1.1 DATA COMMUNICATIONS

When we communicate, we are sharing information. This sharing can be local or remote. Between individuals, local communication usually occurs face to face, while remote communication takes place over distance. The term *telecommunication,* which includes telephony, telegraphy, and television, means communication at a distance *(tele* is Greek for "far").

The word *data* refers to information presented in whatever form is agreed upon by the parties creating and using the data.

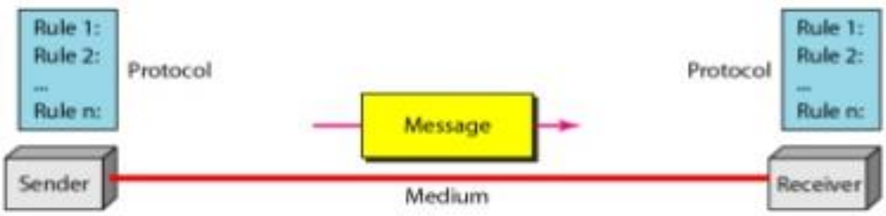
Data communications are the exchange of data between two devices via some form of transmission medium such as a wire cable. For data communications to occur, the communicating devices must be part of a communication system made up of a combination of hardware (physical equipment) and software (programs). The effectiveness of a data communications system depends on four fundamental characteristics: delivery, accuracy, timeliness, and jitter.

- **Delivery.** The system must deliver data to the correct destination. Data must be received by the intended device or user and only by that device or user.
- **Accuracy.** The system must deliver the data accurately. Data that have been altered in transmission and left uncorrected are unusable.
- **Timeliness.** The system must deliver data in a timely manner. Data delivered late are useless. In the case of video and audio, timely delivery means delivering data as they are produced, in the same order that they are produced, and without significant delay. This kind of delivery is called *real-time* transmission.
- **Jitter.** Jitter refers to the variation in the packet arrival time. It is the uneven delay in the delivery of audio or video packets. For example, let us assume that video packets are sent every 30 ms. If some of the packets arrive with 3D-ms delay and others with 40-ms delay, an uneven quality in the video is the result.

Components

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

Figure 1.1 Five components of data communication



1. **Message.** The message is the information (data) to be communicated. Popular forms of information include text, numbers, pictures, audio, and video.
2. **Sender.** The sender is the device that sends the data message. It can be a computer, workstation, telephone handset, video camera, and so on.
3. **Receiver.** The receiver is the device that receives the message. It can be a computer, workstation, telephone handset, television, and so on.
4. **Transmission medium**. The transmission medium is the physical path by which a message travels from sender to receiver. Some examples of transmission media include twisted-pair wire, coaxial cable, fiber-optic cable, and radio waves.
5. **Protocol.** A protocol is a set of rules that govern data communications. It represents an agreement between the communicating devices. Without a protocol, two devices may be connected but not communicating, just as a person speaking French cannot be understood by a person who speaks only Japanese.

Data Representation

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

Text: In data communications, text is represented as a bit pattern, a sequence of bits (0s or 1s). Different sets of bit patterns have been designed to represent text symbols. Each set is called a code, and the process of representing symbols is called coding. Unicode, 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.

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.

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.

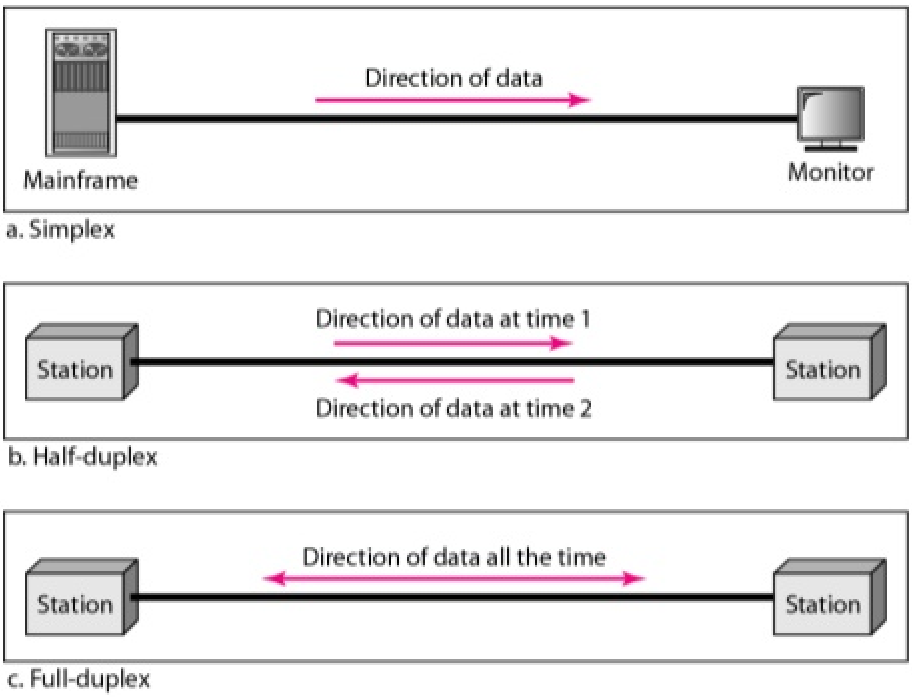
Audio refers to the recording or broadcasting of sound or music. Audio is by nature different from text, numbers, or images. It is continuous, not discrete. Even when we use a microphone to change voice or music to an electric signal, we create a continuous signal.

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.

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

One common example of full-duplex communication is the telephone network. When two people are communicating by a telephone line, both can talk and listen at the same time. The full-duplex mode is used when communication in both directions is required all the time. The capacity of the channel, however, must be divided between the two directions.

1.2 NETWORKS

A network is 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

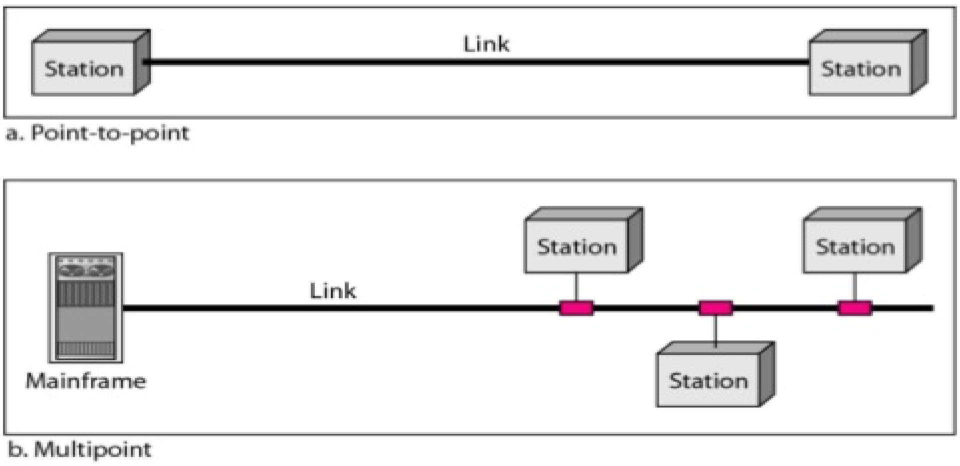
A network is two or more devices connected through links. A link is a communications pathway that transfers data from one device to another. For visualization purposes, it is simplest to imagine any link as a line drawn between two points. For communication to occur, two devices must be connected in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

There are two possible types of connections: point-to-point and multipoint.

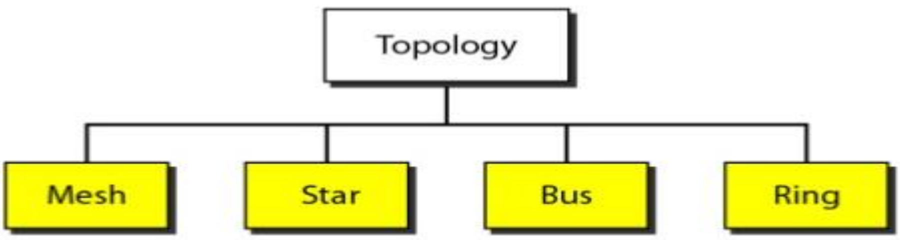
Point-to-Point A point-to-point connection provides a dedicated link between two devices. The entire capacity of the link is reserved for transmission between those two devices. Most point-to-point connections use an actual length of wire or cable to connect the two ends, but other options, such as microwave 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 multi-drop) 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.

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



The term *physical topology* refers to the way in which a network is laid out physically. Two or more devices connect to a link; two or more links form a topology. The topology of a network is the geometric representation of the relationship of all the links and linking devices (usually called nodes) to one another. There are four basic topologies possible: mesh, star, bus, and ring (see Figure 1.4).
Figure 1.4 Categories of topology



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* - 1 nodes, node 2 must be connected to *n-1* nodes, and finally node *n* must be connected to *n* - 1 nodes. We need *n(n* - 1) physical links. However, if each physical link allows communication in both directions (duplex mode), we can divide the number of links by 2. In other words, we can say that in a mesh topology, we need $\frac{n(n-1)}{2}$ duplex –mode links.

Advantages of Mesh topology

1. No data traffic issues as there is a dedicated link between two devices which means the link is only available for those two devices.
2. Mesh topology is reliable and robust as failure of one link doesn't affect other links and the communication between other devices on the network.
3. Mesh topology is secure because there is a point to point link thus unauthorized access is not possible.
4. Fault detection is easy.

Disadvantages of Mesh topology

1. Amount of wires required to connected each system is tedious and headache.
2. Since each device needs to be connected with other devices, number of I/O ports required must be huge.
3. Scalability issues because a device cannot be connected with large number of devices with a dedicated point to point link.

Star Topology

In star topology each device in the network is connected to a central device called hub. Unlike Mesh topology, star topology doesn't allow direct communication between devices, a device must have to communicate through hub. If one device wants to send data to other device, it has to first send the data to hub and then the hub transmit

that data to the designated device.

