JAWAHARLAL NEHRU TECHNOLOGICAL UNIVERSITY ANANTAPUR

B.Tech. III - II Sem.	**Th**	**Tu**	**C**
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(13A05601) COMPUTER NETWORKS

Course Objective

- *Study the evolution of computer networks and future direction*
- *Study the concepts of computer networks from layered perspective*
- *Study the issues open for research in computer networks*

Learning Outcome:

- *Use appropriate transmission media to connect to a computer network and Internet*
- *Work on the open issues for their project*
- *Start using the Internet effectively*
- *Able to design new protocols for computer network*

UNIT I

Introduction: Networks, Network Types, Internet History, Standards and Administration, Network Models: Protocol Layering, TCP/IP Protocol Suite, The ISO Model.

Introduction to physical layer: Data and Signals, Transmission impairment, Data rate limits,Performance, Transmission media: Introduction, Guided Media, Unguided Media, Switching:Introduction, Circuit Switched Networks, Packet switching.

UNIT II

Introduction to Data Link Layer: Introduction, Link layer addressing, Error detection and Correction: Cyclic codes, Checksum, Forward error correction, Data link control: DLC Services, Datalink layer protocols, HDLC, Point to Point Protocol, Media Access control: Random Access,Controlled Access, Channelization, Connecting devices and virtual LANs: Connecting Devices.

UNIT III

The Network Layer: Network layer design issues, Routing algorithms, Congestion control algorithms, Quality of service, Internetworking, The network layer in the Internet: IPV4 Addresses,IPV6, Internet Control protocol, OSPF, BGP, IP, ICMPv4, IGMP.

UNIT IV

The Transport Layer: The Transport Service, Elements of Transport Protocols, Congestion Control,The internet transport protocols: UDP, TCP, Performance problems in computer networks, Networkperformance measurement.

UNIT V

Introduction to Application Layer: Introduction, Client Server Programming, WWW and HTTP,FTP, e-mail, TELNET, Secure Shell, Domain Name System, SNMP.

Text Books:

1. *"Data communications and networking" 5th edition, 2012, Behrouz A. Forouzan, TMH.*
2. *"Computer Networks", 5th edition, 2010, Andrew S. Tanenbaum, Wetherall, Pearson.*

Reference Books:

1. *"Internetworking with TCP/IP – Principles, protocols, and architecture- Volume 1, Douglas E. Comer, 5th edition, PHI*
2. *"Computer Networks", 5E, Peterson, Davie, Elsevier.*
3. *"Introduction to Computer Networks and Cyber Security", Chawan- Hwa Wu, Irwin, CRC Publications.*
4. *"Computer Networks and Internets with Internet Applications", Comer.*

UNIT-1 COMPUTER NETWORKS(13A05601)

Network: A network is the interconnection of a set of devices capable of communication. In this definition, a device can be a host (or an *end system* as it is sometimes called) such as a large computer, desktop, laptop, workstation, cellular phone, or security system. A device in this definition can also be a connecting device such as a router, which connects the network to other networks, a switch, which connects devices together, a modem (modulator-demodulator), which changes the form of data, and so on. These devices in a network are connected using wired or wireless transmission media such as cable or air. When we connect two computers at home using a plug-and-play router, we have created a network, although very small.

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 throughputs 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.

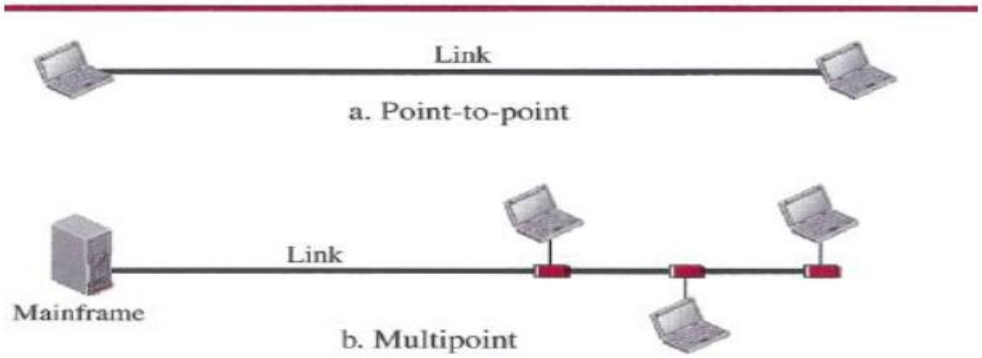
Point-to-Point

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. When we change television channels by infrared remote control, we 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.

