1.Introduction to data communications

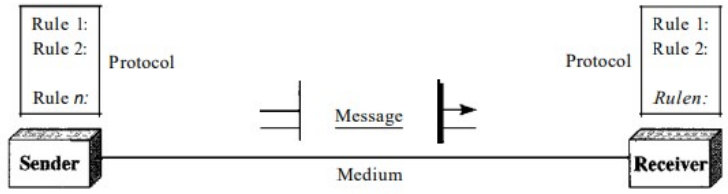
Data communication:

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

Components:

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

Figure 1.1 *Five components ofdata communication*



1. Message: The message is the information (data) to be communicated. Popular forms of information include text, numbers, pictures, audio, and video.

2. Sender: The sender is the device that sends the data message. It can be a computer, workstation, telephone handset, video camera, and so on.

3. Receiver: The receiver is the device that receives the message. It can be a computer, workstation, telephone handset, television, and so on.

4. Transmission medium: The transmission medium is the physical path by which a message travels from sender to receiver. Some examples of transmission media include twisted-pair wire, coaxial cable, fiber-optic cable, and radio waves

5. Protocol: A protocol is a set of rules that govern data communications. It represents an agreement between the communicating devices. Without a protocol, two devices may be connected but not communicating, just as a person speaking French cannot be understood by a person who speaks only Japanese.

Data representation:

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

Text:

In data communications, text is represented as a bit pattern, a sequence of bits (Os or Is). Different sets of bit patterns have been designed to represent text symbols.

Each set is called a code, and the process of representing symbols is called coding. Today, the prevalent coding system is called Unicode, which uses 32 bits to represent a symbol or character used in any language in the world. The American Standard Code for Information Interchange (ASCII), developed some decades ago in the United States, now constitutes the first 127 characters in Unicode and is also referred to as Basic Latin. Appendix A includes part of the Unicode.

Numbers:

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

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

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

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

There are several methods to represent color images. One method is called RGB, so called because each color is made of a combination of three primary colors: red, green, and blue. The intensity of each color is measured, and a bit pattern is assigned to it. Another method is called YCM, in which a color is made of a combination of three other primary colors: yellow, cyan, and magenta.

Audio:

Audio refers to the recording or broadcasting of sound or music. Audio is by nature different from text, numbers, or images. It is continuous, not discrete. Even when we use a microphone to change voice or music to an electric signal, we create a continuous signal. In Chapters 4 and 5, we learn how to change sound or music to a digital or an analog signal.

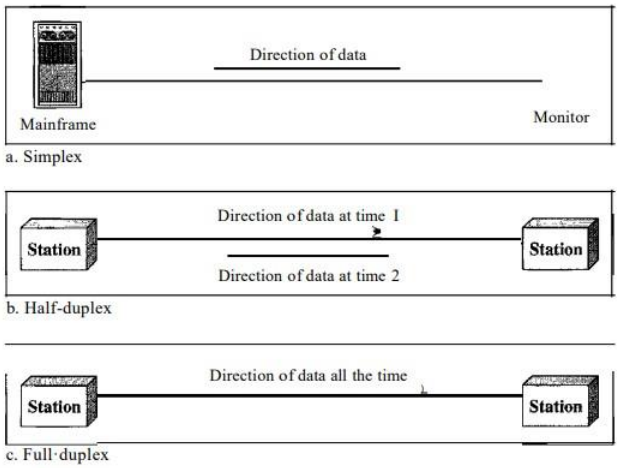
Video:

Video refers to the recording or broadcasting of a picture or movie. Video can either be produced as a continuous entity (e.g., by a TV camera), or it can be a combination of images, each a discrete entity, arranged to convey the idea of motion. Again we can change video to a digital or an analog signal, as we will see in Chapters 4 and 5

Data flow:

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

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



Simplex

In simplex mode, the communication is unidirectional, as on a one-way street. Only one of the two devices on a link can transmit; the other can only receive (see Figure 1.2a). Keyboards and traditional monitors are examples of simplex devices. The keyboard can only introduce input; the monitor can only accept output. The simplex mode can use the entire capacity of the channel to send data in one direction.

Half-Duplex

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

The half-duplex mode is like a one-lane road with traffic allowed in both directions. When cars are traveling in one direction, cars going the other way must wait. In a half-duplex transmission, the entire capacity of a channel is taken over by whichever of the two devices is transmitting at the time. Walkie-talkies and CB (citizens band) radios are both half-duplex systems. The half-duplex mode is used in cases where there is no need for communication in both directions at the same time; the entire capacity of the channel can be utilized for each direction.

Full-Duplex

In full-duplex mode (also called duplex), both stations can transmit and receive simultaneously (see Figure 1.2c)The full-duplex mode is like a two-wayy street with traffic flowing in both directions at the same time. In full-duplex mode, signals going in one direction share the capacity of the link: with signals going in the other direction. This sharing can occur in two ways: Either the link must contain two physically separate transmission paths, one for sending and the other for receiving; or the capacity of the channel is divided between signals traveling in both directions.

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

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

Networks-distributed processing:

Network:

A network is a set of devices (often referred to as nodes) connected by communication links. A node can be a computer, printer, or any other device capable of sending and/or receiving data generated by other nodes on the network.

Distributed processing:

Most networks use distributed processing, in which a task is divided among multiple computers. Instead of one single large machine being responsible for all aspects of a process, separate computers (usually a personal computer or workstation) handle a subset.

Network criteria:

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

Performance:

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

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

Reliability:

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

Security:

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

Physical structures:

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

Type of Connection:

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

Point-to-Point:

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

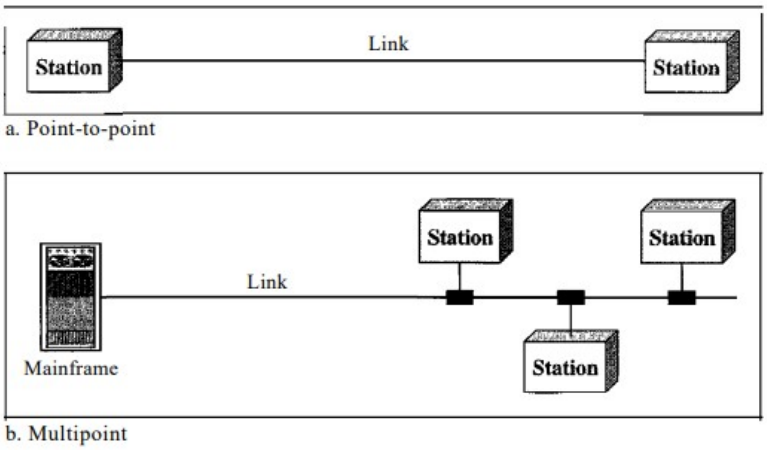
Multipoint:

A multipoint (also called multidrop) connection is one in which more than two specific devices share a single link (see Figure 1.3b). In a multipoint environment, the capacity of the channel is shared, either spatially or temporally. If several devices can use the link simultaneously, it is a spatially shared connection. If users must take turns, it is a timeshared connection.

Physical Topology:

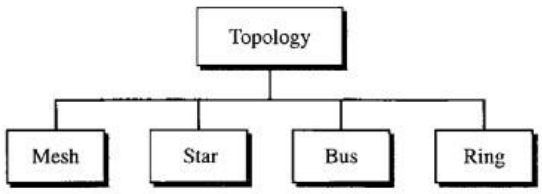
The term physical topology refers to the way in which a network is laid out physically or more devices connect to a link; two or more links form a topology. The topology

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



of a network is the geometric representation of the relationship of all the links and linking devices (usually called nodes) to one another. There are four basic topologies possible: mesh, star, bus, and ring (see Figure 1.4).

Figure 1.4 *Categories oftopology*



Mesh In a mesh topology, every device has a dedicated point-to-point link to every other device. The term dedicated means that the link carries traffic only between the two devices it connects. To find the number of physical links in a fully connected mesh network with n nodes, we first consider that each node must be connected to every other node. Node 1 must be connected to n - I nodes, node 2 must be connected to n - 1 nodes, and finally node n must be connected to n - 1 nodes. We need

$$n(n - 1)$$

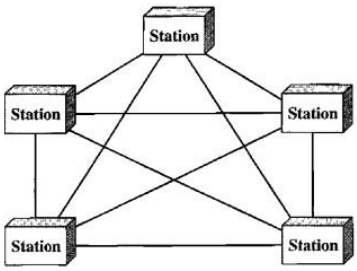
physical links. However, if each physical link allows communication in both directions (duplex mode), we can divide the number of links by 2. In other words, we can say that in a mesh topology, we need

$$n(n -1) /2$$

duplex-mode links.

To accommodate that many links, every device on the network must have n - 1 input/output (VO) ports (see Figure 1.5) to be connected to the other n - 1 stations.

Figure 1.5 *A fully connected mesh topology (five devices)*



Mesh:

A mesh offers several advantages over other network topologies. First, the use of dedicated links guarantees that each connection can carry its own data load, thus eliminating the traffic problems that can occur when links must be shared by multiple devices. Second, a mesh topology is robust. If one link becomes unusable, it does not incapacitate the entire system.

Third, there is the advantage of privacy or security. When every message travels along a dedicated line, only the intended recipient sees it. Physical boundaries prevent other users from gaining access to messages. Finally, point-to-point links make fault identification and fault isolation easy. Traffic can be routed to avoid links with suspected problems. This facility enables the network manager to discover the precise location of the fault and aids in finding its cause and solution.

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

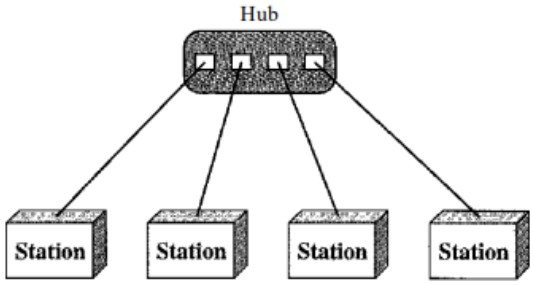
One practical example of a mesh topology is the connection of telephone regional offices in which each regional office needs to be connected to every other regional office.

Star Topology:

In a star topology, each device has a dedicated point-to-point link only to a central controller, usually called a hub. The devices are not directly linked to one another. Unlike a mesh topology, a star topology does not allow direct traffic between devices. The controller acts as an exchange: If one device wants to send data to another, it sends the data to the controller, which then relays the data to the other connected device (see Figure 1.6) .

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

Figure 1.6 *A star topology connecting four stations*



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

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

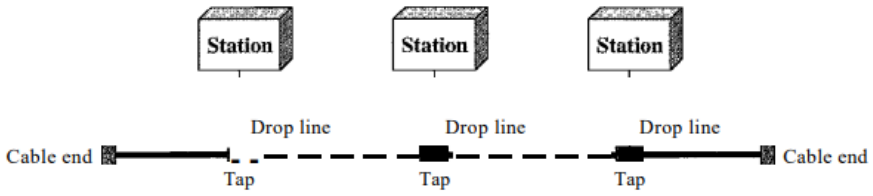
Although a star requires far less cable than a mesh, each node must be linked to a central hub. For this reason, often more cabling is required in a star than in some other topologies (such as ring or bus).

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

Bus Topology:

The preceding examples all describe point-to-point connections. A **bus topology,** on the other hand, is multipoint. One long cable acts as a **backbone** to link all the devices in a network (see Figure 1.7).

Figure 1.7 *A bus topology connecting three stations*



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

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

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

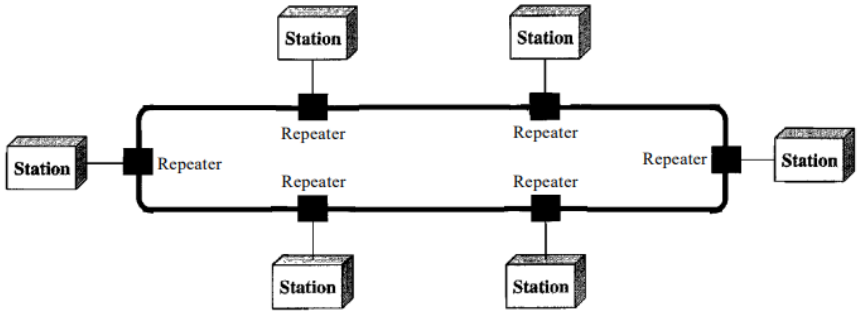
In addition, a fault or break in the bus cable stops all transmission, even between devices on the same side of the problem. The damaged area reflects signals back in the direction of origin, creating noise in both directions.

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

Ring Topology :

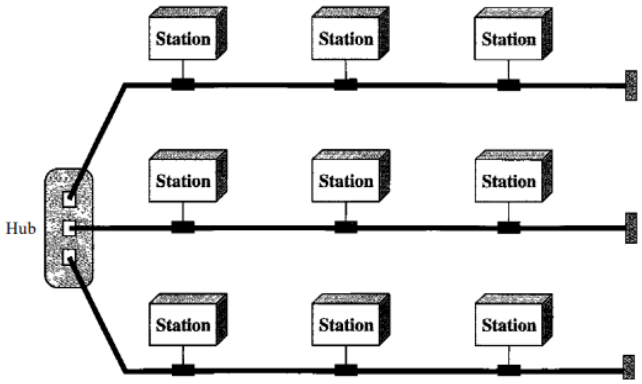
In a ring topology, each device has a dedicated point-to-point connection with only the two devices on either side of it. A signal is passed along the ring in one direction, from device to device, until it reaches its destination. Each device in the ring incorporates a repeater. When a device receives a signal intended for another device, its repeater regenerates the bits and passes them along (see Figure 1.8).