Advantages of Star topology

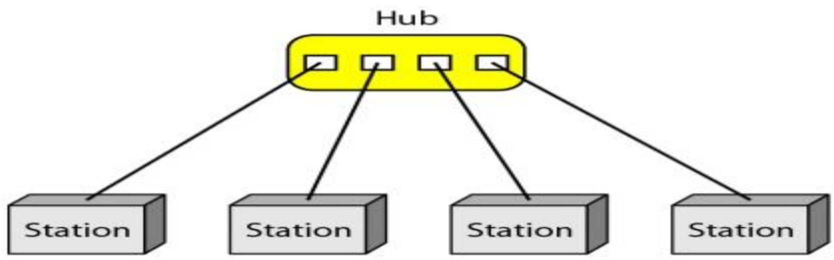
1. Less expensive because each device only need one I/O port and needs to be connected with hub with one link.
2. Easier to install
3. Less amount of cables required because each device needs to be connected with the hub only.
4. Robust, if one link fails, other links will work just fine.
5. Easy fault detection because the link can be easily identified.

Disadvantages of Star topology

1. If hub goes down everything goes down, none of the devices can work without hub.

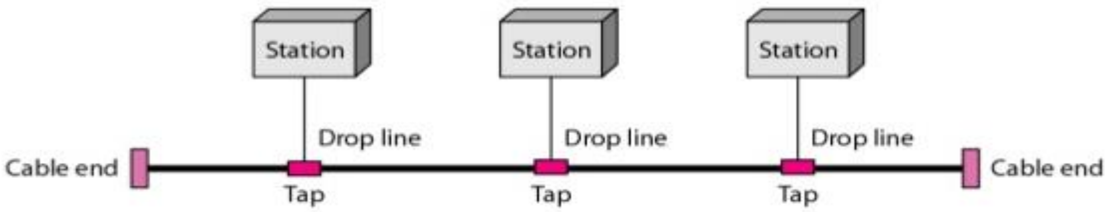
2. Hub requires more resources and regular maintenance because it is the central system of star topology.

Figure 1.6 A star topology connecting four stations



Bus Topology: Mesh and star topologies are examples of point-to-point connections. Bus topology is a multi-point. 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 bus topology

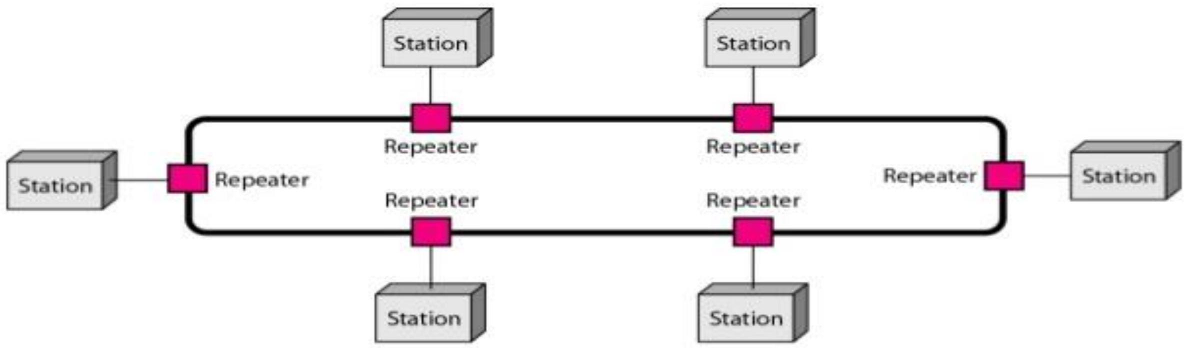
1. Easy installation, each cable needs to be connected with backbone cable.
2. Less cables required than Mesh and star topology

Disadvantages of bus topology

1. Difficultly in fault detection.
2. Not scalable as there is a limit of how many nodes you can connect with backbone cable.

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***



In ring topology each device is connected with the two devices on either side of it. There are two dedicated point to point links a device has with the devices on the either side of it. This structure forms a ring thus it is known as ring topology. If a device wants to send data to another device then it sends the data in one direction, each device in ring topology has a repeater, if the received data is intended for other device then repeater forwards this data until the intended device receives it.

Advantages of Ring Topology

1. Easy to install.
2. Managing is easier as to add or remove a device from the topology only two links are required to be changed.

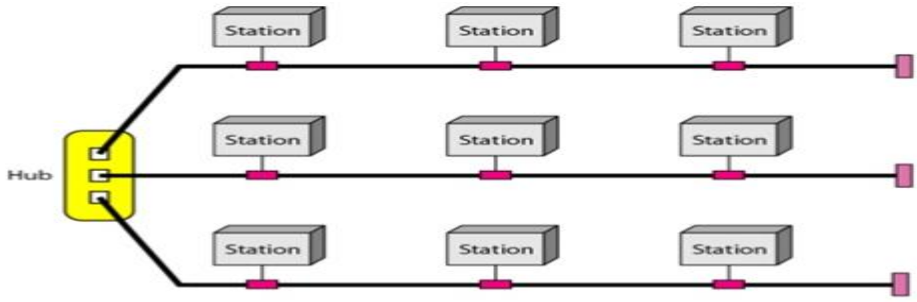
Disadvantages of Ring Topology

1. A link failure can fail the entire network as the signal will not travel forward due to failure.
2. Data traffic issues, since all the data is circulating in a ring.

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.

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



Advantages of Hybrid topology

1. We can choose the topology based on the requirement for example, scalability is our concern then we can use star topology instead of bus technology.
2. Scalable as we can further connect other computer networks with the existing networks with different topologies.

Disadvantages of Hybrid topology

1. Fault detection is difficult.
2. Installation is difficult.
3. Design is complex so maintenance is high thus expensive.

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.

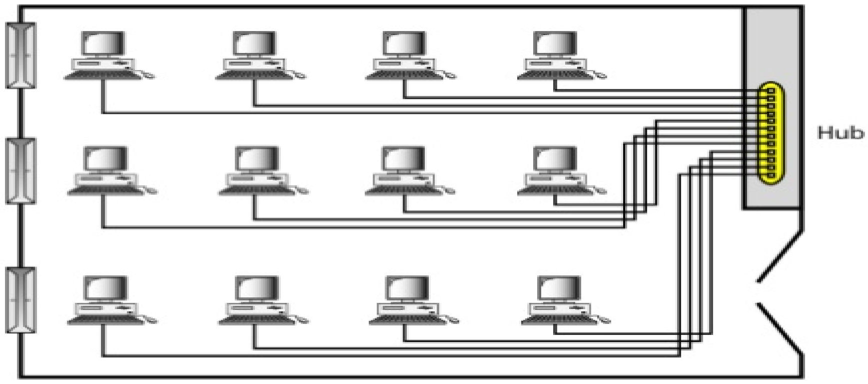
Categories of Networks

There are 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 miles; a WAN can be worldwide. Networks of a size in between are normally referred to as metropolitan area 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 LAN 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.

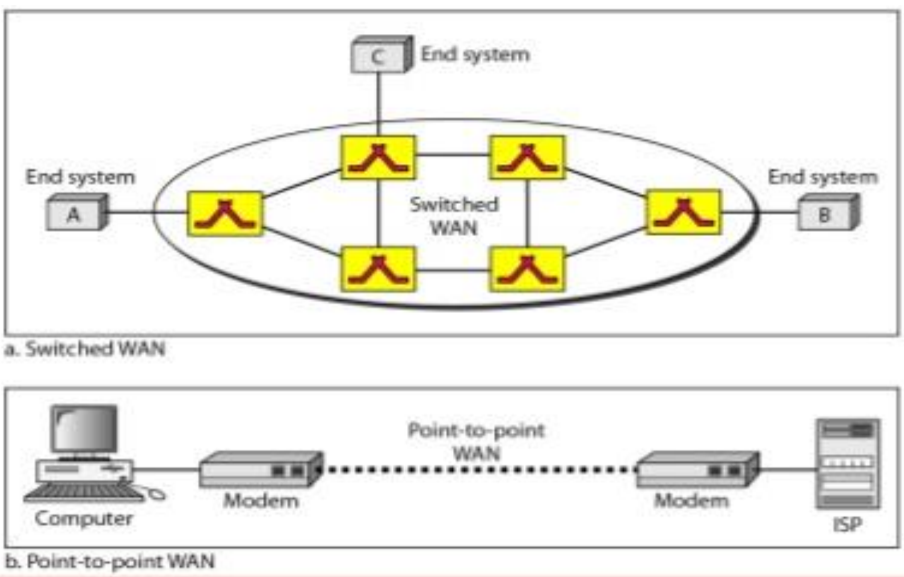
LANs are distinguished from other types of networks by their transmission media and topology. In general, a given LAN will use only one type of transmission medium. The most common LAN topologies are bus, ring, and star. Early LANs had data rates in the 4 to 16 megabits per second (Mbps) range. Today, however, speeds are normally 100 or 1000 Mbps.

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. A WAN connects to the Internet or as simple as a dial-up line that connects a home computer to the Internet. The first is a switched WAN and to the second is 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 (lSP). This type of WAN is often used to provide Internet access.

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



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.

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.

- An **intranet** is a series of networks belonging to one organization (i.e., a company), usually separated by building,cities, region, and or countries/continents. The important thing is that they are all owned by the same organization.

- An **internet** (small “i”) is a series of networks owned by two or more organizations (Microsoft and Google, for example) and set up to communicate with each other.

- The **Internet** (big “I”) is the worldwide network comprising of all other networks interconnected and communicating on the open Web. The IP addresses and DNS names are organized and issued by organization designated by the host country. This is what we refer to, public network that connects the globe.

1.3 PROTOCOLS AND STANDARDS

There are two widely used terms: protocols and standards.

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").

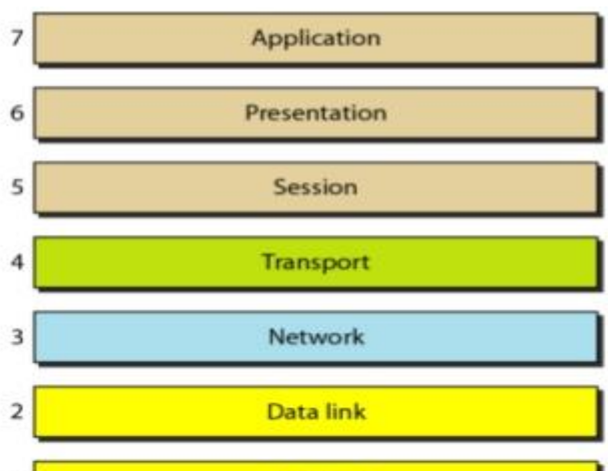
Standards Organizations

- International Organization for Standardization (ISO).
- International Telecommunication Union-Telecommunication Standards Sector (ITU-T).
- American National Standards Institute (ANSI).
- Institute of Electrical and Electronics Engineers (IEEE).
- Electronic Industries Association (EIA).

