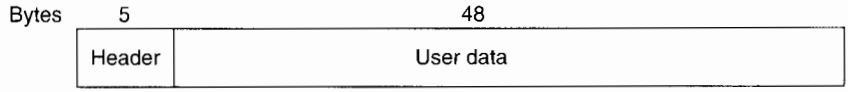

Fig. 1-29. An ATM cell.

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.

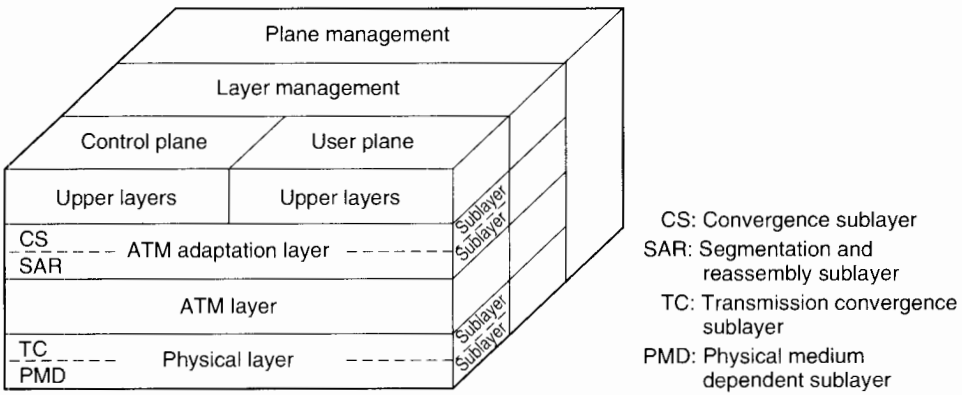

Fig. 1-30. The B-ISDN ATM reference model.

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

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

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.

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.