Figure 1.8 *A ring topology connecting six stations*



A ring is relatively easy to install and reconfigure. Each device is linked to only its immediate neigbhours (either physically or logically). To add or delete a device requires changing only two connections. The only constraints are media and traffic considerations (maximum ring length and number of devices). In addition, fault isolation is simplified. Generally in a ring, a signal is circulating at all times. If one device does not receive a signal within a specified period, it can issue an alarm. The alarm alerts the network operator to the problem and its location. However, unidirectional traffic can be a disadvantage. In a simple ring, a break in the ring (such as a disabled station) can disable the entire network. This weakness can be solved by using a dual ring or a switch capable of closing off the break.

Figure 1.9 A *hybrid topology: a star backbone with three bus networks*



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

Hybrid Topology

A network can be hybrid. For example, we can have a main star topology with each branch connecting several stations in a bus topology as shown in Figure 1.9.

Network models:

Computer networks are created by different entities. Standards are needed so that these heterogeneous networks can communicate with one another. The two best-known standards are the OSI model and the Internet model. In Chapter 2 we discuss these two models. The OSI (Open Systems Interconnection) model defines a seven-layer network; the Internet model defines a five-layer network. This book is based on the Internet model with occasional references to the OSI model.

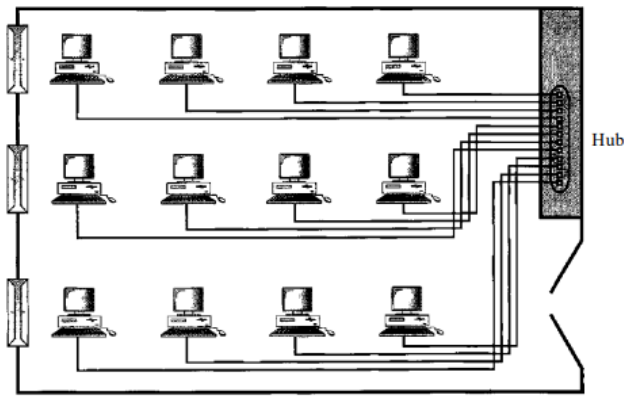
Categories of Networks:

Today when we speak of networks, we are generally referring to two primary categories: local-area networks and wide-area networks. The category into which a network falls is determined by its size. A LAN normally covers an area less than 2 mi; a WAN can be worldwide. Networks of a size in between are normally referred to as metropolitanarea networks and span tens of miles.

Local Area Network:

A local area network (LAN) is usually privately owned and links the devices in a single office, building, or campus (see Figure 1.10). Depending on the needs of an organization and the type of technology used, a LAN can be as simple as two PCs and a printer in someone's home office; or it can extend throughout a company and include audio and video peripherals. Currently, LAN size is limited to a few kilometers.

Figure 1.10 *An isolated IAN connecting* 12 *computers to a hub in a closet*



LANs are designed to allow resources to be shared between personal computers or workstations. The resources to be shared can include hardware (e.g., a printer), software (e.g., an application program), or data.

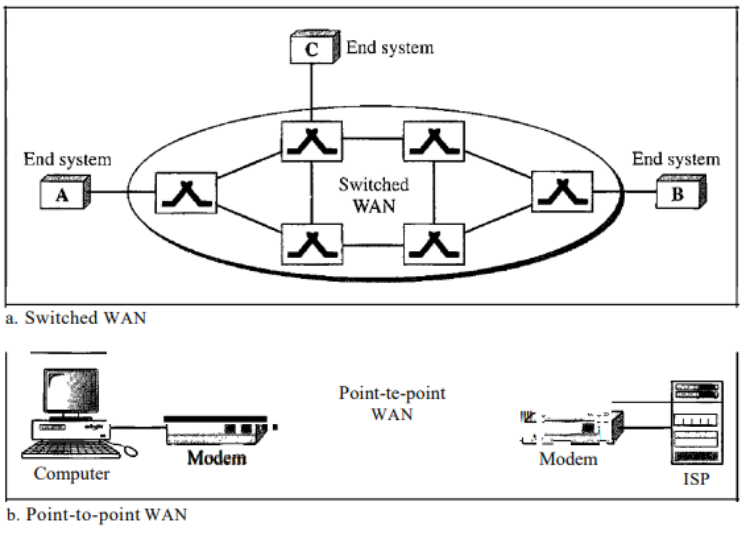
A common example of a LAN, found in many business environments, links a workgroup of task-related computers, for example, engineering workstations or accounting PCs. One of the computers may be given a large capacity disk drive and may become a server to clients. Software can be stored on this central server and used as needed by the whole group.

In this example, the size of the LAN may be determined by licensing restrictions on the number of users per copy of software, or by restrictions on the number of users licensed to access the operating system. In addition to size, LANs are distinguished from other types of networks by their transmission media and topology. In general, a given LAN will use only one type of transmission medium. The most common LAN topologies are bus, ring, and star. Early LANs had data rates in the 4 to 16 megabits per second (Mbps) range. Today, however, speeds are normally 100 or 1000 Mbps. LANs are discussed at length in Chapters 13, 14, and 15. Wireless LANs are the newest evolution in LAN technology. We discuss wireless LANs in detail in Chapter 14.

Wide Area Network:

A wide area network (WAN) provides long-distance transmission of data, image, audio, and video information over large geographic areas that may comprise a country, a continent, or even the whole world. In Chapters 17 and 18 we discuss wide-area networks in greater detail. A WAN can be as complex as the backbones that connect the Internet or as simple as a dial-up line that connects a home computer to the Internet. We normally refer to the first as a switched WAN and to the second as a point-to-point WAN (Figure 1.11). The switched WAN connects the end systems, which usually comprise a router (internetworking connecting device) that connects to another LAN or WAN. The point-to-point WAN is normally a line leased from a telephone or cable TV provider that connects a home computer or a small LAN to an Internet service provider (ISP). This type of WAN is often used to provide Internet access.

Figure 1.11 *WANs: a switched WAN and a point-to-point WAN*



An early example of a switched WAN is X.25, a network designed to provide connectivity between end users. As we will see in Chapter 18, X.25 is being gradually replaced by a high-speed, more efficient network called Frame Relay.

A good example of a switched WAN is the asynchronous transfer mode (ATM) network, which is a network with fixed-size data unit packets called cells. We will discuss ATM in Chapter 18. Another example of WANs is the wireless WAN that is becoming more and more popular. We discuss wireless WANs and their evolution in Chapter 16.

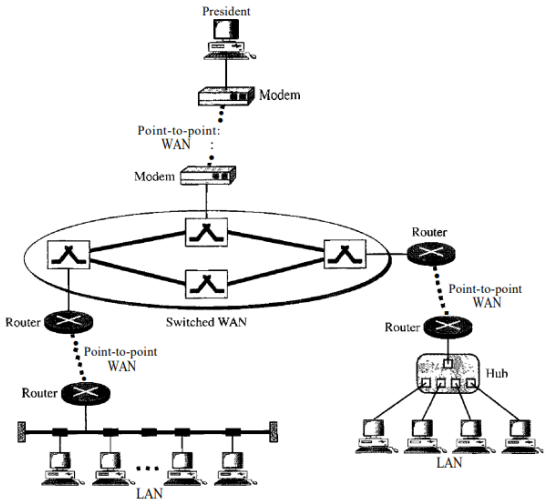
Metropolitan Area Networks

A metropolitan area network (MAN) is a network with a size between a LAN and a WAN. It normally covers the area inside a town or a city. It is designed for customers who need a high-speed connectivity, normally to the Internet, and have endpoints spread over a city or part of city. A good example of a MAN is the part of the telephone company network that can provide a high-speed DSL line to the customer. Another example is the cable TV network that originally was designed for cable TV, but today can also be used for high-speed data connection to the Internet. We discuss DSL lines and cable TV networks in Chapter 9.

Interconnection of Networks: Internetwork

Today, it is very rare to see a LAN, a MAN, or a LAN in isolation; they are connected to one another. When two or more networks are connected, they become an internetwork, or internet. As an example, assume that an organization has two offices, one on the east coast and the other on the west coast. The established office on the west coast has a bus topology LAN; the newly opened office on the east coast has a star topology LAN. The president of the company lives somewhere in the middle and needs to have control over the company from her horne. To create a backbone WAN for connecting these three entities (two LANs and the president's computer), a switched WAN (operated by a service provider such as a telecom company) has been leased. To connect the LANs to this switched WAN, however, three point-to-point WANs are required. These point-to-point WANs can be a high-speed DSL line offered by a telephone company or a cable modern line offered by a cable TV provider as shown in Figure 1.12.

Figure 1.12 *A heterogeneous network made offour WANs and two LANs*



The internet:

The Internet has revolutionized many aspects of our daily lives. It has affected the way we do business as well as the way we spend our leisure time. Count the ways you've used the Internet recently. Perhaps you've sent electronic mail (e-mail) to a business associate, paid a utility bill, read a newspaper from a distant city, or looked up a local movie schedule-all by using the Internet. Or maybe you researched a medical topic, booked a hotel reservation, chatted with a fellow Trekkie, or comparison-shopped for a car. The Internet is a communication system that has brought a wealth of information to our fingertips and organized it for our use.

The Internet is a structured, organized system. We begin with a brief history of the Internet. We follow with a description of the Internet today.

A brief History:

A network is a group of connected communicating devices such as computers and printers. An internet (note the lowercase letter i) is two or more networks that can communicate with each other. The most notable internet is called the Internet (uppercase letter I), a collaboration of more than hundreds of thousands of interconnected networks. Private individuals as well as various organizations such as government agencies, schools, research facilities, corporations, and libraries in more than 100 countries use the Internet. Millions of people are users. Yet this extraordinary communication system only came into being in 1969.

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

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

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

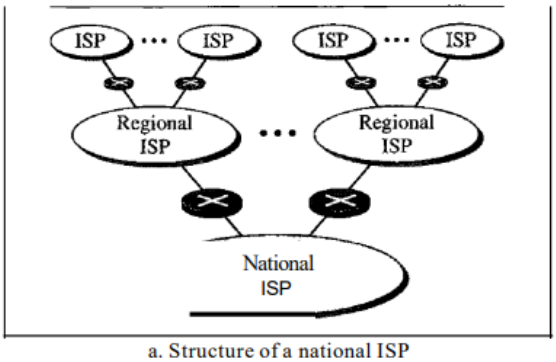
In 1972, Vint Cerf and Bob Kahn, both of whom were part of the core ARPANET group, collaborated on what they called the *Internetting Projec1*. Cerf and Kahn's landmark 1973 paper outlined the protocols to achieve end-to-end delivery of packets. This paper on Transmission Control Protocol (TCP) included concepts such as encapsulation, the datagram, and the functions of a gateway.

Shortly thereafter, authorities made a decision to split TCP into two protocols: Transmission Control Protocol (TCP) and Internetworking Protocol (lP). IP would handle datagram routing while TCP would be responsible for higher-level functions such as segmentation, reassembly, and error detection. The internetworking protocol became known as TCPIIP.

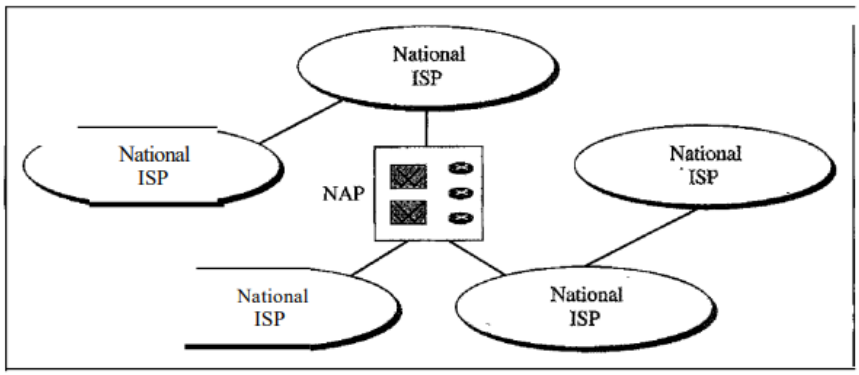
The internet today:

The Internet has come a long way since the 1960s. The Internet today is not a simple hierarchical structure. It is made up of many wide- and local-area networks joined by connecting devices and switching stations. It is difficult to give an accurate representation of the Internet because it is continually changing-new networks are being added, existing networks are adding addresses, and networks of defunct companies are being removed. Today most end users who want Internet connection use the services of Internet service providers (ISPs). There are international service providers, national service providers, regional service providers, and local service providers. The Internet today is run by private companies, not the government. Figure 1.13 shows a conceptual (not geographic) view of the Internet.

Figure 1.13 *Hierarchical organization of the Internet*



a. Structure of a national ISP



b. Interconnection of national ISPs

International Internet Service Providers:

At the top of the hierarchy are the international service providers that connect nations together.

National Internet Service Providers :

The national Internet service providers are backbone networks created and maintained by specialized companies. There are many national ISPs operating in North America; some of the most well known are SprintLink, PSINet, UUNet Technology, AGIS, and internet Mel. To provide connectivity between the end users, these backbone networks are connected by complex switching stations (normally run by a third party) called network access points (NAPs). Some national ISP networks are also connected to one another by private switching stations called peering points. These normally operate at a high data rate (up to 600 Mbps).

Regional Internet Service Providers:

Regional internet service providers or regional ISPs are smaller ISPs that are connected to one or more national ISPs. They are at the third level of the hierarchy with a smaller data rate.