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.



Layered Architecture

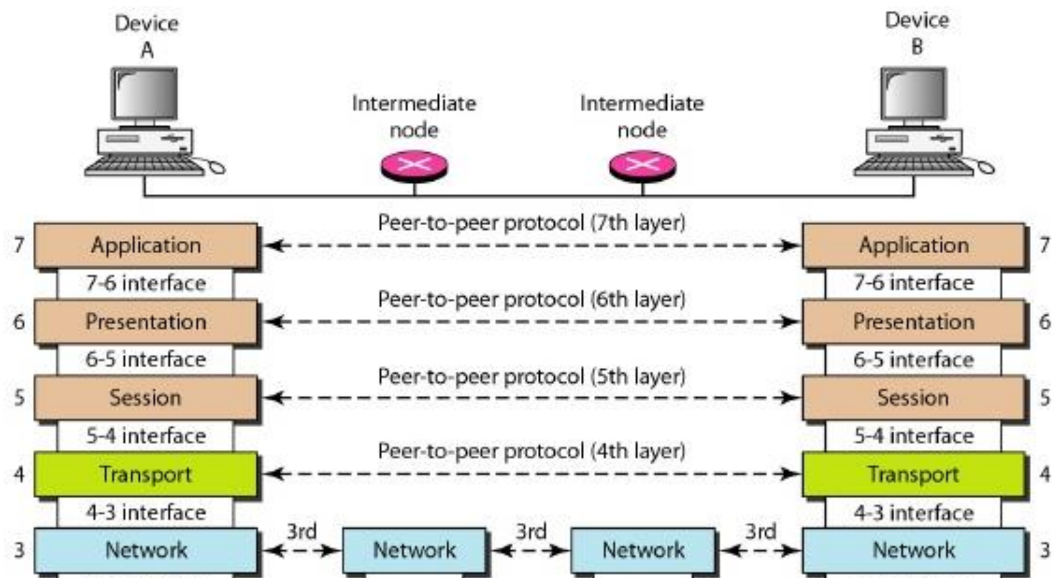
The OSI model is composed of seven ordered layers: physical (layer 1), data link (layer 2), network (layer 3), transport (layer 4), session (layer 5), presentation (layer 6), and application (layer 7). Figure 2.3 shows the layers involved when a message is sent from device A to device B. As the message 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.

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.

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 1, 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.



Interfaces Between 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.

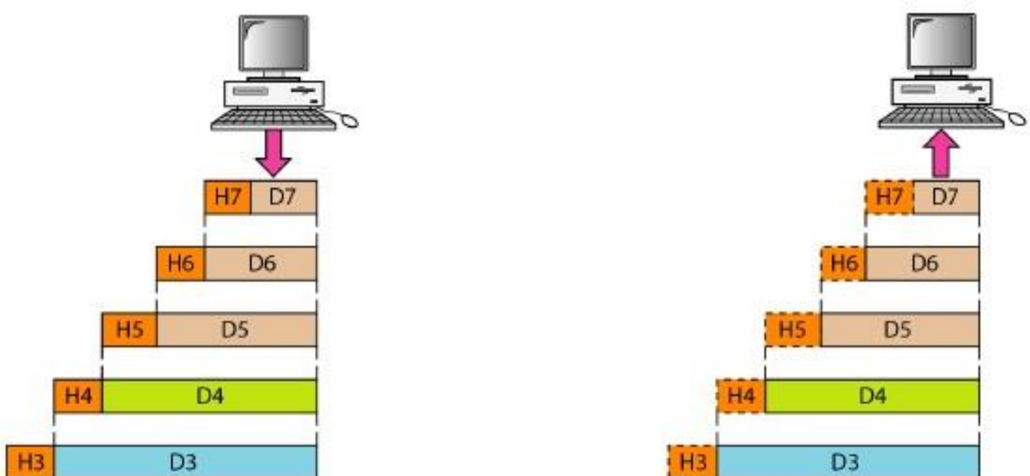
Organization of the Layers

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 OSI 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.

2.3 LAYERS 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.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 0s 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 0s and 1 s are changed to signals).

Data rate. The transmission rate-the number of bits sent each second.

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.

Data Link Layer

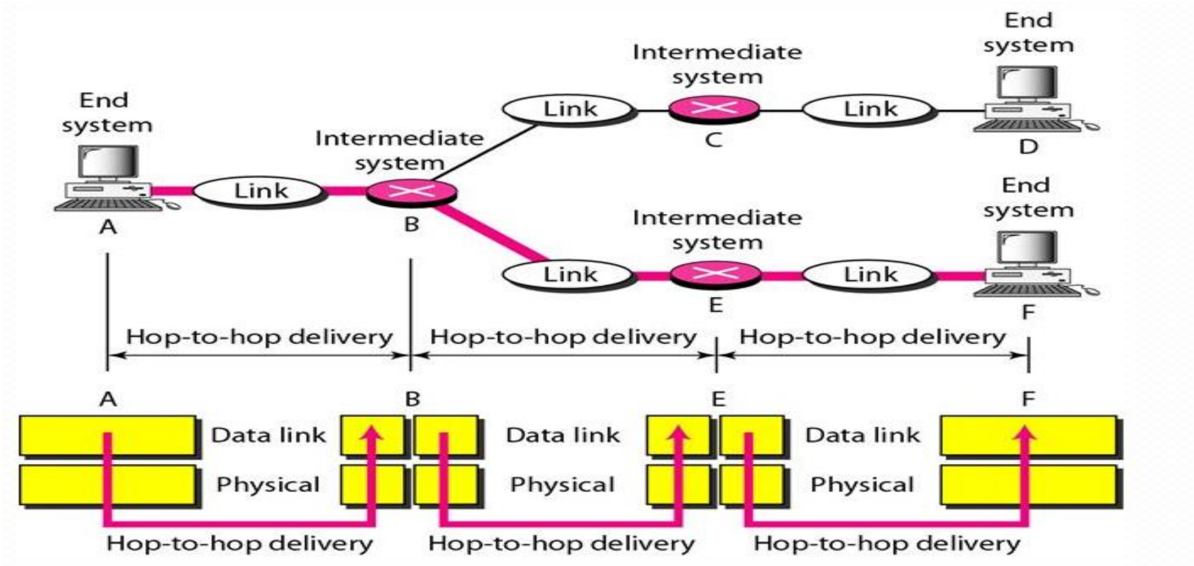
The data link layer transforms the physical layer, a raw transmission facility, to a reliable link. It makes the physical layer appear error-free to the upper layer (network layer). Figure 2.6 shows the relationship of the data link layer to the network and physical layers. The responsibilities of the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

Figure 2.7 *Hop-to-hop delivery*

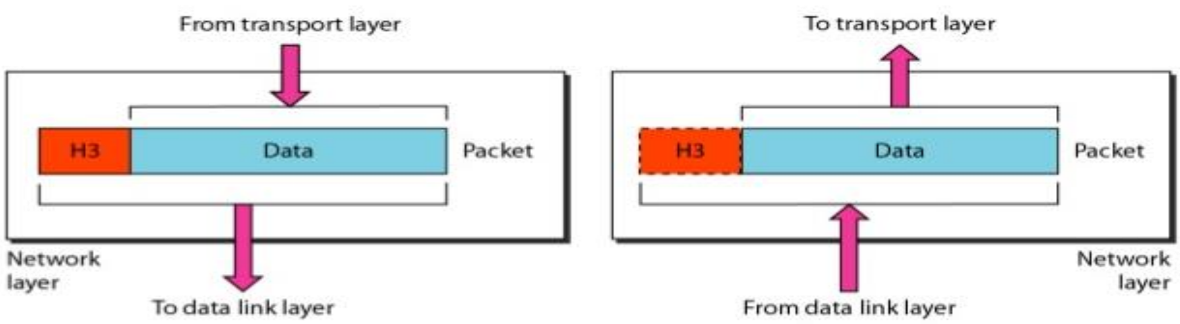


Finally, the data link layer at E sends a new frame to the data link layer at F. Note that the frames that are exchanged between the three nodes have different values in the headers. The frame from A to B has B as the destination address and A as the source address. The frame from B to E has E as the 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.

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.

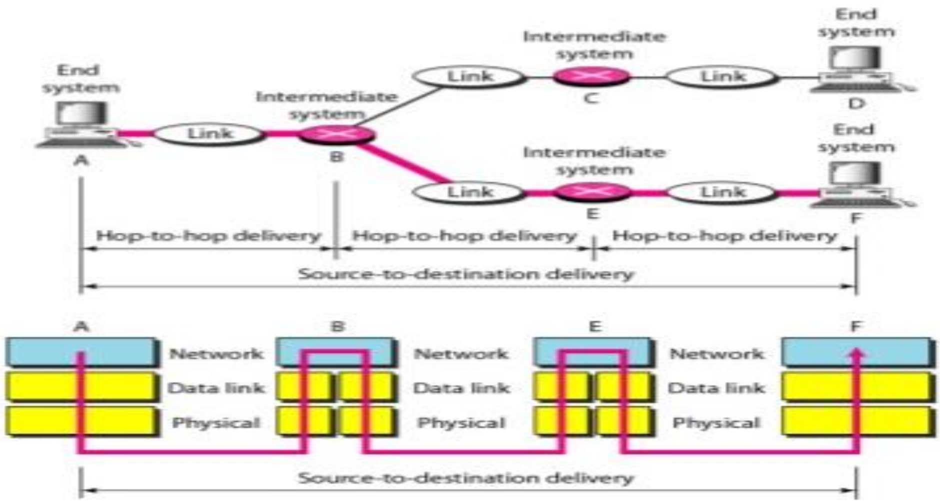
If two systems are connected to the same link, there is usually no need for a network layer. However, if the two systems are attached to different networks (links) with connecting devices between the networks (links), there is often a need for the network layer to accomplish source-to-destination delivery.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



- 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.

- Routing. When independent networks or links are connected to create *internetworks* (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 illustrates end-to-end delivery by the network layer.



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.

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).
- **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.

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.
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.

Presentation Layer

The presentation layer is concerned with the syntax and semantics of the information exchanged between two systems.

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 information 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.

Specific services provided by the application layer include the following:

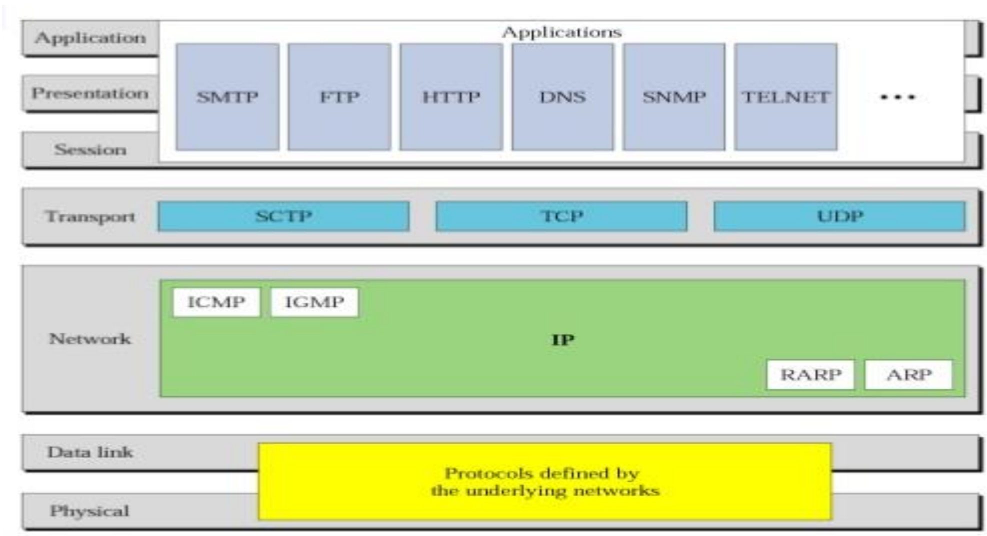
- **Network virtual terminal**. A network virtual terminal is a software version of a physical terminal, and it allows a user to log on to a remote host.

- **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.

2.4 TCP/IP PROTOCOL SUITE

The TCP/IP 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, 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 TCP/IP taking care of part of the duties of the session layer. TCP/IP protocol suite is made of five layers: physical, data link, network, transport, and application.



Physical and Data Link Layers

At the physical and data link layers, *TCP/IP* does not define any specific protocol. It supports all the standard and proprietary protocols. A network in a *TCP/IP* 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.

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.

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.

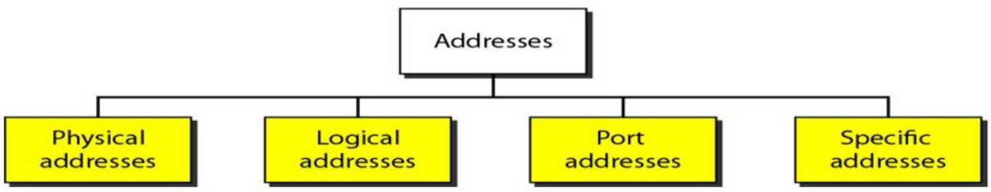
- ***User Datagram Protocol**:* The User Datagram Protocol (UDP) is the simpler of the two standard TCPIIP transport protocols. It is a process-to-process protocol that adds only port addresses, checksum error control, and length information to the data from the upper layer.
- ***Transmission Control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

Application Layer

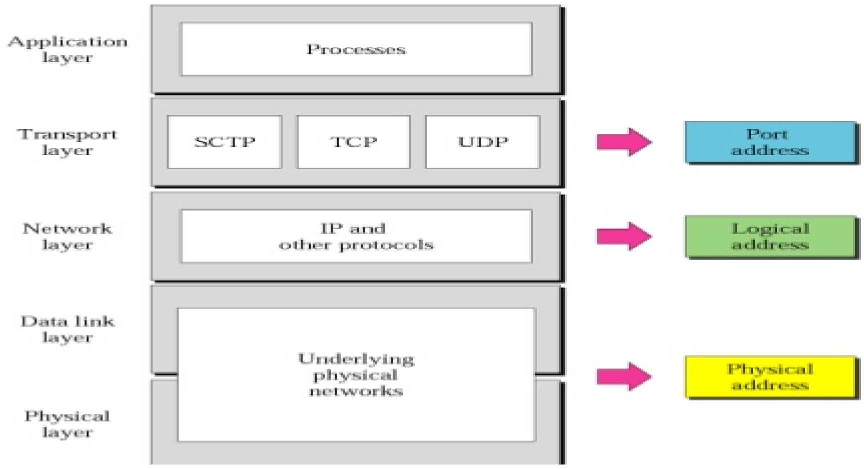
The *application layer* in TCP/IP is equivalent to the combined session, presentation, and application layers in the OSI model.

2.5 ADDRESSING

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.



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



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).

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.

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. 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.

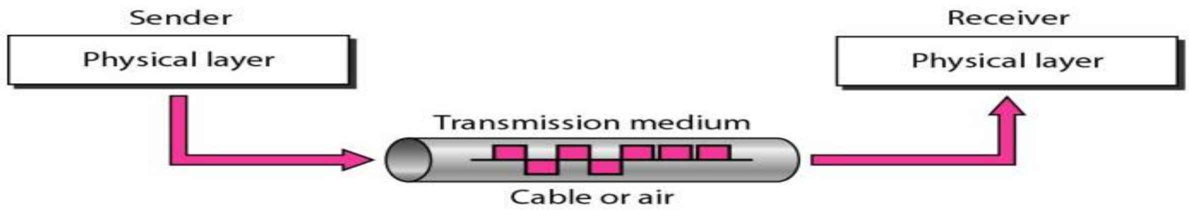
Specific Addresses

Some applications have user-friendly addresses that are designed for that specific address. Examples include the e-mail address (for example, rama@vnrvjiet.in) and the Universal Resource Locator (URL) (for example, www.vnrvjiet.ac.in.).

Transmission Media

A transmission **medium** carries information from a source to a destination. *For example, the transmission medium for two people having a dinner conversation is the air*. In data communications the definition of the information and the transmission medium is more specific. The transmission medium is usually free space, metallic cable, or fiber-optic cable. The information is usually a signal that is the result of a conversion of data from another form. *Figure 7.1 shows the position of transmission media in relation to the physical layer.*

Figure 7.1 Transmission medium and physical layer



The use of long-distance communication using electric signals started with the invention of the telegraph by Morse in the 19th century. Communication by telegraph was slow and dependent on a metallic medium. Extending the range of the human voice became possible when the telephone was invented in 1869. Telephone communication at that time also needed a metallic medium to carry the electric signals that were the result of a conversion from the human voice. The communication was, however, unreliable due to the poor quality of the wires. The lines were often noisy and the technology was unsophisticated.

Wireless communication started in 1895 when Hertz was able to send high frequency signals. Later, Marconi devised a method to send telegraph-type messages over the Atlantic Ocean. Better metallic media have been invented (twisted pair and coaxial cables, for example). The use of optical fibers has increased the data rate incredibly. Free space (air, vacuum, and water) is used more efficiently due to the technologies (such as modulation and multiplexing) . Computers and other telecommunication devices use signals to represent data. These signals are transmitted from one device to another in the form of electromagnetic energy, which is propagated through transmission media.

Electromagnetic energy, a combination of electric and magnetic fields vibrating in relation to each other, includes power, radio waves, infrared light, visible light, ultraviolet light, and X, gamma, and cosmic rays. Each of these constitutes a portion of the electromagnetic spectrum. Not all portions of the spectrum are currently usable for telecommunications, however. The media to harness those that are usable are also limited to a few types.

Some factors need to be considered for designing the transmission media:

- **Bandwidth:** The greater the bandwidth of a medium, the higher the data transmission rate of a signal.
- **Transmission impairment:** When the received signal is not identical to the transmitted one due to the transmission impairment, the quality of the signals will get destroyed due to transmission impairment.
- **Interference:** An interference is defined as the process of disrupting a signal when it travels over a communication medium on the addition of some unwanted signal.

Causes of Transmission Impairment:

- **Attenuation:** Attenuation means the loss of energy, i.e., the strength of the signal decreases with increasing the distance which causes the loss of energy.
- **Distortion:** Distortion occurs when there is a change in the shape of the signal. This type of distortion is examined from different signals having different frequencies. Each frequency component has its own propagation speed, so they reach at a different time which leads to the delay distortion.
- **Noise:** When data is travelled over a transmission medium, some unwanted signal is added to it which creates the noise.

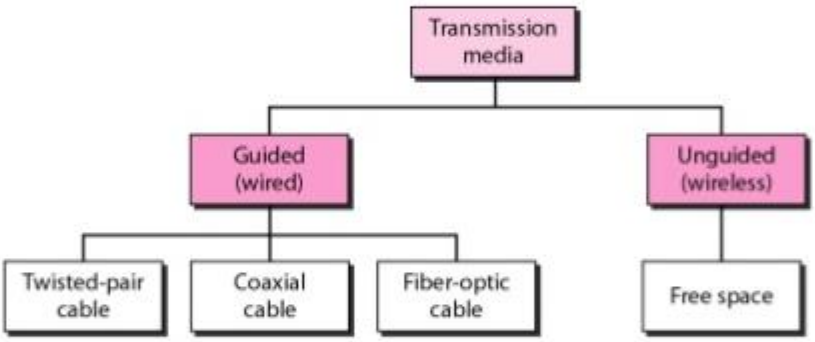
In telecommunications, transmission media can be divided into two broad categories: guided and unguided. Guided media include twisted-pair cable, coaxial cable, and fiber-optic cable. Unguided medium is free space. Figure 7.2 shows this taxonomy.