Local Internet Service Providers:

Local Internet service providers provide direct service to the end users. The local ISPs can be connected to regional ISPs or directly to national ISPs. Most end users are connected to the local ISPs. Note that in this sense, a local ISP can be a company that just provides Internet services, a corporation with a network that supplies services to its own employees, or a nonprofit organization, such as a college or a university, that runs its own network. Each of these local ISPs can be connected to a regional or national service provider.

Protocols and standard:

In this section, we define two widely used terms: protocols and standards. First, we define **protocol,** which is synonymous with rule. Then we discuss **standards**, which are agreed-upon rules.

Protocols :

In computer networks, communication occurs between entities in different systems. An entity is anything capable of sending or receiving information. However, two entities cannot simply send bit streams to each other and expect to be understood. For communication to occur, the entities must agree on a protocol. A protocol is a set of rules that govern data communications. A protocol defines what is communicated, how it is communicated, and when it is communicated. The key elements of a protocol are syntax, semantics, and timing.

Syntax: The term syntax refers to the structure or format of the data, meaning the order in which they are presented. For example, a simple protocol might expect the first 8 bits of data to be the address of the sender, the second 8 bits to be the address of the receiver, and the rest of the stream to be the message itself.

Semantics: The word semantics refers to the meaning of each section of bits. How is a particular pattern to be interpreted, and what action is to be taken based on that interpretation? .For example, does an address identify the route to be taken or the final destination of the message?

Timing: The term timing refers to two characteristics: when data should be sent and how fast they can be sent. For example, if a sender produces data at 100 Mbps but the receiver can process data at only 1 Mbps, the transmission will overload the receiver and some data will be lost.

Standards:

Standards are essential in creating and maintaining an open and competitive market for equipment manufacturers and in guaranteeing national and international interoperability of data and telecommunications technology and processes. Standards provide guidelines to manufacturers, vendors, government agencies, and other service providers to ensure the kind of interconnectivity necessary in today's marketplace and in international communications. Data communication standards fall into two categories: de facto (meaning "by fact" or "by convention") and de jure (meaning "by law" or "by regulation").

De facto: Standards that have not been approved by an organized body but have been adopted as standards through widespread use are de facto standards. De facto standards are often established originally by manufacturers who seek to define the functionality of a new product or technology.

De jure: Those standards that have been legislated by an officially recognized body are de jure standards.

Standards Organizations :

Standards are developed through the cooperation of standards creation committees, forums, and government regulatory agencies.

Standards Creation Committees

While many organizations are dedicated to the establishment of standards, data telecommunications in North America rely primarily on those published by the following:

International Organization for Standardization (ISO). The ISO is a multinational body whose membership is drawn mainly from the standards creation committees of various governments throughout the world. The ISO is active in developing cooperation in the realms of scientific, technological, and economic activity.

International Telecommunication Union-Telecommunication Standards Sector (ITU-T). By the early 1970s, a number of countries were defining national standards for telecommunications, but there was still little international compatibility. The United Nations responded by forming, as part of its International Telecommunication Union (ITU), a committee, the Consultative Committee for International Telegraphy and Telephony (CCITT). This committee was devoted to the research and establishment of standards for telecommunications in general and for phone and data systems in particular. On March 1, 1993, the name of this committee was changed to the International Telecommunication Union Telecommunication Standards Sector (ITU-T).

American National Standards Institute (ANSI). Despite its name, the American National Standards Institute is a completely private, nonprofit corporation not affiliated with the U.S. federal government. However, all ANSI activities are undertaken with the welfare of the United States and its citizens occupying primary importance.

Institute of Electrical and Electronics Engineers (IEEE). The Institute of Electrical and Electronics Engineers is the largest professional engineering society in the world. International in scope, it aims to advance theory, creativity, and product quality in the fields of electrical engineering, electronics, and radio as well as in all related branches of engineering. As one of its goals, the IEEE oversees the development and adoption of international standards for computing and communications.

Electronic Industries Association (EIA). Aligned with ANSI, the Electronic Industries Association is a non profit organization devoted to the promotion of electronics manufacturing concerns. Its activities include public awareness education and lobbying efforts in addition to standards development. In the field of information technology, the EIA has made significant contributions by defining physical connection interfaces and electronic signaling specifications for data communication.

***Forums**:*

Telecommunications technology development is moving faster than the ability of standards committees to ratify standards. Standards committees are procedural bodies and by nature slow-moving. To accommodate the need for working models and agreements and to facilitate the standardization process, many special-interest groups have developed forums made up of representatives from interested corporations. The forums work with universities and users to test, evaluate, and standardize new technologies. By concentrating their efforts on a particular technology, the forums are able to speed acceptance and use of those technologies in the telecommunications community. The forums present their conclusions to the standards bodies.

Regulatory Agencies:

All communications technology is subject to regulation by government agencies such as the **Federal Communications Commission (FCC)** in the United States. The purpose of these agencies is to protect the public interest by regulating radio, television, and wire/cable communications. The FCC has authority over interstate and international commerce as it relates to communications.

Internet Standards

An **Internet standard** is a thoroughly tested specification that is useful to and adhered to by those who work with the Internet. It is a formalized regulation that must be followed. There is a strict procedure by which a specification attains Internet standard status. A specification begins as an Internet draft. An **Internet draft** is a working document (a work in progress) with no official status and a 6-month lifetime. Upon recommendation from the Internet authorities, a draft may be published as a **Request for Comment** (RFC). Each RFC is edited, assigned a number, and made available to all interested parties. RFCs go through maturity levels and are categorized according to their requirement level.

Network models:

A network is a combination of hardware and software that sends data from one location to another. The hardware consists of the physical equipment that carries signals from one point of the network to another. The software consists of instruction sets that make possible the services that we expect from a network.

We can compare the task of networking to the task of solving a mathematics problem with a computer. The fundamental job of solving the problem with a computer is done by computer hardware. However, this is a very tedious task if only hardware is involved. We would need switches for every memory location to store and manipulate data. The task is much easier if software is available. At the highest level, a program can direct the problem-solving process; the details of how this is done by the actual hardware can be left to the layers of software that are called by the higher levels.

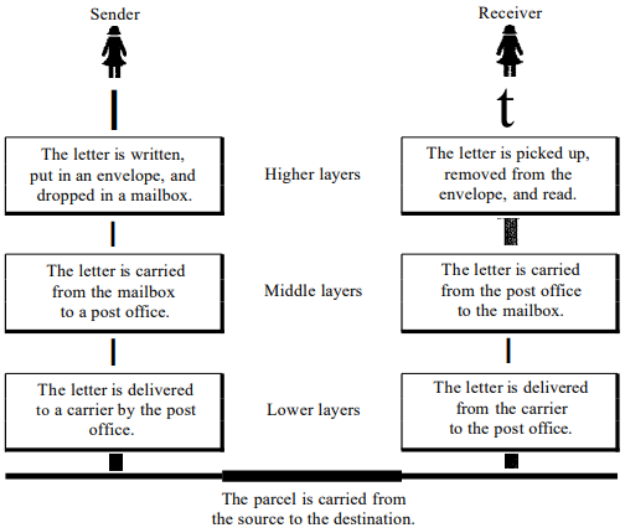
Compare this to a service provided by a computer network. For example, the task of sending an e-mail from one point in the world to another can be broken into several tasks, each performed by a separate software package. Each software package uses the services of another software package. At the lowest layer, a signal, or a set of signals, is sent from the source computer to the destination computer.

In this chapter, we give a general idea of the layers of a network and discuss the functions of each. Detailed descriptions of these layers follow in later chapters.

Layered tasks:

We use the concept of layers in our daily life. As an example, let us consider two friends who communicate through postal mail The process of sending a letter to a friend would be complex if there were no services available from the post office. Figure 2.1 show the steps in this task.

Figure 2.1 *Tasks involved in sending a letter*



Sender, Receiver, and Carrier

In Figure 2.1 we have a sender, a receiver, and a carrier that transports the letter. There is a hierarchy of tasks.

At the Seller Site

Let us first describe, in order, the activities that take place at the sender site.

Higher layer: The sender writes the letter, inserts the letter in an envelope, writes the sender and receiver addresses, and drops the letter in a mailbox.

Middle layer: The letter is picked up by a letter carrier and delivered to the post office.

Lower layer: The letter is sorted at the post office; a carrier transports the letter.

011 the Way :

The letter is then on its way to the recipient. On the way to the recipient's local post office, the letter may actually go through a central office. In addition, it may be transported by truck, train, airplane, boat, or a combination of these.

At the Receiver Site

Lower layer: The carrier transports the letter to the post office.

Middle layer: The letter is sorted and delivered to the recipient's mailbox.

Higher layer: The receiver picks up the letter, opens the envelope, and reads it.

Hierarchy :

According to our analysis, there are three different activities at the sender site and another three activities at the receiver site. The task of transporting the letter between the sender and the receiver is done by the carrier. Something that is not obvious immediately is that the tasks must be done in the order given in the hierarchy. At the sender site, the letter must be written and dropped in the mailbox before being picked up by the letter carrier and delivered to the post office. At the receiver site, the letter must be dropped in the recipient mailbox before being picked up and read by the recipient.

Services:

Each layer at the sending site uses the services of the layer immediately below it. The sender at the higher layer uses the services of the middle layer. The middle layer uses the services of the lower layer. The lower layer uses the services of the carrier.

The layered model that dominated data communications and networking literature before 1990 was the Open Systems Interconnection (OSI) model. Everyone believed that the OSI model would become the ultimate standard for data communications, but this did not happen. The TCPIIP protocol suite became the dominant commercial architecture because it was used and tested extensively in the Internet; the OSI model was never fully implemented.

In this chapter, first we briefly discuss the OSI model, and then we concentrate on TCPIIP as a protocol suite.

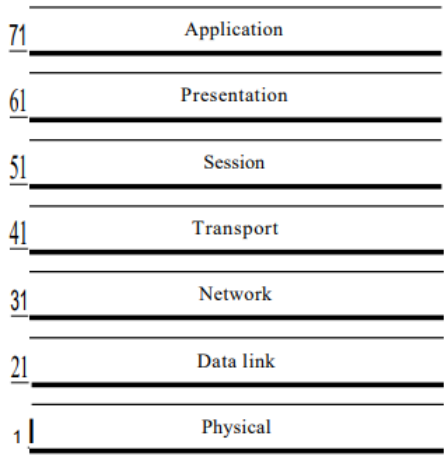
The OSI model:

Established in 1947, the International Standards Organization (ISO) is a multinational body dedicated to worldwide agreement on international standards. An ISO standard that covers all aspects of network communications is the Open Systems Interconnection model. It was first introduced in the late 1970s. An open system is a set of protocols that allows any two different systems to communicate regardless of their underlying architecture. The purpose of the OSI model is to show how to facilitate communication between different systems without requiring changes to the logic of the underlying hardware and software. The OSI model is not a protocol; it is a model for understanding and designing a network architecture that is flexible, robust, and interoperable.

ISO is the organization. OSI is the model.

The OSI model is a layered framework for the design of network systems that allows communication between all types of computer systems. It consists of seven separate but related layers, each of which defines a part of the process of moving information across a network (see Figure 2.2). An understanding of the fundamentals of the OSI model provides a solid basis for exploring data communications.

Figure 2.2 *Seven layers of the OSI model*



Layered **Architecture:**

The OSI model is composed of seven ordered layers: physical (layer 1), data link (layer 2), network (layer 3), transport (layer 4), session (layer 5), presentation (layer 6), and application (layer 7). Figure 2.3 shows the layers involved when a message is sent from device A to device B. As the message travels from A to B, it may pass through many intermediate nodes. These intermediate nodes usually involve only the first three layers of the OSI model.

In developing the model, the designers distilled the process of transmitting data to its most fundamental elements. They identified which networking functions had related uses and collected those functions into discrete groups that became the layers. Each layer defines a family of functions distinct from those of the other layers. By defining and localizing functionality in this fashion, the designers created an architecture that is both comprehensive and flexible. Most importantly, the OSI model allows complete interoperability between otherwise incompatible systems.

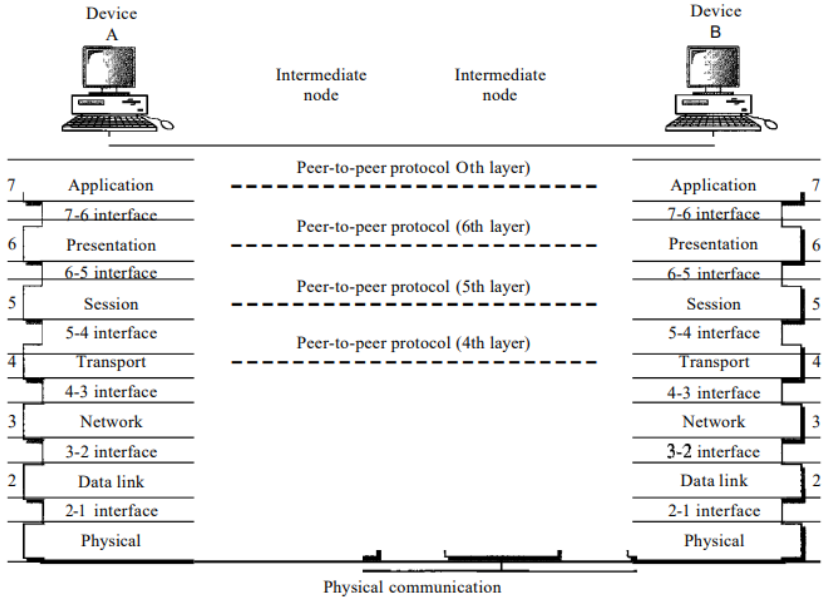
Within a single machine, each layer calls upon the services of the layer just below it. Layer 3, for example, uses the services provided by layer 2 and provides services for layer 4. Between machines, layer x on one machine communicates with layer x on another machine. This communication is governed by an agreed-upon series of rules and conventions called protocols. The processes on each machine that communicate at a given layer are called peer-to-peer processes. Communication between machines is therefore a peer-to-peer process using the protocols appropriate to a given layer.

Peer-to-Peer Processes:

At the physical layer, communication is direct: In Figure 2.3, device A sends a stream of bits to device B (through intermediate nodes). At the higher layers, however, communication must move down through the layers on device A, over to device B, and then back up through the layers. Each layer in the sending device adds its own information to the message it receives from the layer just above it and passes the whole package to the layer just below it.

At layer I .the entire package is converted to a form that can be transmitted to the receiving device. At the receiving machine, the message is unwrapped layer by layer, with each process receiving and removing the data meant for it. For example, layer 2. removes the data meant for it, then passes the rest to layer 3. Layer 3 then removes the data meant for it and passes the rest to layer 4, and so on.

Figure 2.3 *The interaction between layers in the OSI model*



Interfaces Between Layers:

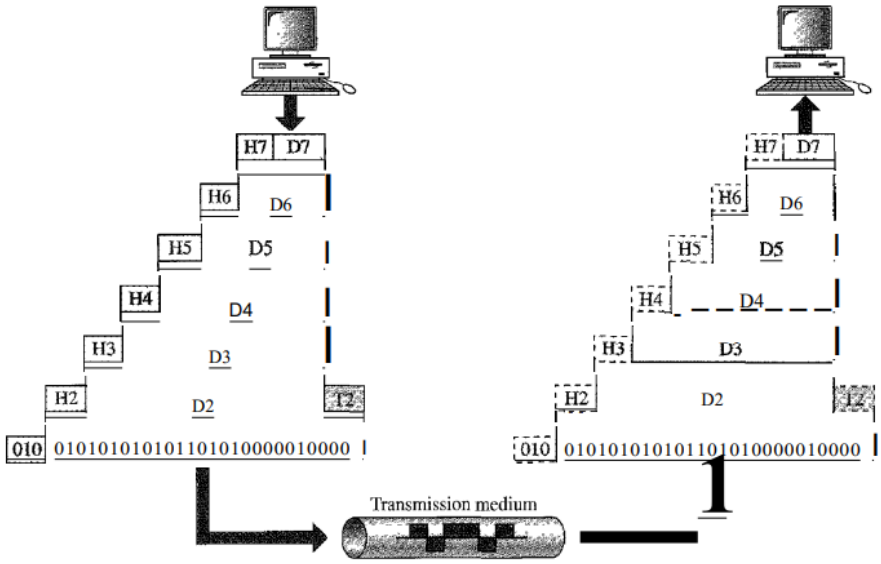
The passing of the data and network information down through the layers of the sending device and back up through the layers of the receiving device is made possible by an interface between each pair of adjacent layers. Each interface defines the information and services a layer must provide for the layer above it. Well-defined interfaces and layer functions provide modularity to a network. As long as a layer provides the expected services to the layer above it, the specific implementation of its functions can be modified or replaced without requiring changes to the surrounding layers.

Organization of the Layers:

The seven layers can be thought of as belonging to three subgroups. Layers I, 2, and 3-physical, data link, and network-are the network support layers; they deal with the physical aspects of moving data from one device to another (such as electrical specifications, physical connections, physical addressing, and transport timing and reliability). Layers 5, 6, and 7-session, presentation, and application-can be thought of as the user support layers; they allow interoperability among unrelated software systems. Layer 4, the transport layer, links the two subgroups and ensures that what the lower layers have transmitted is in a form that the upper layers can use. The upper OSI layers are almost always implemented in software; lower layers are a combination of hardware and software, except for the physical layer, which is mostly hardware.

In Figure 2.4, which gives an overall view of the OSI layers, D7 means the data unit at layer 7, D6 means the data unit at layer 6, and so on. The process starts at layer 7 (the application layer), then moves from layer to layer in descending, sequential order. At each layer, a **header**, or possibly a **trailer**, can be added to the data unit. Commonly, the trailer is added only at layer 2. When the formatted data unit passes through the physical layer (layer 1), it is changed into an electromagnetic signal and transported along a physical link.

Figure 2.4 *An exchange using the OS! model*



Upon reaching its destination, the signal passes into layer 1 and is transformed back into digital form. The data units then move back up through the OSI layers. As each block of data reaches the next higher layer, the headers and trailers attached to it at the corresponding sending layer are removed, and actions appropriate to that layer are taken. By the time it reaches layer 7, the message is again in a form appropriate to the application and is made available to the recipient.

Encapsulation:

Figure 2.3 reveals another aspect of data communications in the OSI model: encapsulation. A packet (header and data) at level 7 is encapsulated in a packet at level 6. The whole packet at level 6 is encapsulated in a packet at level 5, and so on.

In other words, the data portion of a packet at level N - 1 carries the whole packet (data and header and maybe trailer) from level N. The concept is called encapsulation; level N - 1 is not aware of which part of the encapsulated packet is data and which part is the header or trailer. For level N - 1, the whole packet coming from level N is treated as one integral unit.

LAYERS IN THE OSI MODEL:

In this section we briefly describe the functions of each layer in the OSI model.

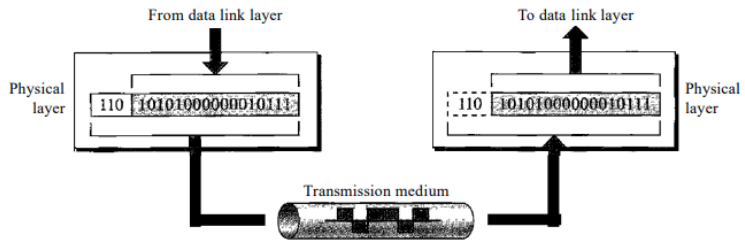
Physical Layer

The physical layer coordinates the functions required to carry a bit stream over a physical medium

It deals with the mechanical and electrical specifications of the interface and transmission medium.

It also defines the procedures and functions that physical devices and interfaces have to perform for transmission to Occur. Figure 2.5 shows the position of the physical layer with respect to the transmission medium and the data link layer.

Figure 2.5 *Physical layer*



The physical layer is responsible for movements of individual bits from one hop (node) to the next.

The physical layer is also concerned with the following:

Physical characteristics of interfaces and medium: The physical layer defines the characteristics of the interface between the devices and the transmission medium. It also defines the type of transmission medium.

Representation of bits: The physical layer data consists of a stream of bits (sequence of Os or 1s) with no interpretation. To be transmitted, bits must be encoded into signals--electrical or optical. The physical layer defines the type of encoding (how Os and Is are changed to signals).

Data rate: The transmission rate-the number of bits sent each second-is also defined by the physical layer. In other words, the physical layer defines the duration of a bit, which is how long it lasts.

Synchronization of bits: The sender and receiver not only must use the same bit rate but also must be synchronized at the bit level. In other words, the sender and the receiver clocks must be synchronized.

Line configuration: The physical layer is concerned with the connection of devices to the media. In a point-to-point configuration, two devices are connected through a dedicated link. In a multipoint configuration, a link is shared among several devices.

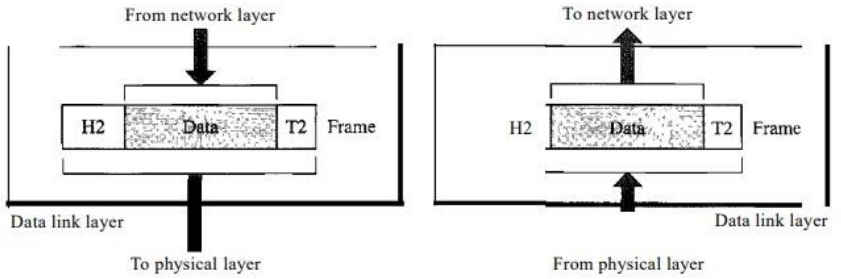
Physical topology: The physical topology defines how devices are connected to make a network. Devices can be connected by using a mesh topology (every device is connected to every other device), a star topology (devices are connected through a central device), a ring topology (each device is connected to the next, forming a ring), a bus topology (every device is on a common link), or a hybrid topology (this is a combination of two or more topologies).

Transmission mode: The physical layer also defines the direction of transmission between two devices: simplex, half-duplex, or full-duplex. In simplex mode, only one device can send; the other can only receive. The simplex mode is a one-way communication. In the half-duplex mode, two devices can send and receive, but not at the same time. In a full-duplex (or simply duplex) mode, two devices can send and receive at the same time.

Data Link Layer

The data link layer transforms the physical layer, a raw transmission facility, to a reliable link. It makes the physical layer appear error-free to the upper layer (network layer). Figure 2.6 shows the relationship of the data link layer to the network and physical layers.

Figure 2.6 *Data link layer*



The data link layer is responsible for moving frames from one hop (node) to the next.

Other responsibilities of the data link layer include the following:

Framing. The data link layer divides the stream of bits received from the network layer into manageable data units called frames.

Physical addressing. If frames are to be distributed to different systems on the network, the data link layer adds a header to the frame to define the sender and/or receiver of the frame. If the frame is intended for a system outside the sender's network, the receiver address is the address of the device that connects the network to the next one.

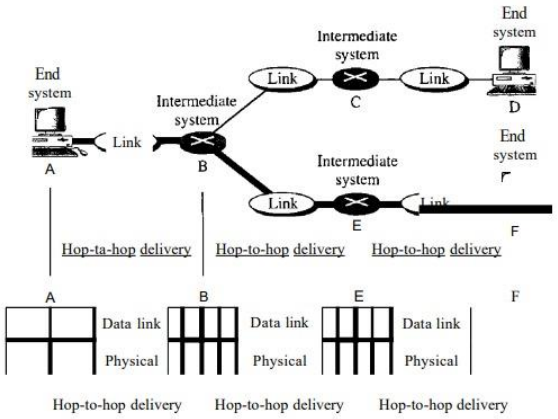
Flow control. If the rate at which the data are absorbed by the receiver is less than the rate at which data are produced in the sender, the data link layer imposes a flow control mechanism to avoid overwhelming the receiver.

Error control. The data link layer adds reliability to the physical layer by adding mechanisms to detect and retransmit damaged or lost frames. It also uses a mechanism to recognize duplicate frames. Error control is normally achieved through a trailer added to the end of the frame.

Access control. When two or more devices are connected to the same link, data link layer protocols are necessary to determine which device has control over the link at any given time. Figure 2.7 illustrates hop-to-hop (node-to-node) delivery by the data link layer.

As the figure shows, communication at the data link layer occurs between two adjacent nodes. To send data from A to F, three partial deliveries are made. First, the data link layer at A sends a frame to the data link layer at B (a router). Second, the data link layer at B sends a new frame to the data link layer at E. Finally, the data link layer at E sends a new frame to the data link layer at F. Note that the frames that are exchanged between the three nodes have different values in the headers. The frame from A to B has B as the destination address and A as the source address. The frame from B to E has E as the destination address and B as the source address. The frame from E to F has F as the destination address and E as the source address. The values of the trailers can also be different if error checking includes the header of the frame.

Figure 2.7 *Hop-to-hop delivery*

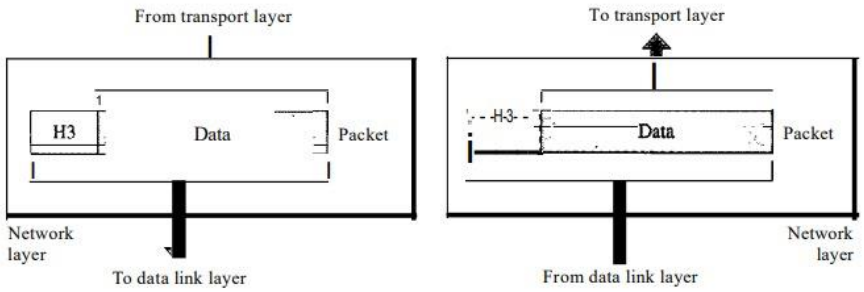


Network Layer

The network layer is responsible for the source-to-destination delivery of a packet, possibly across multiple networks (links). Whereas the data link layer oversees the delivery of the packet between two systems on the same network (links), the network layer ensures that each packet gets from its point of origin to its final destination.

If two systems are connected to the same link, there is usually no need for a network layer. However, if the two systems are attached to different networks (links) with connecting devices between the networks (links), there is often a need for the network layer to accomplish source-to-destination delivery. Figure 2.8 shows the relationship of the network layer to the data link and transport layers.