GUIDED MEDIA

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

Let's first take a closer look at the different types of guided transmission media one at a time.

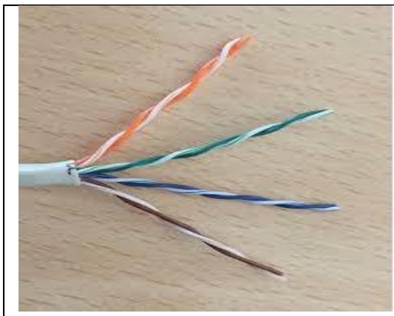
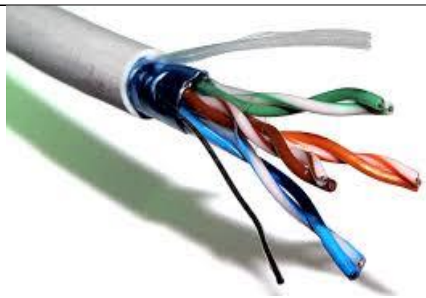
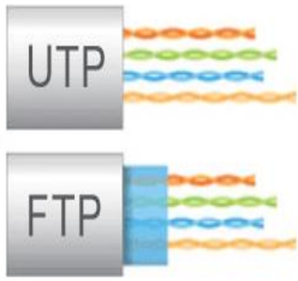
Figure 7.2 ***Classes of transmission media***



1. Twisted Pair Cable

Twisted pair cables have been around for a long time. They were mainly invented for voice transmissions. Twisted pair is a widely used medium in networking because it's lighter, cheaper, more flexible, easy to install, and provides greater speeds than coaxial cables. There are two types of twisted pair cables: the unshielded twisted pair (UTP) and the shielded twisted pair (STP). Let's take a closer look at each of them.

The unshielded twisted pair cable has 4 pairs of copper wires that are present inside a plastic sheath. These wires are twisted to protect them from interference. The only protection available for a UTP cable is a plastic sheath that is thin in size.

		
Unshielded Twisted Pair Cable	Shielded Twisted Pair Cable	UTP and STP Difference

An unshielded twisted pair is widely used in telecommunication. Following are the categories of the unshielded twisted pair cable:

- **Category 1:** Category 1 is used for telephone lines that have low-speed data.
- **Category 2:** It can support up to 4Mbps.
- **Category 3:** It can support up to 16Mbps.
- **Category 4:** It can support up to 20Mbps. Therefore, it can be used for long-distance communication.
- **Category 5:** It can support up to 200Mbps.

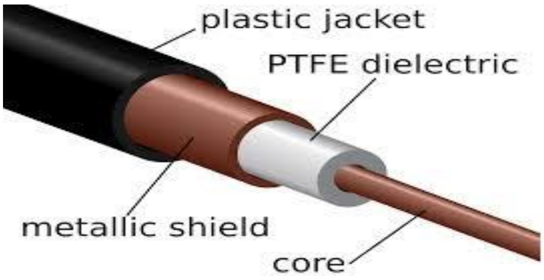
The shielded twisted pair cable is widely used in high-speed networks. The major difference between UTP and shielded twisted pair is that STP makes use of a metallic shield to wrap the wires. This metallic shield prevents interference to a better extent than UTP. These STP cables come with numbering; the higher the numbering, the better the interference prevention. As an example: most computer networks must go with CAT 3 or CAT 5, and nothing less than this.

Coaxial Cables

The coaxial cables have a central copper conductor, surrounded by an insulating layer, a conducting shield, and the outermost plastic sheath. Thus, there are three insulation layers for the inner copper cable. There are two basic modes of data transmission in coaxial cables: baseband mode that has dedicated bandwidth, and broadband mode that has distributed cable bandwidth.

Cable TV and analog televisions mainly use coaxial cables. Coaxial cables have better resistance to cross talk than twisted pair cables. The coaxial cables are used for long distance communication. The most widely used types of coaxial cables are RG-59 and RG-6 (RG stands for 'radio guide'). RG-59 has lesser shielding and is suitable for short cable lengths and cable TV connections.

RG-6 has better insulation than RG-59 and is used for satellite TV and digital signal transmissions for better strength and longer distances.



Coaxial Cable

There are many advantages to coaxial cables, including the following:

- High bandwidth
- Easy and cheap installation
- Better immunity from noise
- Better scaling

However, there are also a number of disadvantages to coaxial cables, which include the following:

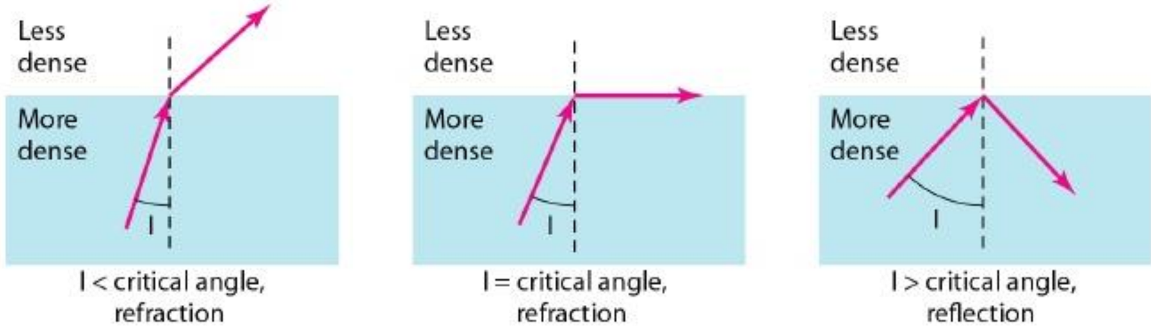
- They're more prone to lightning strikes.
- They cover less distance than fiber optic cables.

Now let's move onto a different type of guided transmission media.

Fiber-Optic Cable

A fiber-optic cable is made of glass or plastic and transmits signals in the form of light. To understand optical fiber, we first need to explore several aspects of the nature of light. Light travels in a straight line as long as it is moving through a single uniform substance. If a ray of light traveling through one substance suddenly enters another substance (of a different density), the ray changes direction. Figure 7.10 shows how a ray of light changes direction when going from a more dense to a less dense substance.

Figure 7.10 ***Bending of light ray***



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

Optical fibers use reflection to guide light through a channel. A glass or plastic core is surrounded by a cladding of less dense glass or plastic.

Propagation Modes

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

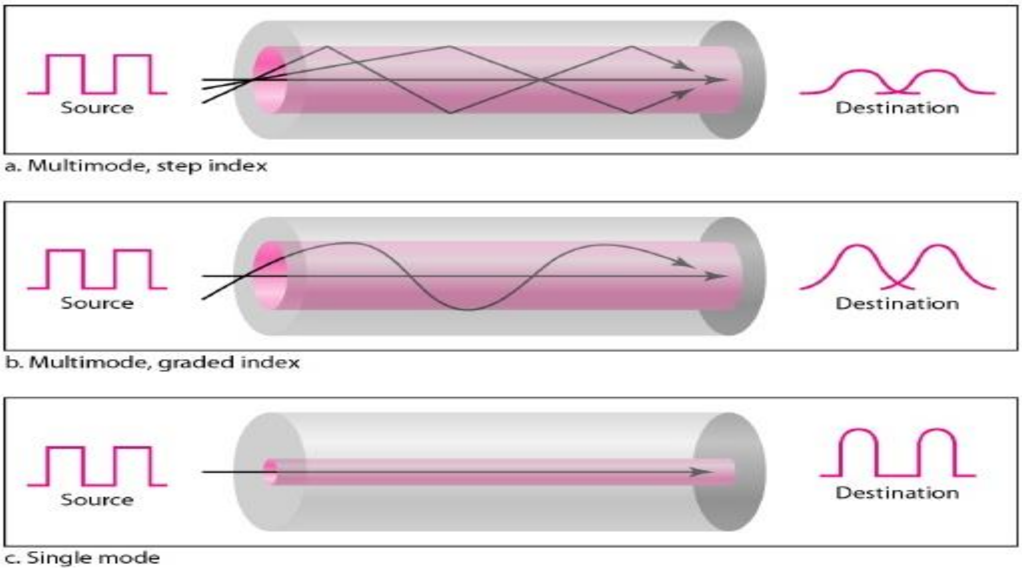
Multimode: Here multiple beams from a light source move through the core in different paths. The beams that move within the cable depends on the structure of the core, as shown in Figure 7.13.

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

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

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

Figure 7.13 Modes



The advantages of optical fibers include the following:

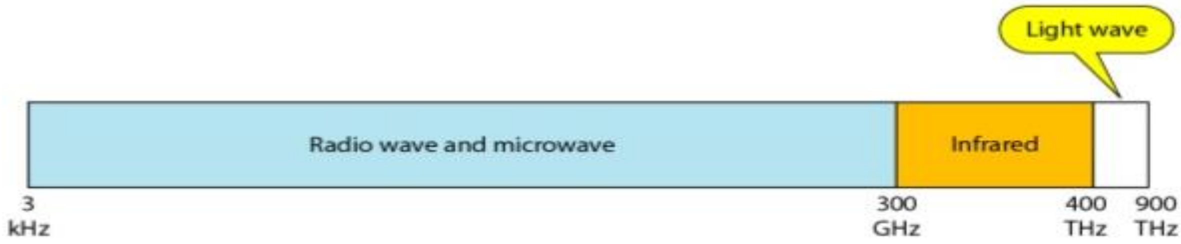
- There is zero interference and covers major cities and countries.
- They have high speed and high bandwidth.
- They're highly secure.

There also are a number of disadvantages, including the following:

- Installation and maintenance are difficult.
- Cabling is costly.
- Retrofitting an existing network is difficult, since optical fibers are incompatible with many types of electronic networking equipment.