Figure 2.8 *Network layer*



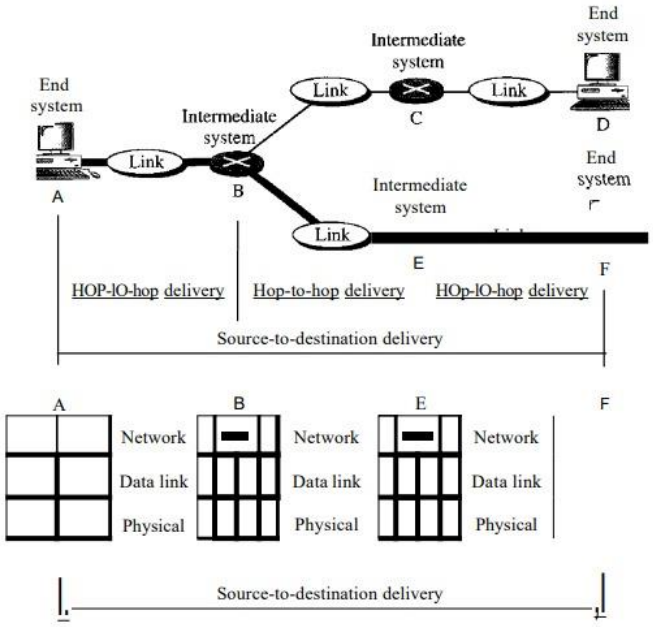
The network layer is responsible for the delivery of individual packets from the source host to the destination host.

Other responsibilities of the network layer include the following:

Logical addressing. The physical addressing implemented by the data link layer handles the addressing problem locally. If a packet passes the network boundary, we need another addressing system to help distinguish the source and destination systems. The network layer adds a header to the packet coming from the upper layer that, among other things, includes the logical addresses of the sender and receiver. We discuss logical addresses later in this chapter.

Routing : When independent networks or links are connected to create *intemetworks (*network of networks) or a large network, the connecting devices (called routers or switches) route or switch the packets to their final destination. One of the functions of the network layer is to provide this mechanism.

Figure 2.9 *Source-to-destination delivery*



As the figure shows, now we need a source-to-destination delivery. The network layer at A sends the packet to the network layer at B. When the packet arrives at router B, the router makes a decision based on the final destination (F) of the packet. As we will see in later chapters, router B uses its routing table to find that the next hop is router E. The network layer at B, therefore, sends the packet to the network layer at E. The network layer at E, in tum, sends the packet to the network layer at F.

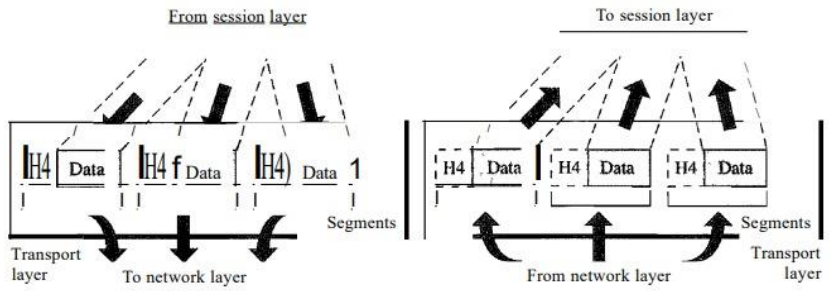
Transport Layer

The transport layer is responsible for process-to-process delivery of the entire message. A process is an application program running on a host. Whereas the network layer oversees source-to-destination delivery of individual packets, it does not recognize any relationship between those packets. It treats each one independently, as though each piece belonged to a separate message, whether or not it does. The transport layer, on the other hand, ensures that the whole message arrives intact and in order, overseeing both error control and flow control at the source-to-destination level. Figure 2.10 shows the relationship of the transport layer to the network and session layers.

Other responsibilities of the transport layer include the following:

Service-point addressing. Computers often run several programs at the same time. For this reason, source-to-destination delivery means delivery not only from one computer to the next but also from a specific process (running program) on one computer to a specific process (running program) on the other. The transport layer header must therefore include a type of address called a service-point address (or port address). The network layer gets each packet to the correct computer; the transport layer gets the entire message to the correct process on that computer.

Figure 2.10 *Transport layer*



The transport layer is responsible for the delivery of a message from one process to another.

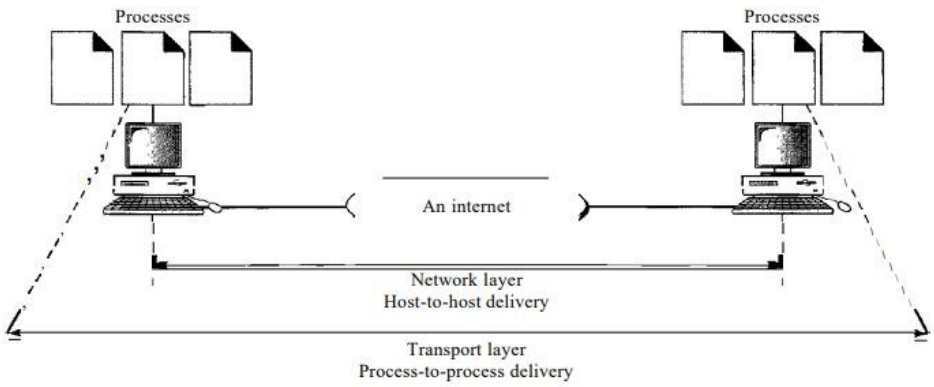
Segmentation and reassembly. A message is divided into transmittable segments, with each segment containing a sequence number. These numbers enable the transport layer to reassemble the message correctly upon arriving at the destination and to identify and replace packets that were lost in transmission.

Connection control. The transport layer can be either connectionless or connection oriented. A connectionless transport layer treats each segment as an independent packet and delivers it to the transport layer at the destination machine. A connection oriented transport layer makes a connection with the transport layer at the destination machine first before delivering the packets. After all the data are transferred, the connection is terminated.

Flow control. Like the data link layer, the transport layer is responsible for flow control. However, flow control at this layer is performed end to end rather than across a single link.

Error control. Like the data link layer, the transport layer is responsible for error control. However, error control at this layer is performed process-to-process rather than across a single link. The sending transport layer makes sure that the entire message arrives at the receiving transport layer without error (damage, loss, or duplication). Error correction is usually achieved through retransmission.

Figure 2.11 *Reliable process-to-process delivery of a message*



Session Layer

The services provided by the first three layers (physical, data link, and network) are not sufficient for some processes. The session layer is the network dialog controller. It establishes, maintains, and synchronizes the interaction among communicating systems.

The session layer is responsible for dialog control and synchronization.

Specific responsibilities of the session layer include the following:

Dialog control. The session layer allows two systems to enter into a dialog. It allows the communication between two processes to take place in either half duplex (one way at a time) or full-duplex (two ways at a time) mode.

Synchronization. The session layer allows a process to add checkpoints, or synchronization points, to a stream of data. For example, if a system is sending a file of 2000 pages, it is advisable to insert checkpoints after every 100 pages to ensure that each 100-page unit is received and acknowledged independently. In this case, if a crash happens during the transmission of page 523, the only pages that need to be resent after system recovery are pages 501 to 523. Pages previous to 501 need not be resent. Figure 2.12 illustrates the relationship of the session layer to the transport and presentation layers.

Presentation Layer

The presentation layer is concerned with the syntax and semantics of the information exchanged between two systems. Figure 2.13 shows the relationship between the presentation layer and the application and session layers.

Figure 2.12 *Session layer*

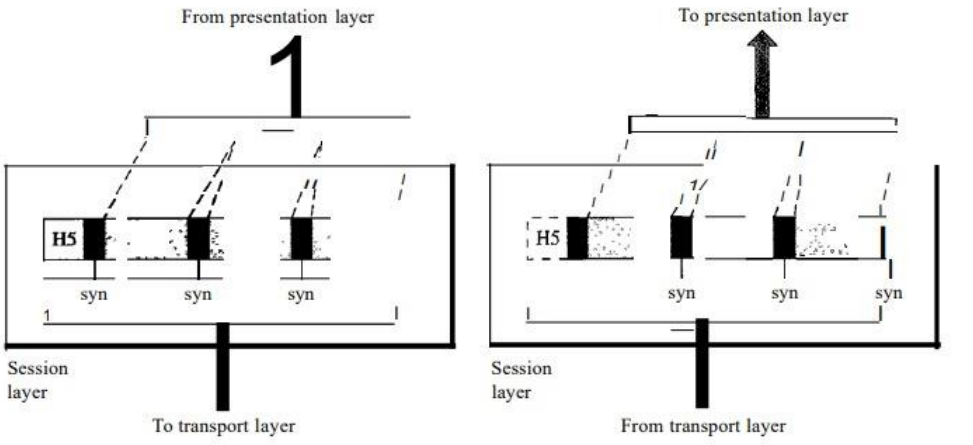
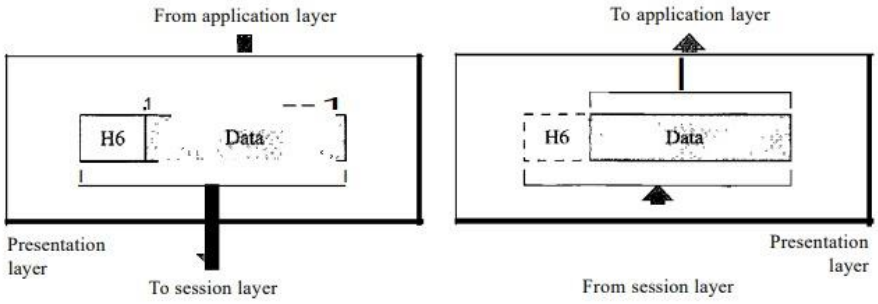


Figure 2.13 *Presentation layer*



The presentation layer is responsible for translation, compression, and encryption.

Specific responsibilities of the presentation layer include the following:

Translation. The processes (running programs) in two systems are usually exchanging information in the form of character strings, numbers, and so on. The infonnation must be changed to bit streams before being transmitted. Because different computers use different encoding systems, the presentation layer is responsible for interoperability between these different encoding methods. The presentation layer at the sender changes the information from its sender-dependent format into a common format. The presentation layer at the receiving machine changes the common format into its receiver-dependent format.

Encryption. To carry sensitive information, a system must be able to ensure privacy. Encryption means that the sender transforms the original information to another form and sends the resulting message out over the network. Decryption reverses the original process to transform the message back to its original form.

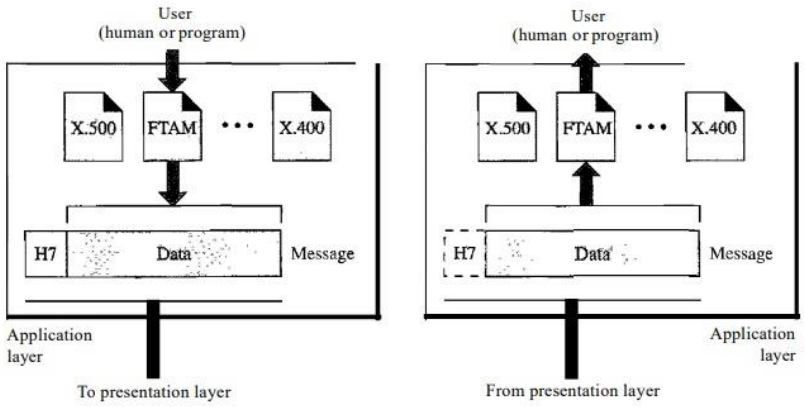
Compression. Data compression reduces the number of bits contained in the information. Data compression becomes particularly important in the transmission of multimedia such as text, audio, and video.

Application Layer

The application layer enables the user, whether human or software, to access the network. It provides user interfaces and support for services such as electronic mail, remote file access and transfer, shared database management, and other types of distributed information services.

Figure 2.14 shows the relationship of the application layer to the user and the presentation layer. Of the many application services available, the figure shows only three: XAOO (message-handling services), X.500 (directory services), and file transfer, access, and management (FTAM). The user in this example employs XAOO to send an e-mail message.

Figure 2.14 *Application layer*



The application layer is responsible for providing services to the user.

Specific services provided by the application layer include the following:

Network virtual terminal. A network virtual terminal is a software version of a physical terminal, and it allows a user to log on to a remote host. To do so, the application creates a software emulation of a terminal at the remote host. The user's computer talks to the software terminal which, in turn, talks to the host, and vice versa. The remote host believes it is communicating with one of its own terminals and allows the user to log on.

File transfer, access, and management. This application allows a user to access files in a remote host (to make changes or read data), to retrieve files from a remote computer for use in the local computer, and to manage or control files in a remote computer locally.

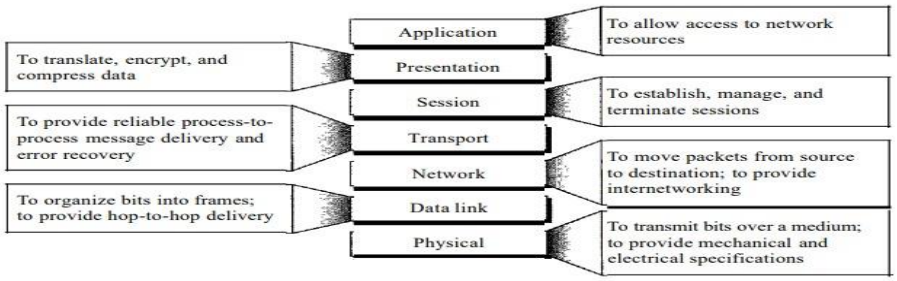
Mail services. This application provides the basis for e-mail forwarding and storage.

Directory services. This application provides distributed database sources and access for global information about various objects and services.

Summary of Layers

Figure 2.15 shows a summary of duties for each layer.

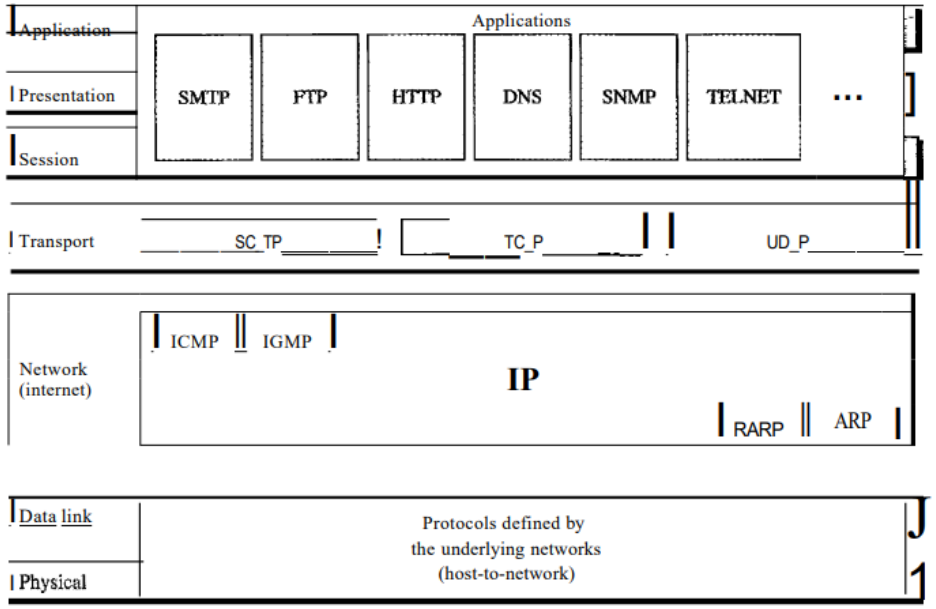
Figure 2.15 *Summary of layers*



TCP/IP PROTOCOL SUITE:

The TCPIIP protocol suite was developed prior to the OSI model. Therefore, the layers in the TCP/IP protocol suite do not exactly match those in the OSI model. The original TCP/IP protocol suite was defined as having four layers: host-to-network, internet, transport, and application. However, when TCP/IP is compared to OSI, we can say that the host-to-network layer is equivalent to the combination of the physical and data link layers. The internet layer is equivalent to the network layer, and the application layer is roughly doing the job of the session, presentation, and application layers with the transport layer in TCPIIP taking care of part of the duties of the session layer. So in this book, we assume that the TCPIIP protocol suite is made of five layers: physical, data link, network, transport, and application. The first four layers provide physical standards, network interfaces, internetworking, and transport functions that correspond to the first four layers of the OSI model. The three topmost layers in the OSI model, however, are represented in TCPIIP by a single layer called the application layer (see Figure 2.16).

Figure 2.16 *TCPIIP and OSI model*



TCP/IP is a hierarchical protocol made up of interactive modules, each of which provides a specific functionality; however, the modules are not necessarily interdependent. Whereas the OSI model specifies which functions belong to each of its layers, the layers of the TCP/IP protocol suite contain relatively independent protocols that can be mixed and matched depending on the needs of the system. The term *hierarchical means* that each upper-level protocol is supported by one or more lower-level protocols.

At the transport layer, *TCP/IP* defines three protocols: Transmission Control Protocol (TCP), User Datagram Protocol (UDP), and Stream Control Transmission Protocol (SCTP). At the network layer, the main protocol defined by TCP/IP is the Internetworking Protocol (IP); there are also some other protocols that support data movement in this layer.

Physical and Data Link Layers:

At the physical and data link layers, TCPIIP does not define any specific protocol. It supports all the standard and proprietary protocols. A network in a TCPIIP internetwork can be a local-area network or a wide-area network.

Network Layer:

At the network layer (or, more accurately, the internetwork layer), TCP/IP supports the Internetworking Protocol. IP, in turn, uses four supporting protocols: ARP, RARP, ICMP, and IGMP. Each of these protocols is described in greater detail in later chapters.

Internetworking Protocol (IP):

The Internetworking Protocol (IP) is the transmission mechanism used by the TCP/IP protocols. It is an unreliable and connectionless protocol-a best-effort delivery service. The term best effort means that IP provides no error checking or tracking. IP assumes the unreliability of the underlying layers and does its best to get a transmission through to its destination, but with no guarantees.

IP transports data in packets called datagrams, each of which is transported separately. Datagrams can travel along different routes and can arrive out of sequence or be duplicated. IP does not keep track of the routes and has no facility for reordering datagrams once they arrive at their destination.

The limited functionality of IP should not be considered a weakness, however. IP provides bare-bones transmission functions that free the user to add only those facilities necessary for a given application and thereby allows for maximum efficiency. IP is discussed in Chapter 20.

Address Resolution Protocol:

The Address Resolution Protocol (ARP) is used to associate a logical address with a physical address. On a typical physical network, such as a LAN, each device on a link is identified by a physical or station address, usually imprinted on the network interface card (NIC). ARP is used to find the physical address of the node when its Internet address is known. ARP is discussed in Chapter 21.

Reverse Address Resolution Protocol

the Reverse Address Resolution Protocol (RARP) allows a host to discover its Internet address when it knows only its physical address. It is used when a computer is connected to a network for the first time or when a diskless computer is booted. We discuss RARP in Chapter 21.

Internet Control Message Protocol:

The Internet Control Message Protocol (ICMP) is a mechanism used by hosts and gateways to send notification of datagram problems back to the sender. ICMP sends query and error reporting messages. We discuss ICMP in Chapter 21.

Internet Group Message Protocol:

The Internet Group Message Protocol (IGMP) is used to facilitate the simultaneous transmission of a message to a group of recipients. We discuss IGMP in Chapter 22.

Transport Layer:

Traditionally the transport layer was represented in TCP/IP by two protocols: TCP and UDP. IP is a host-to-host protocol, meaning that it can deliver a packet from one physical device to another. UDP and TCP are transport level protocols responsible for delivery of a message from a process (running program) to another process. A new transport layer protocol, SCTP, has been devised to meet the needs of some newer applications.

User Datagram Protocol:

The User Datagram Protocol (UDP) is the simpler of the two standard TCPIIP transport protocols. It is a process-to-process protocol that adds only port addresses, checksum error control, and length information to the data from the upper layer. UDP is discussed in Chapter 23.

Transmission Control Protocol:

The Transmission Control Protocol (TCP) provides full transport-layer services to applications. TCP is a reliable stream transport protocol. The term stream, in this context, means connection-oriented: A connection must be established between both ends of a transmission before either can transmit data.

At the sending end of each transmission, TCP divides a stream of data into smaller units called segments. Each segment includes a sequence number for reordering after receipt, together with an acknowledgment number for the segments received. Segments are carried across the internet inside of IP datagrams. At the receiving end, TCP collects each datagram as it comes in and reorders the transmission based on sequence numbers. TCP is discussed in Chapter 23.

Stream Control Transmission Protocol:

The Stream Control Transmission Protocol (SCTP) provides support for newer applications such as voice over the Internet. It is a transport layer protocol that combines the best features of UDP and TCP. We discuss SCTP in Chapter 23.

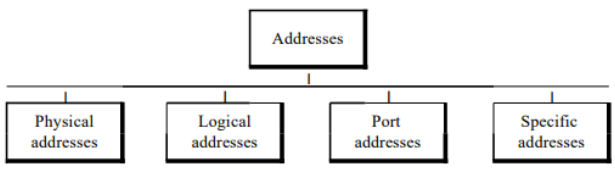
Application Layer:

The application layer in TCPIIP is equivalent to the combined session, presentation, and application layers in the OSI model Many protocols are defined at this layer. We cover many of the standard protocols in later chapters.

Addressing introduction:

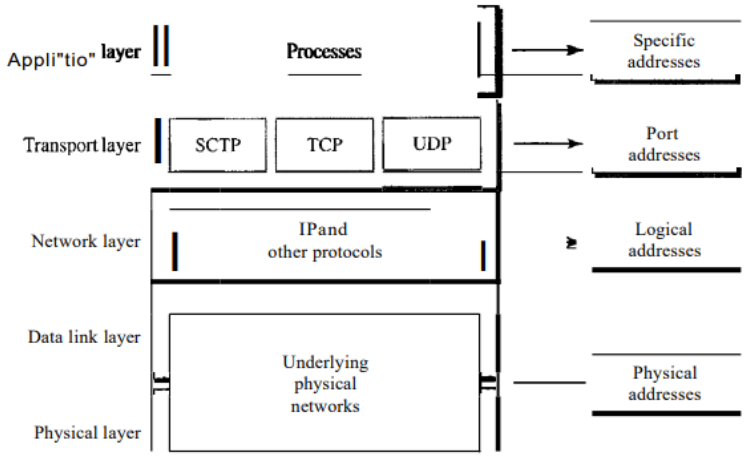
Four levels of addresses are used in an internet employing the TCP/IP protocols: physical (link) addresses, logical (IP) addresses, port addresses, and specific addresses (see Figure 2.17).

Figure 2.17 *Addresses in TCPIIP*



Each address is related to a specific layer in the TCPIIP architecture, as shown in Figure 2.18.

Figure 2.18 *Relationship of layers and addresses in TCPIIP*



Physical Addresses:

The physical address, also known as the link address, is the address of a node as defined by its LAN or WAN. It is included in the frame used by the data link layer. It is the lowest-level address.

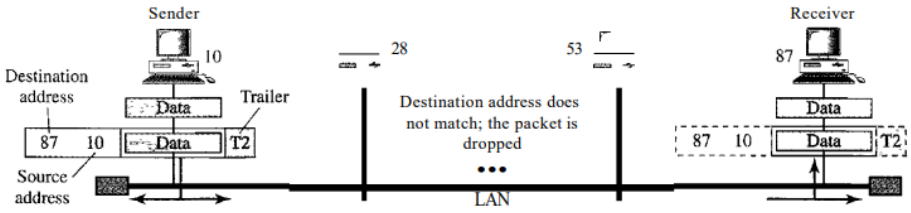
The physical addresses have authority over the network (LAN or WAN). The size and format of these addresses vary depending on the network. For example, Ethernet uses a 6-byte (48-bit) physical address that is imprinted on the network interface card (NIC). Local Talk (Apple), however, has a I-byte dynamic address that changes each time the station comes up.

Example 2.1

In Figure 2.19 a node with physical address 10 sends a frame to a node with physical address 87. The two nodes are connected by a link (bus topology LAN). At the data link layer, this frame contains physical (link) addresses in the header. These are the only addresses needed. The rest of the header contains other information needed at this level. The trailer usually contains extra bits needed for error detection. As the figure shows, the computer with physical address lO is the sender, and the computer with physical address 87 is the receiver. The data link layer at the sender receives data from an upper layer. It encapsulates the data in a frame, adding a header and a trailer. The header, among other pieces of information, carries the receiver and the sender physical (link) addresses. Note that in most data link protocols, the destination address, 87 in this case, comes before the source address (10 in this case).

We have shown a bus topology for an isolated LAN. In a bus topology, the frame is propagated in both directions (left and right). The frame propagated to the left dies when it reaches the end of the cable if the cable end is terminated appropriately. The frame propagated to the right is sent to every station on the network.

Figure 2.19 *Physical addresses*



Each station with a physical addresses other than 87 drops the frame because the destination address in the frame does not match its own physical address. The intended destination computer, however, finds a match between the destination address in the frame and its own physical address. The frame is checked, the header and trailer are dropped, and the data part is decapsulated and delivered to the upper layer.

Example 2.2

As we will see in Chapter 13, most local-area networks use a 48-bit (6-byte) physical address written as 12 hexadecimal digits; every byte (2 hexadecimal digits) is separated by a colon, as shown below:

07:01:02:01 :2C :4B

A 6-byte (12 hexadecimal digits) physical address

Logical Addresses

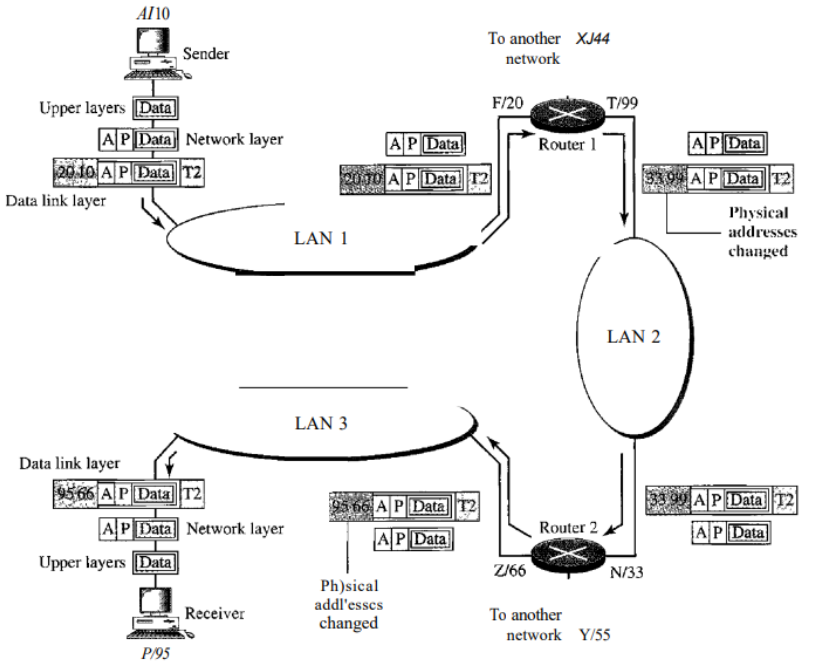
Logical addresses are necessary for universal communications that are independent of underlying physical networks. Physical addresses are not adequate in an internetwork environment where different networks can have different address formats. A universal addressing system is needed in which each host can be identified uniquely, regardless of the underlying physical network.