UNGUIDED MEDIA: WIRELESS

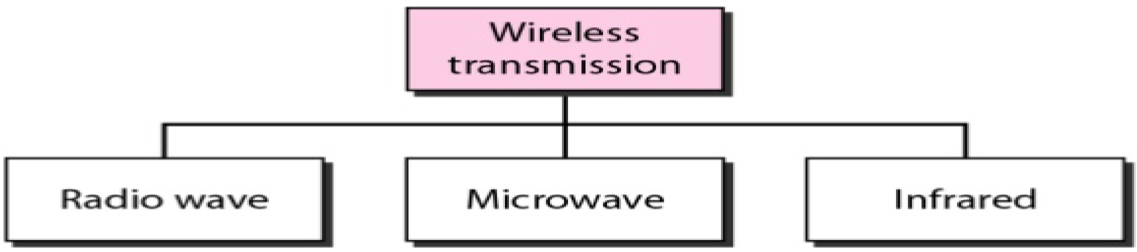
Unguided media transport electromagnetic waves without using a physical conductor. This type of communication is often referred to as wireless communication. Signals are normally broadcast through free space and thus are available to anyone who has a device capable of receiving them.
Figure 7.17 shows the part of the electromagnetic spectrum, ranging from 3 kHz to 900 THz, used for wireless communication.



Unguided signals can travel from the source to destination in several ways: ground propagation, sky propagation, and line-of-sight propagation.
In ground propagation, radio waves travel through the lowest portion of the atmosphere, hugging the earth. These low-frequency signals emanate in all directions from the transmitting antenna and follow the curvature of the planet. Distance depends on the amount of power in the signal: The greater the power, the greater the distance.

Sky propagation, higher-frequency radio waves radiate upward into the ionosphere (the layer of atmosphere where particles exist as ions) where they are reflected back to earth. This type of transmission allows for greater distances with lower output power.

In line-or-sight propagation, very high-frequency signals are transmitted in straight lines directly from antenna to antenna. Antennas must be directional, facing each other.



Radio Waves

Although there is no clear-cut demarcation between radio waves and microwaves, electromagnetic waves ranging in frequencies between **3 kHz and 1 GHz are normally called radio waves**; waves ranging in frequencies **between 1GHz and 300 GHz are called microwaves.** However, the behavior of the waves, rather than the frequencies, is a better criterion for classification.

Radio waves are omnidirectional means they are propagated in all directions. This means that the sending and receiving antennas do not have to be aligned. A sending antenna sends waves that can be received by any receiving antenna. The omnidirectional property has a disadvantage, where the radio waves transmitted by one antenna are susceptible to interference by another antenna that may send signals using the same frequency or band.

Applications

The omnidirectional characteristics of radio waves make them useful for multicasting, in which there is one sender but many receivers. AM and FM radio, television etc.,.

Microwaves

Microwaves are unidirectional. When an antenna transmits microwave waves, they can be narrowly focused. This means that the sending and receiving antennas need to be aligned. The unidirectional property has an obvious advantage. A pair of antennas can be aligned without interfering with another pair of aligned antennas. The following describes some characteristics of microwave propagation:

- Microwave propagation is line-of-sight. Since the towers with the mounted antennas need to be in direct sight of each other, towers that are far apart need to be very tall. Repeaters are often needed for long distance communication.
- Very high-frequency microwaves cannot penetrate walls. This characteristic can be a disadvantage if receivers are inside buildings.
- The microwave band is relatively wide, almost 299 GHz. Therefore wider sub-bands can be assigned, and a high data rate is possible
- Use of certain portions of the band requires permission from authorities.

Microwaves need unidirectional antennas that send out signals in one direction. Two types of antennas are used for microwave communications: the parabolic dish and the horn.

Applications

Microwaves, due to their unidirectional properties, are very useful when unicast (one-to-one) communication is needed between the sender and the receiver. They are used in cellular phones.

Infrared

Infrared waves, with frequencies from 300 GHz to 400 THz (wavelengths from 1 mm to 770 nm), can be used for short-range communication. Infrared waves, having high frequencies, cannot penetrate walls. This advantageous characteristic prevents interference between one system and another; a short-range communication system in one room cannot be affected by another system in the next room. When we use our infrared remote control, we do not interfere with the use of the remote by our neighbors. However, this same characteristic makes infrared signals useless for long-range communication.

Multiplexing

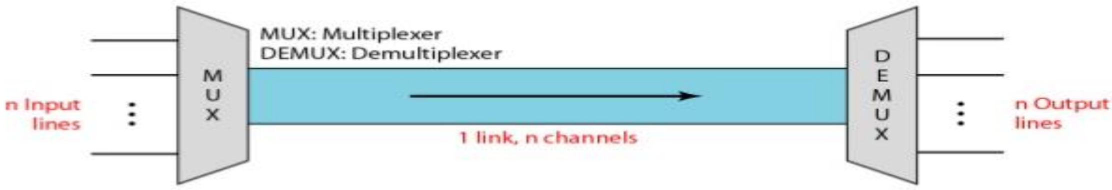
In real life, we have links with limited bandwidths. The effective utilization of these bandwidths is one of the main challenges of electronic communications. There is a need to combine several low-bandwidth channels to make use of one channel with a larger bandwidth. Sometimes there is a need to expand the bandwidth of a channel to achieve goals such as privacy and anti-jamming. There are two broad categories of bandwidth utilization: multiplexing and spreading. In multiplexing, the goal is efficiency. In spreading, the goals are privacy and anti-jamming, the bandwidth of the channel is expanded to insert redundancy, which is necessary to achieve these goals.

6.1 MULTIPLEXING

Whenever the bandwidth of a medium linking two devices is greater than the bandwidth needs of the devices, the link can be shared. Multiplexing is the set of techniques that allows the simultaneous transmission of multiple signals across a single data link. As data and telecommunications use increases, so does traffic. This can be accommodated by adding individual links each time a new channel is needed; or higher-bandwidth links are installed to carry multiple signals.

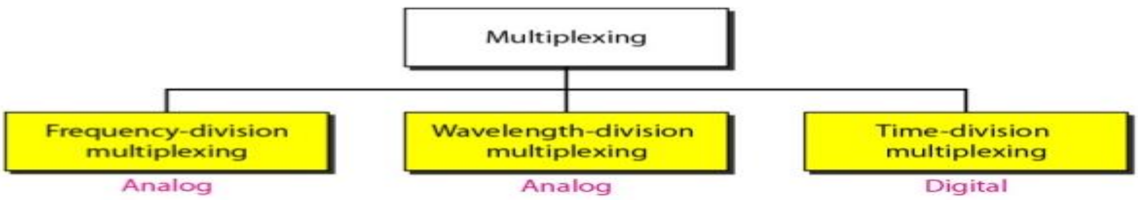
In a multiplexed system, *n* lines share the bandwidth of one link. Figure 6.1 shows the basic format of a multiplexed system. The lines on the left direct their transmission streams to a multiplexer (MUX), which combines them into a single stream (many-to-one). At the receiving end, that stream is fed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

Figure 6.1 Dividing a link into channels



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

Figure 6.2 Categories of multiplexing



Frequency-Division Multiplexing

Frequency-division multiplexing (FDM) is an analog technique that can be applied when the bandwidth of a link (in hertz) is greater than the combined bandwidths of the signals to be transmitted. In FOM, signals generated by each sending device modulate different carrier frequencies. These modulated signals are then combined into a single composite signal that can be transported by the link.

Carrier frequencies are separated by sufficient bandwidth to accommodate the modulated signal. These bandwidth ranges are the channels through which the various signals travel. Channels can be separated by strips of unused bandwidth-guard bands-to prevent signals from overlapping. In addition, carrier frequencies must not interfere with the original data frequencies. Figure 6.3 gives a conceptual view of FDM. In this illustration, the transmission path is divided into three parts, each representing a channel that carries one transmission.

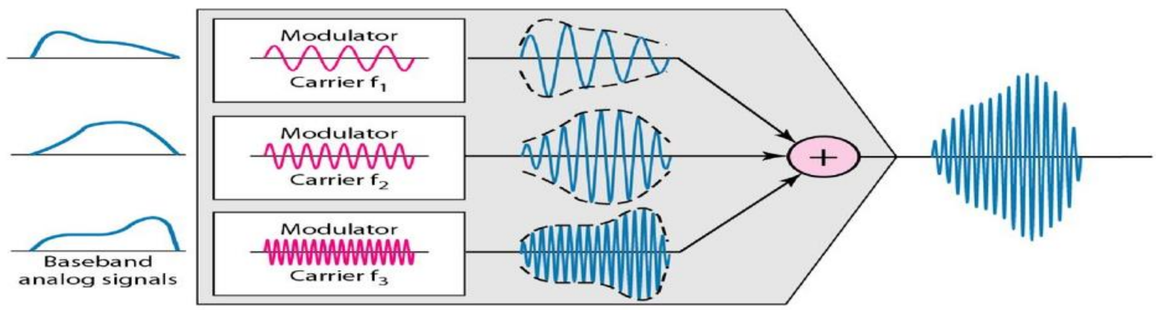
Figure 6.3 Frequency-division multiplexing

FDM is considered to be an analog multiplexing technique; however, this does not mean that FDM cannot be used to combine sources sending digital signals. A digital signal can be converted to an analog signal before FDM is used to multiplex them.

Multiplexing Process

Figure 6.4 is a conceptual illustration of the multiplexing process. Each source generates a signal of a similar frequency range. Inside the multiplexer, these similar signals modulates different carrier frequencies *(f1,f2,f3).* The resulting modulated signals are then combined into a single composite signal that is sent out over a media link that has enough bandwidth to accommodate it.