The logical addresses are designed for this purpose. A logical address in the Internet is currently a 32-bit address that can uniquely define a host connected to the Internet. No two publicly addressed and visible hosts on the Internet can have the same IP address.

Example 2.3

Figure 2.20 shows a part of an internet with two routers connecting three LANs. Each device (computer or router) has a pair of addresses (logical and physical) for each connection. In this case, each computer is connected to only one link and therefore has only one pair of addresses. Each router, however, is connected to three networks (only two are shown in the figure). So each router has three pairs of addresses, one for each connection. Although it may obvious that each router must have a separate physical address for each connection, it may not be obvious why it needs a logical address for each connection. We discuss these issues in Chapter 22 when we discuss routing.

Figure 2.20 *IP addresses*



The computer with logical address A and physical address 10 needs to send a packet to the computer with logical address P and physical address 95. We use letters to show the logical addresses and numbers for physical addresses, but note that both are actually numbers, as we will see later in the chapter.

The sender encapsulates its data in a packet at the network layer and adds two logical addresses (A and P). Note that in most protocols, the logical source address comes before the logical destination address (contrary to the order of physical addresses). The network layer, however, needs to find the physical address of the next hop before the packet can be delivered. The network layer consults its routing table (see Chapter 22) and finds the logical address of the next hop (router I) to be F. The ARP discussed previously finds the physical address of router 1 that corresponds to the logical address of 20. Now the network layer passes this address to the data link layer, which in tum, encapsulates the packet with physical destination address 20 and physical source address 10.

The frame is received by every device on LAN 1, but is discarded by all except router 1, which finds that the destination physical address in the frame matches with its own physical address. The router decapsulates the packet from the frame to read the logical destination address P. Since the logical destination address does not match the router's logical address, the router knows that the packet needs to be forwarded. The router consults its routing table and ARP to find the physical destination address of the next hop (router 2), creates a new frame, encapsulates the packet, and sends it to router 2.

Note the physical addresses in the frame. The source physical address changes from 10 to 99. The destination physical address changes from 20 (router 1 physical address) to 33 (router 2 physical address). The logical source and destination addresses must remain the same; otherwise the packet will be lost.

At router 2 we have a similar scenario. The physical addresses are changed, and a new frame is sent to the destination computer. When the frame reaches the destination, the packet is decapsulated. The destination logical address P matches the logical address of the computer. The data are decapsulated from the packet and delivered to the upper layer. Note that although physical addresses will change from hop to hop, logical addresses remain the same from the source to destination. There are some exceptions to this rule that we discover later in the book.

The physical addresses will change from hop to hop,
but the logical addresses usually remain the same.

Port Addresses :

The IP address and the physical address are necessary for a quantity of data to travel from a source to the destination host. However, arrival at the destination host is not the final objective of data communications on the Internet. A system that sends nothing but data from one computer to another is not complete. Today, computers are devices that can run multiple processes at the same time. The end objective of Internet communication is a process communicating with another process. For example, computer A can communicate with computer C by using TELNET. At the same time, computer A communicates with computer B by using the File Transfer Protocol (FTP). For these processes to receive data simultaneously, we need a method to label the different processes. In other words, they need addresses. In the TCPIIP architecture, the label assigned to a process is called a port address. A port address in TCPIIP is 16 bits in length.

Example 2.4

Figure 2.21 shows two computers communicating via the Internet. The sending computer is running three processes at this time with port addresses a, b, and c. The receiving computer is running two processes at this time with port addresses j and k. Process a in the sending computer needs to communicate with process j in the receiving computer. Note that although both computers are using the same application, FTP, for example, the port addresses are different because one is a client program and the other is a server program, as we will see in Chapter 23. To show that data from process a need to be delivered to process j, and not k, the transport layer encapsulates data from the application layer in a packet and adds two port addresses (a and j), source and destination. The packet from the transport layer is then encapsulated in another packet at the network layer with logical source and destination addresses (A and P). Finally, this packet is encapsulated in a frame with the physical source and destination addresses of the next hop. We have not shown the physical addresses because they change from hop to hop inside the cloud designated as the Internet. Note that although physical addresses change from hop to hop, logical and port addresses remain the same from the source to destination. There are some exceptions to this rule that we discuss later in the book.

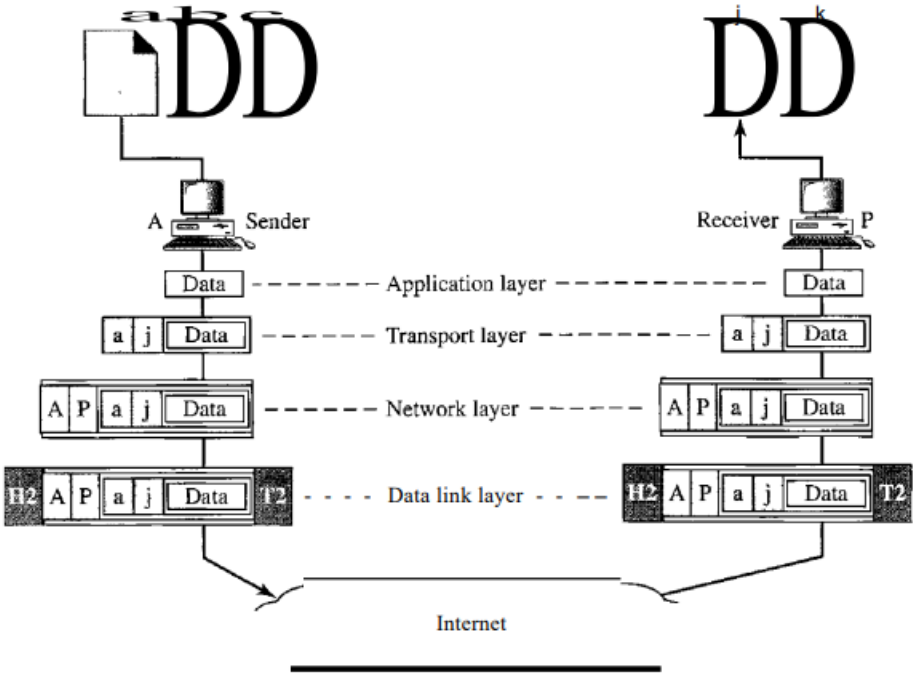
Example 2.5

As we will see in Chapter 23, a port address is a 16-bit address represented by one decimal number as shown

753

A 16-bit port address represented as one single number

Figure 2.21 *Port addresses*



The physical addresses change from hop to hop, but the logical and port addresses usually remain the same.

Specific Addresses:

Some applications have user-friendly addresses that are designed for that specific address. Examples include the e-mail address (for example, forouzan@fhda.edu) and the Universal Resource Locator (URL) (for example, www.mhhe.com). The first defines the recipient of an e-mail (see Chapter 26); the second is used to find a document on the World Wide Web (see Chapter 27). These addresses, however, get changed to the corresponding port and logical addresses by the sending computer, as we will see in Chapter 25.

Wireless Links and Network Characteristics

Let's begin by considering a simple wired network, say a home network, with a wired Ethernet switch (see Section 5.4) interconnecting the hosts. If we replace the wired Ethernet with a wireless 802.11 network, a wireless network interface would replace the host's wired Ethernet interface, and an access point would replace the Ethernet switch, but virtually no changes would be needed at the network layer or above. This suggests that we focus our attention on the link layer when looking for important differences between wired and wireless networks. Indeed, we can find a number of important differences between a wired link and a wireless link:

• Decreasing signal strength. Electromagnetic radiation attenuates as it passes through matter (e.g., a radio signal passing through a wall). Even in free space, the signal will disperse, resulting in decreased signal strength (sometimes referred to as **path loss**) as the distance between sender and receiver increases.

• Interference from other sources. Radio sources transmitting in the same frequency band will interfere with each other. For example, 2.4 GHz wireless phones and 802.11b wireless LANs transmit in the same frequency band. Thus, the 802.11b wireless LAN user talking on a 2.4 GHz wireless phone can expect that neither the network nor the phone will perform particularly well. In addition to interference from transmitting sources, electromagnetic noise within the environment (e.g., a nearby motor, a microwave) can result in interference.

• Multipath propagation. **Multipath propagation** occurs when portions of the electromagnetic wave reflect off objects and the ground, taking paths of different lengths between a sender and receiver. This results in the blurring of the received signal at the receiver. Moving objects between the sender and receiver can cause multipath propagation to change over time.

For a detailed discussion of wireless channel characteristics, models, and measurements, see [Anderson 1995].

The discussion above suggests that bit errors will be more common in wireless links than in wired links. For this reason, it is perhaps not surprising that wireless link protocols (such as the 802.11 protocol we'll examine in the following section) employ not only powerful CRC error detection codes, but also link-level reliable data-transfer protocols that retransmit corrupted frames.

Having considered the impairments that can occur on a wireless channel, let's next turn our attention to the host receiving the wireless signal. This host receives an electromagnetic signal that is a combination of a degraded form of the original signal transmitted by the sender (degraded due to the attenuation and multipath propagation effects that we discussed above, among others) and background noise in the environment. The **signal-to-noise ratio** (SNR) is a relative measure of the strength of the received signal (i.e., the information being transmitted) and this noise. The SNR is typically measured in units of decibels (dB), a unit of measure that some think is used by electrical engineers primarily to confuse computer scientists. The SNR, measured in dB, is twenty times the ratio of the base-10 logarithm of the amplitude of the received signal to the amplitude of the noise. For our purposes here, we need only know that a larger SNR makes it easier for the receiver to extract the transmitted signal from the background noise.

Figure 6.3 (adapted from [Holland 2001]) shows the bit error rate (BER)— roughly speaking, the probability that a transmitted bit is received in error at the receiver—versus the SNR for three different modulation techniques for encoding information for transmission on an idealized wireless channel. The theory of modulation and coding, as well as signal extraction and BER, is well beyond the scope of this text (see [Schwartz 1980] for a discussion of these topics). Nonetheless, Figure 6.3 illustrates several physical-layer characteristics that are important in understanding higher-layer wireless communication protocols:

• For a given modulation scheme, the higher the SNR, the lower the BER. Since a sender can increase the SNR by increasing its transmission power, a sender can decrease the probability that a frame is received in error by increasing its transmission power. Note, however, that there is arguably little

practical gain in increasing the power beyond a certain threshold, say to decrease the BER from 10-12 to 10-13. There are also disadvantages associated with increasing the transmission power: More energy must be expended by the sender (an important concern for battery-powered mobile users), and the sender's transmissions are more likely to interfere with the transmissions of another sender (see Figure 6.4(b)).

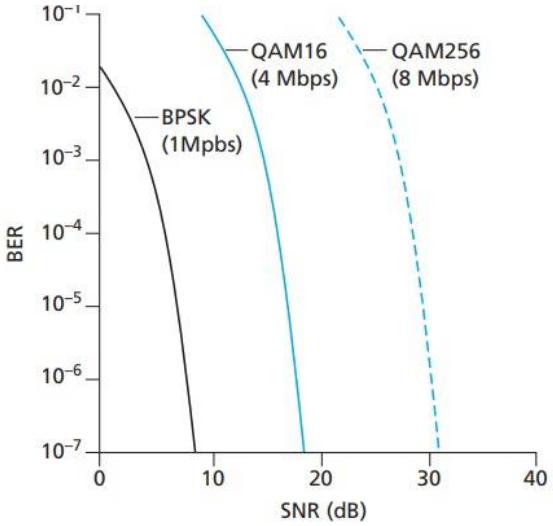


Figure 6.3 ♦ Bit error rate, transmission rate, and SNR

• *For a given SNR, a modulation technique with a higher bit transmission rate (whether in error or not) will have a higher* BER. For example, in Figure 6.3, with an SNR of 10 dB, BPSK modulation with a transmission rate of 1 Mbps has a BER of less than 10-7, while with QAM16 modulation with a transmission rate of 4 Mbps, the BER is 10-1, far too high to be practically useful. However, with an SNR of 20 dB, QAM16 modulation has a transmission rate of 4 Mbps and a BER of 10-7, while BPSK modulation has a transmission rate of only 1 Mbps and a BER that is so low as to be (literally) "off the charts." If one can tolerate a BER of 10-7, the higher transmission rate offered by QAM16 would make it the preferred modulation technique in this situation. These considerations give rise to the final characteristic, described next.

• *Dynamic selection of the physical-layer modulation technique can be used to adapt the modulation technique to channel conditions.* The SNR (and hence the BER) may change as a result of mobility or due to changes in the environment. Adaptive modulation and coding are used in cellular data systems and in the 802.11 WiFi and 3G cellular data networks that we'll study in Sections 6.3 and 6.4. This allows, for example, the selection of a modulation technique that provides the highest transmission rate possible subject to a constraint on the BER, for given channel characteristics.

A higher and time-varying bit error rate is not the only difference between a wired and wireless link. Recall that in the case of wired broadcast links, all nodes receive the transmissions from all other nodes. In the case of wireless links, the situation is not as simple, as shown in Figure 6.4. Suppose that Station A is transmitting to Station B. Suppose also that Station C is transmitting to Station B. With the so called **hidden terminal problem**, physical obstructions in the environment (for example, a mountain or a building) may prevent A and C from hearing each other's transmissions,

even though A's and C's transmissions are indeed interfering at the destination, B. This is shown in Figure 6.4(a). A second scenario that results in undetectable collisions at the receiver results from the fading of a signal's strength as it propagates through the wireless medium. Figure 6.4(b) illustrates the case where A and C are placed such that their signals are not strong enough to detect each other's transmissions, yet their signals are strong enough to interfere with each other at station B. As we'll see in Section 6.3, the hidden terminal problem and fading make multiple access in a wireless network considerably more complex than in a wired network.