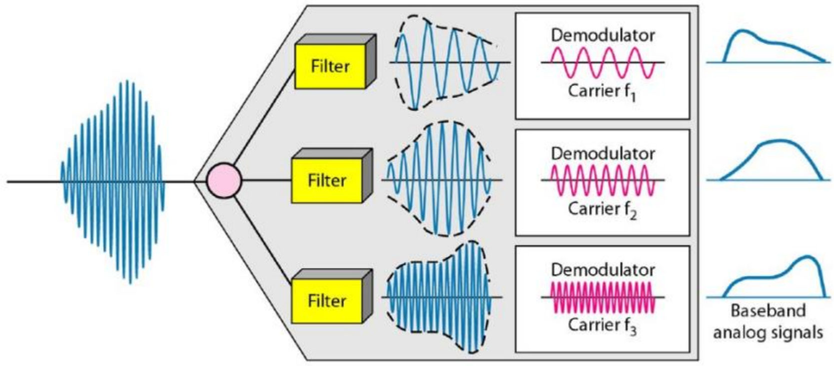
Figure 6.5 FDM demultiplexing example



Demultiplexing Process

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

Figure 6.5 FDM demultiplexing example



Example 6.1

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

Solution

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

Example 6.2

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

Solution

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

Figure 6.6 Example 6.1

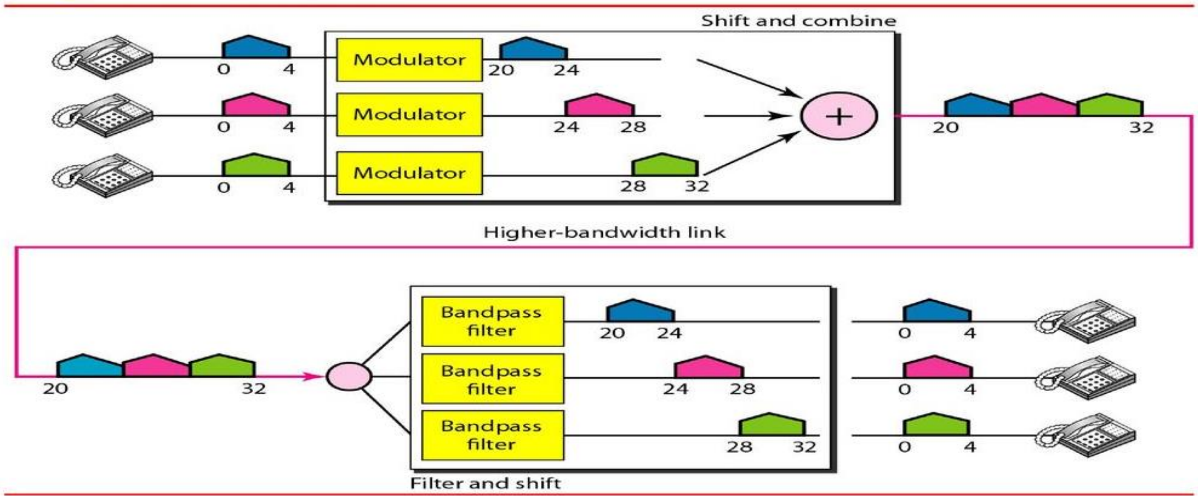
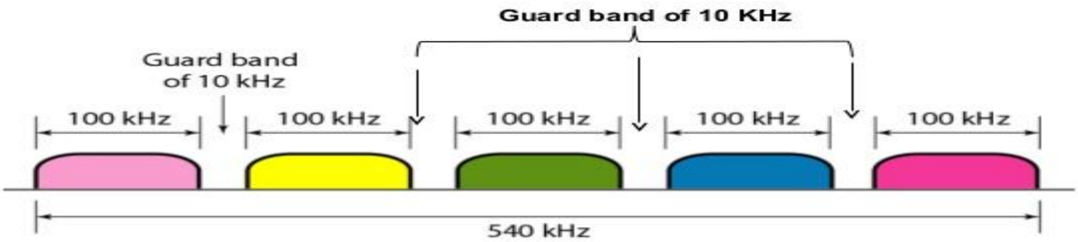


Figure 6.7 Example 6.2



Example 6.3

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

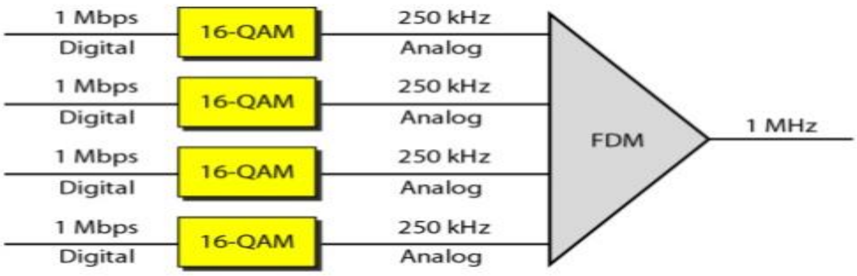
Solution

The satellite channel is analog. We divide it into four channels, each channel having a 2S0-kHz bandwidth. Each digital channel of I Mbps is modulated such that each 4 bits is modulated to 1 Hz. One solution is 16-QAM modulation. Figure 6.8 shows one possible configuration.

The Analog Carrier System

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

Figure 6.8 Example 6.3



Implementation

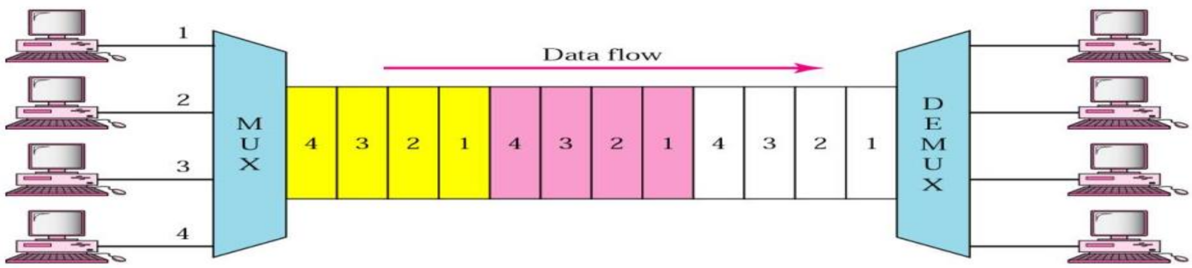
FDM can be implemented very easily. In many cases, such as radio and television broadcasting, there is no need for a physical multiplexer or demultiplexer. As long as the stations agree to send their broadcasts to the air using different carrier frequencies, multiplexing is achieved. In other cases,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

Synchronous Time-Division Multiplexing

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

Note that in Figure 6.12 we are concerned with only multiplexing, not switching. This means that all the data in a message from source 1 always go to one specific destination, be it 1, 2, 3, or 4. The delivery is fixed and unvarying, unlike switching. We also need to remember that TDM is, in principle, a digital multiplexing technique. Digital data from different sources are combined into one timeshared link. However, this does not mean that the sources cannot produce analog data; analog data can be sampled, changed to digital data, and then multiplexed by using TDM.

Figure 6.12 TDM

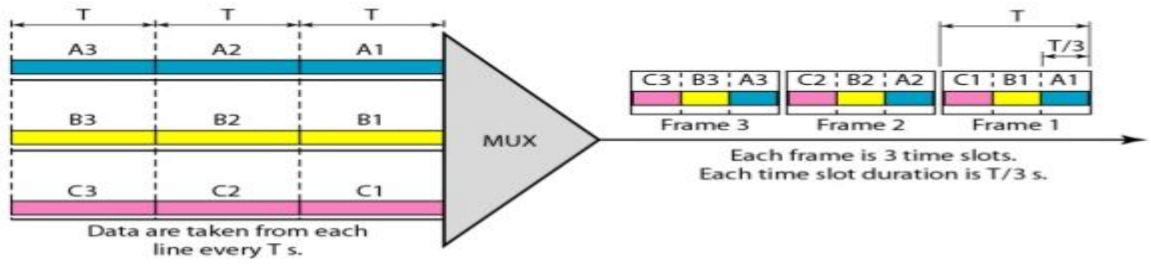


The TDM can be divided into two different schemes: synchronous and statistical. In synchronous TDM, each input connection has an allotment in the output even if it is not sending data.

Time Slots and Frames

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

Figure 6.13 Synchronous time-division multiplexing



In synchronous TDM, If we have *n* connections, a frame is divided into *n* time slots and one slot is allocated for each unit, one for each input line. If the duration of the input unit is *T,* the duration of each slot is *T/n* and the duration of each frame is *T* (unless a frame carries some other information).

The data rate of the output link must be *n* times the data rate of a connection to guarantee the flow of data. In Figure 6.13, the data rate of the link is 3 times the data rate of a connection; likewise, the duration of a unit on a connection is 3 times that of the time slot (duration of a unit on the link). In the figure we represent the data prior to multiplexing as 3 times the size of the data after multiplexing. This is just to convey the idea that each unit is 3 times longer in duration before multiplexing than after.

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

Example 6.5

In Figure 6.13, the data rate for each input connection is 3 kbps. If 1 bit at a time is multiplexed (a unit is 1 bit), what is the duration of (a) each input slot, (b) each output slot, and (c) each frame?

Solution

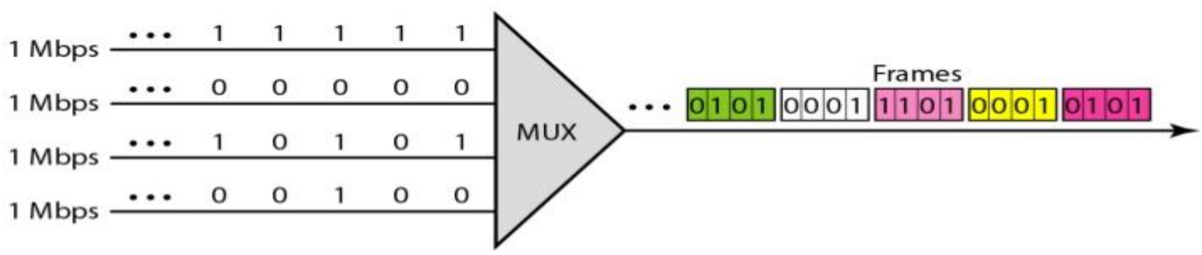
We can answer the questions as follows:

a. The data rate of each input connection is 1 kbps. This means that the bit duration is 111000 or 1 ms. The duration of the input time slot is 1 ms (same as bit duration).
b. The duration of each output time slot is one-third of the input time slot. This means that the duration of the output time slot is 1/3 ms.
c. Each frame carries three output time slots. So the duration of a frame is 3 x 113 ms, or 1ms.The duration of a frame is the same as the duration of an input unit.

Example 6.6

Figure 6.14 shows synchronous TOM with a data stream for each input and one data stream for the output. The unit of data is 1 bit. Find (a) the input bit duration, (b) the output bit duration, (c) the output bit rate, and (d) the output frame rate.

Figure 6.14 Example 6.6



We can answer the questions as follows:

a. The input bit duration is the inverse of the bit rate: 1/1 Mbps = 1 µs.
b. The output bit duration is one-fourth of the input bit duration, or 1/411s.
c. The output bit rate is the inverse of the output bit duration or 1/4 µs, or 4 Mbps. This can also be deduced from the fact that the output rate is 4 times as fast as any input rate; so the output rate =4 x 1 Mbps =4 Mbps.
d. The frame rate is always the same as any input rate. So the frame rate is 1,000,000 frames per second. Because we are sending 4 bits in each frame,

Example 6.7

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

Solution

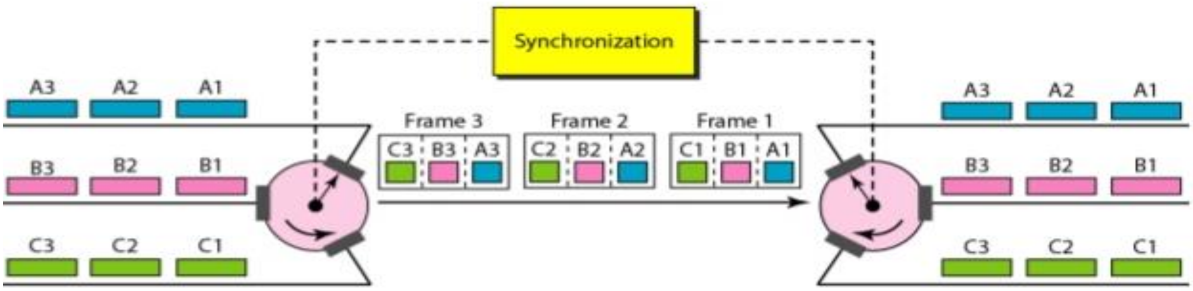
We can answer the questions as follows:

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

Interleaving

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

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



Empty Slots

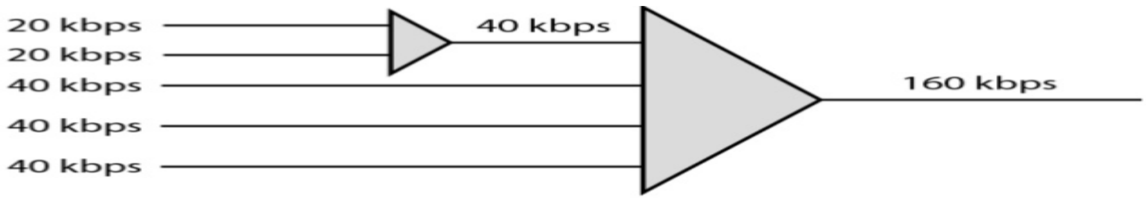
Synchronous TDM is not as efficient as it could be. If a source does not have data to send, the corresponding slot in the output frame is empty.

Data Rate Management

One problem with TDM is how to handle a disparity in the input data rates. If data rates are not the same, three strategies, or a combination of them, can be used. The three strategies are called as **multilevel multiplexing, multiple-slot allocation,** and **pulse stuffing.**

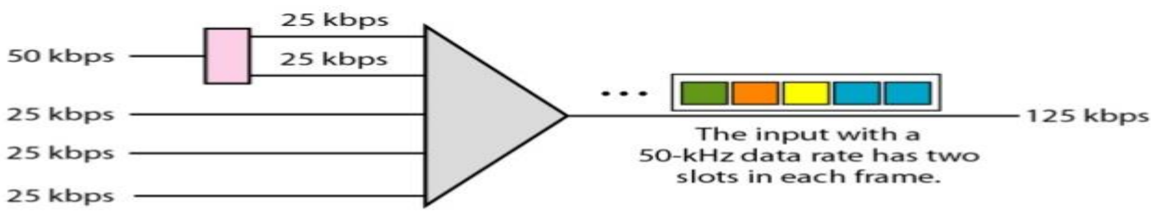
Multilevel Multiplexing Multilevel multiplexing is a technique used when the data rate of an input line is a multiple of others. For example, in Figure 6.19, we have two inputs of 20 kbps and three inputs of 40 kbps. The first two input lines can be multiplexed together to provide a data rate equal to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

Figure 6.19 Multilevel multiplexing



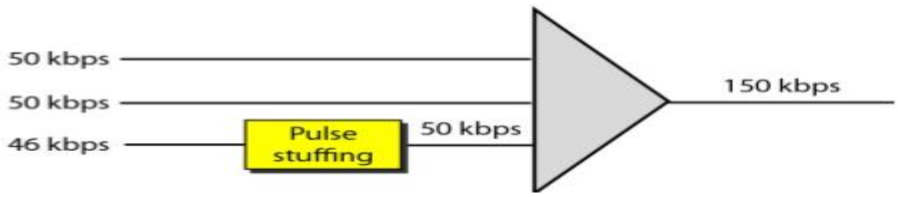
Multiple-Slot Allocation Sometimes it is more efficient to allot more than one slot in a frame to a single input line. For example, we might have an input line that has a data rate that is a multiple of another input. In Figure 6.20, the input line with a SO-kbps data rate can be given two slots in the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

Figure 6.20 Multiple-slot multiplexing



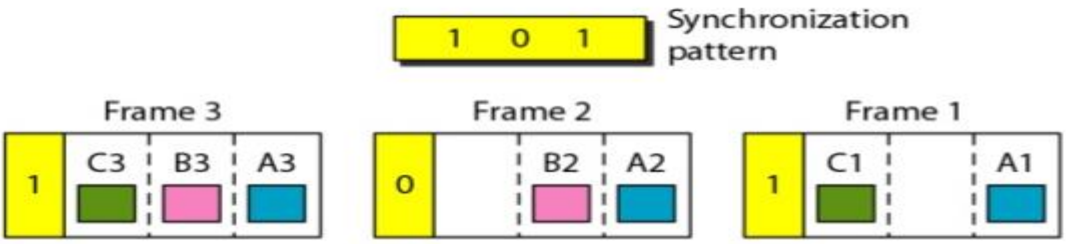
Pulse Stuffing Sometimes the bit rates of sources are not multiple integers of each other. Therefore, neither of the above two techniques can be applied. One solution is to make the highest input data rate the dominant data rate and then add dummy bits to the input lines with lower rates. This will increase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

Figure 6.21 Pulse stuffing



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

Figure 6.22 *Framing bits*



Example 6.10

We have four sources, each creating 250 characters per second. If the interleaved unit is a character and 1 synchronizing bit is added to each frame, find (a) the data rate of each source, (b) the duration of each character in each source, (c) the frame rate, (d) the duration of each frame, (e) the number of bits in each frame, and (f) the data rate of the link.

Solution

We can answer the questions as follows:

a. The data rate of each source is 250 x 8 = 2000 bps = 2 kbps.
b. Each source sends 250 characters per second; therefore, the duration of a character is 1/250 s,or4 ms.
c. Each frame has one character from each source, which means the link needs to send 250 frames per second to keep the transmission rate of each source.
d. The duration of each frame is *11250* s, or 4 ms. Note that the duration of each frame is the same as the duration of each character coming from each source.
e. Each frame carries 4 characters and I extra synchronizing bit. This means that each frame is 4 x 8 + 1 =33 bits.
f. The link sends 250 frames per second, and each frame contains 33 bits. This means that the data rate of the link is 250 x 33, or 8250 bps. Note that the bit rate of the link is greater than the combined bit rates of the four channels. If we add the bit rates of four channels, we get 8000 bps. Because 250 frames are traveling per second and each contains 1 extra bit for synchronizing, we need to add 250 to the sum to get 8250 bps.

Example 6.11

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

Solution

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

Statistical Time-Division Multiplexing

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

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

Addressing

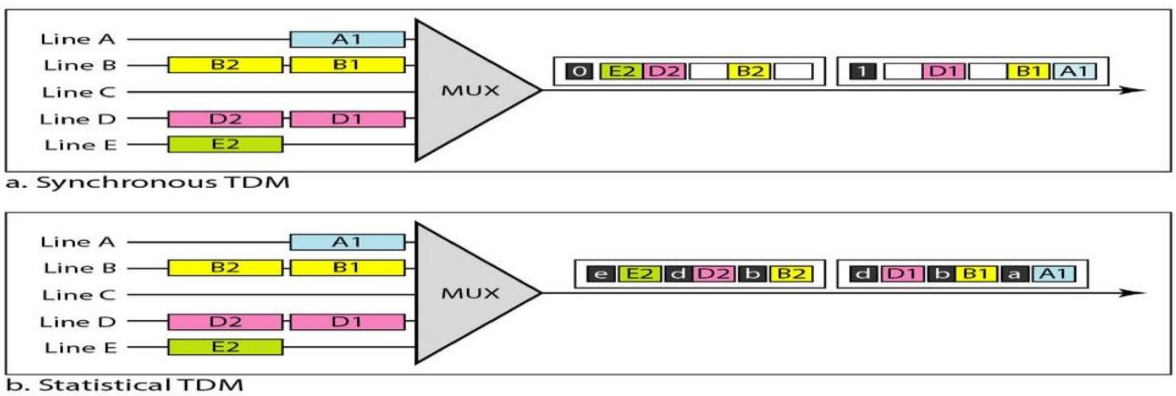
Figure 6.26 also shows a major difference between slots in synchronous TDM and statistical TDM. An output slot in synchronous TDM is totally occupied by data; in statistical TDM, a slot needs to carry data as well as the address of the destination. In synchronous TDM, there is no need for addressing; synchronization and preassigned relationships between the inputs and outputs serve as an address. We know, for example, that input 1 always goes to input 2.

If the multiplexer and the demultiplexer are synchronized, this is guaranteed. In statistical multiplexing, there is no fixed relationship between the inputs and outputs because there are no preassigned or reserved slots. There is a need to include the address of the receiver inside each slot to show where it is to be delivered. The addressing in its simplest form can be *n* bits to define *N* different output lines with *n* =10g2 *N.* For example, for eight different output lines, we need a 3-bit address.

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

No Synchronization Bit
There is another difference between synchronous and statistical TDM, but this time it is at the frame level. The frames in statistical TDM need not be synchronized, so we do not need synchronization bits.

Figure 6.26 TDM slot comparison



Bandwidth

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