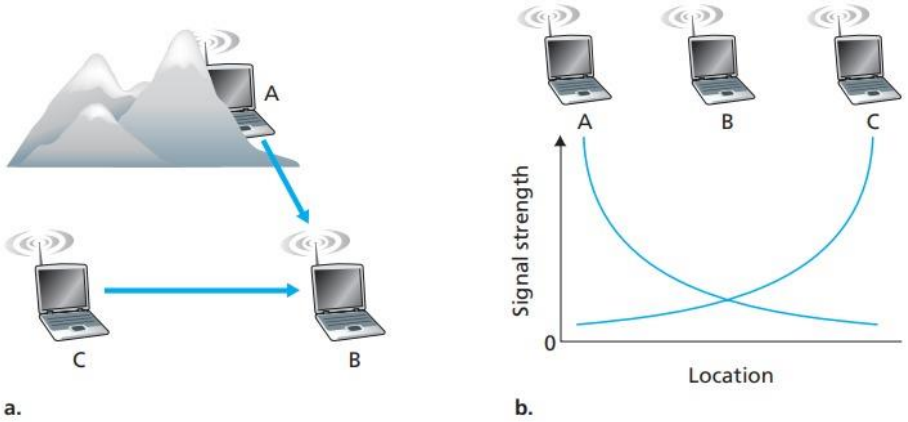


Figure 6.4 ♦ Hidden terminal problem caused by obstacle (a) and fading (b)

WiFi: 802.11 Wireless LANs

Pervasive in the workplace, the home, educational institutions, cafés, airports, and street corners, wireless LANs are now one of the most important access network technologies in the Internet today. Although many technologies and standards for wireless LANs were developed in the 1990s, one particular class of standards has clearly emerged as the winner: the **IEEE 802.11 wireless LAN,** also known as **WiFi.** In this section, we'll take a close look at 802.11 wireless LANs, examining its frame structure, its medium access protocol, and its internetworking of 802.11 LANs with wired Ethernet LANs.

There are several 802.11 standards for wireless LAN technology, including 802.11b, 802.11a, and 802.11g. Table 6.1 summarizes the main characteristics of these standards. 802.11g is by far the most popular technology. A number of dual mode (802.11a/g) and tri-mode (802.11a/b/g) devices are also available.

The three 802.11 standards share many characteristics. They all use the same medium access protocol, CSMA/CA, which we'll discuss shortly. All three use the same frame structure for their link-layer frames as well. All three standards have the ability to reduce their transmission rate in order to reach out over greater distances. And all three standards allow for both "infrastructure mode" and "ad hoc mode," as we'll also shortly discuss. However, as shown in Table 6.1, the three standards have some major differences at the physical layer.

The 802.11b wireless LAN has a data rate of 11 Mbps and operates in the unlicensed frequency band of 2.4–2.485 GHz, competing for frequency spectrum with 2.4 GHz phones and microwave ovens. 802.1la wireless LANs can run at significantly higher bit rates, but do so at higher frequencies.

By operating at a higher frequency, 802.11a LANs have a shorter transmission distance for a given power level and suffer more from multipath propagation. 802.11g LANs, operating in the same lower-frequency band as 802.11b and being backwards compatible with 802.11b (so one can upgrade 802.11b clients incrementally) yet with the higher-speed transmission rates of 802.11a, allows users to have their cake and eat it too.

A relatively new WiFi standard, 802.11n [IEEE 802.11n 2012], uses multiple input multiple-output (MIMO) antennas; i.e., two or more antennas on the sending side and two or more antennas on the receiving side that are transmitting/receiving different signals [Diggavi 2004]. Depending on the modulation scheme used, transmission rates of several hundred megabits per second are possible with 802.11n.

The 802.11 Architecture

Figure 6.7 illustrates the principal components of the 802.11 wireless LAN architecture. The fundamental building block of the 802.11 architecture is the **basic service set (BSS).** A BSS contains one or more wireless stations and a central **base station**, known as an **access point (AP)** in 802.11 parlance. Figure 6.7 shows the AP in each of two BSSs connecting to an interconnection device (such as a switch or router), which in turn leads to the Internet. In a typical home network, there is one AP and one router (typically integrated together as one unit) that connects the BSS to the Internet.

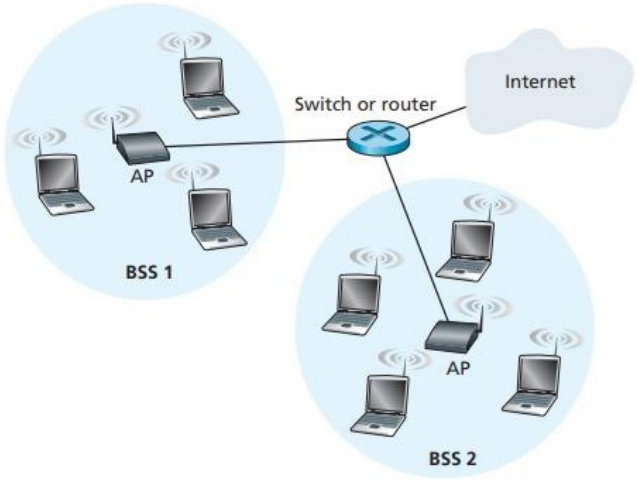


Figure 6.7 ♦ IEEE 802.11 LAN architecture

As with Ethernet devices, each 802.11 wireless station has a 6-byte MAC address that is stored in the firmware of the station's adapter (that is, 802.11 network interface card). Each AP also has a MAC address for its wireless interface. As with Ethernet, these MAC addresses are administered by IEEE and are (in theory) globally unique.

As noted in Section 6.1, wireless LANs that deploy APs are often referred to as infrastructure wireless LANs, with the "infrastructure" being the APs along with the wired Ethernet infrastructure that interconnects the APs and a router. Figure 6.8 shows that IEEE 802.11 stations can also group themselves together to form an ad hoc network—a network with no central control and with no connections to the "outside world." Here, the network is formed "on the fly," by mobile devices that have found themselves in proximity to each other, that have a need to communicate, and that find

no preexisting network infrastructure in their location. An ad hoc network might be formed when people with laptops get together (for example, in a conference room, a train, or a car) and want to exchange data in the absence of a centralized AP. There has been tremendous interest in ad hoc networking, as communicating portable devices continue to proliferate. In this section, though, we'll focus our attention on infrastructure wireless LANs.

Figure 6.8 ♦ An IEEE 802.11 ad hoc network

Channels and Association

In 802.11, each wireless station needs to associate with an AP before it can send or receive network-layer data. Although all of the 802.11 standards use association, we'll discuss this topic specifically in the context of IEEE 802.11b/g.

When a network administrator installs an AP, the administrator assigns a one or two-word Service Set Identifier (SSID) to the access point. (When you "view available networks" in Microsoft Windows XP, for example, a list is displayed showing the SSID of each AP in range.) The administrator must also assign a channel number to the AP. To understand channel numbers, recall that 802.11 operates in the frequency range of 2.4 GHz to 2.485 GHz. Within this 85 MHz band, 802.11 defines 11 partially overlapping channels. Any two channels are non-overlapping if and only if they are separated by four or more channels. In particular, the set of channels 1, 6, and 11 is the only set of three non-overlapping channels. This means that an administrator could create a wireless LAN with an aggregate maximum transmission rate of 33 Mbps by installing three 802.11b APs at the same physical location, assigning channels 1, 6, and 11 to the APs, and interconnecting each of the APs with a switch.

Now that we have a basic understanding of 802.11 channels, let's describe an interesting (and not completely uncommon) situation—that of a **WiFi jungle.** A WiFi jungle is any physical location where a wireless station receives a sufficiently strong signal from two or more APs. For example, in many cafés in New York City, a wireless station can pick up a signal from numerous nearby APs. One of the APs might be managed by the café, while the other APs might be in residential apartments near the café. Each of these APs would likely be located in a different IP subnet and would have been independently assigned a channel.

Now suppose you enter such a WiFi jungle with your portable computer, seeking wireless Internet access and a blueberry muffin. Suppose there are five APs in the WiFi jungle. To gain Internet access, your wireless station needs to join exactly one of the subnets and hence needs to **associate** with exactly one of the APs. Associating means the wireless station creates a virtual wire between itself and the AP. Specifically, only the associated AP will send data frames (that is, frames

containing data, such as a datagram) to your wireless station, and your wireless station will send data frames into the Internet only through the associated AP. But how does your wireless station associate with a particular AP? And more fundamentally, how does your wireless station know which APs, if any, are out there in the jungle?

The 802.11 standard requires that an AP periodically send beacon frames, each of which includes the AP's SSID and MAC address. Your wireless station, knowing that APs are sending out beacon frames, scans the 11 channels, seeking beacon frames from any APs that may be out there (some of which may be transmitting on the same channel—it's a jungle out there!). Having learned about available Aps The 802.11 standard requires that an AP periodically send beacon frames, each of which includes the AP's SSID and MAC address. Your wireless station, knowing that APs are sending out beacon frames, scans the 11 channels, seeking beacon frames from any APs that may be out there (some of which may be transmitting on the same channel—it's a jungle out there!). Having learned about available Aps

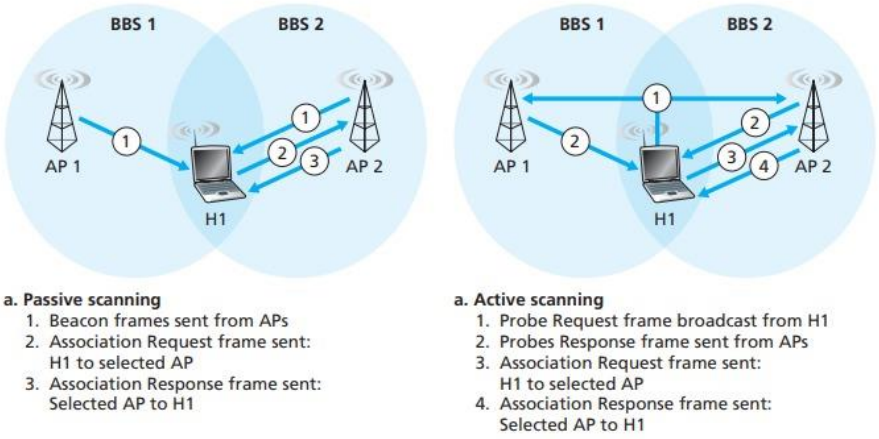
The 802.11 standard does not specify an algorithm for selecting which of the available APs to associate with; that algorithm is left up to the designers of the 802.11 firmware and software in your wireless host. Typically, the host chooses the AP whose beacon frame is received with the highest signal strength. While a high signal strength is good (see, e.g., Figure 6.3), signal strength is not the only AP characteristic that will determine the performance a host receives. In particular, it's possible that the selected AP may have a strong signal, but may be overloaded with other affiliated hosts (that will need to share the wireless bandwidth at that AP), while an unloaded AP is not selected due to a slightly weaker signal. A number of alternative ways of choosing APs have thus recently been proposed [Vasudevan 2005; Nicholson 2006; Sundaresan 2006]. For an interesting and down-to-earth discussion of how signal strength is measured, see [Bardwell 2004].

The process of scanning channels and listening for beacon frames is known as **passive scanning (**see Figure 6.9a). A wireless host can also perform **active scanning**, by broadcasting a probe frame that will be received by all APs within the wireless host's range, as shown in Figure 6.9b. APs respond to the probe request frame with a probe response frame. The wireless host can then choose the AP with which to associate from among the responding APs.

After selecting the AP with which to associate, the wireless host sends an association request frame to the AP, and the AP responds with an association response frame. Note that this second request/response handshake is needed with active scanning, since an AP responding to the initial probe request frame doesn't know which of the (possibly many) responding APs the host will choose to associate with, in much the same way that a DHCP client can choose from among multiple DHCP servers (see Figure 4.21). Once associated with an AP, the host will want to join the subnet (in the IP addressing sense of Section 4.4.2) to which the AP belongs. Thus, the host will typically send a DHCP discovery message (see Figure 4.21) into the subnet via the AP in order to obtain an IP address on the subnet. Once the address is obtained, the rest of the world then views that host simply as another host with an IP address in that subnet.

In order to create an association with a particular AP, the wireless station may be required to authenticate itself to the AP. 802.11 wireless LANs provide a number of alternatives for authentication and access. One approach, used by many companies, is to permit access to a wireless network based on a station's MAC address. A second approach, used by many Internet cafés, employs usernames and passwords. In both cases, the AP typically communicates with an

authentication server, relaying information between the wireless end-point station and the authentication server using a protocol such as RADIUS [RFC 2865] or DIAMETER [RFC 3588]. Separating the authentication server from the AP allows one authentication server to serve many APs, centralizing the (often sensitive) decisions of authentication and access within the single server, and keeping AP costs and complexity low. We'll see in Section 8.8 that the new IEEE 802.11i protocol defining security aspects of the 802.11 protocol family takes precisely this approach.



a. Passive scanning
1. Beacon frames sent from APs
2. Association Request frame sent: H1 to selected AP
3. Association Response frame sent: Selected AP to H1

a. Active scanning
1. Probe Request frame broadcast from H1
2. Probes Response frame sent from APs
3. Association Request frame sent: H1 to selected AP
4. Association Response frame sent: Selected AP to H1

Figure 6.9 ♦ Active and passive scanning for access points