'ypes of connections: point-to-point and multipoint



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.

Networks Types:

Local Area Networks:

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

(1) Their size,
(2) Their transmission technology, and
(3) Their topology.

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

LANs may use a transmission technology consisting of a cable to which all the machines are attached, like the telephone company party lines once used in rural areas. Traditional LANs run at speeds of 10 Mbps to 100 Mbps, have low delay (microseconds or nanoseconds), and make very few errors. Newer LANs operate at up to 10 Gbps Various topologies are possible for broadcast LANs. Figure1 shows two of them. In a bus (i.e., a linear cable) network, at any instant at most one machine is the master and is allowed to transmit. All other machines are required to refrain from sending. An arbitration mechanism is needed to resolve conflicts when two or more machines want to transmit simultaneously. The arbitration mechanism may be centralized or distributed. IEEE 802.3,popularly called Ethernet, for example, is a bus-based broadcast network with decentralized control, usually operating at 10 Mbps to 10 Gbps. Computers on an Ethernet can transmit whenever they want to; if two or more packets collide, each computer just waits a random time and tries again later.

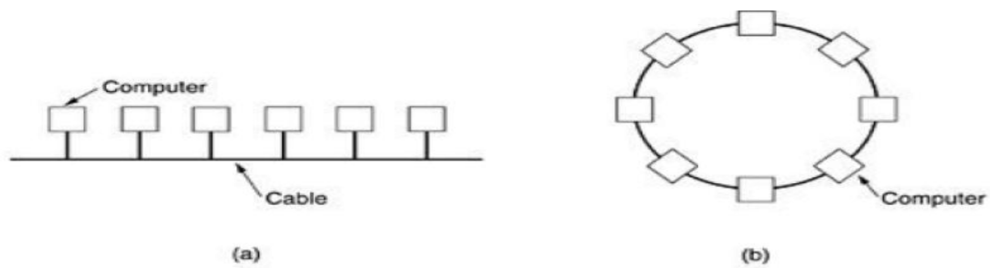


Fig.1: Two broadcast networks . (a) Bus. (b) Ring.

A second type of broadcast system is the ring. In a ring, each bit propagates around on its own, not waiting for the rest of the packet to which it belongs. Typically, each bit circumnavigates the entire ring in the time it takes to transmit a few bits, often before the complete packet has even been transmitted. As with all other broadcast systems, some rule is needed for arbitrating simultaneous accesses to the ring. Various methods, such as having the machines take turns, are in use. IEEE 802.5 (the IBM token ring), is a ring-based LAN operating at 4 and 16 Mbps. FDDI is another example of a ring network.

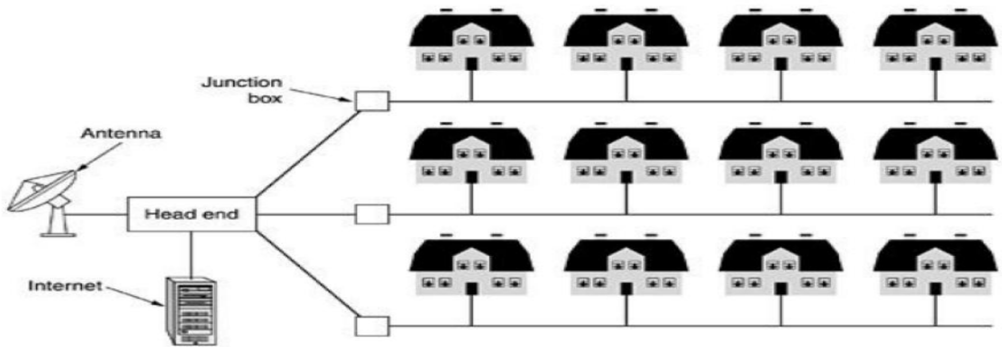
Metropolitan Area Network (MAN).

A metropolitan area network, or MAN, covers a city. The best-known example of a MAN is the cable television network available in many cities. This system grew from earlier community antenna systems used in areas with poor over-the-air television reception. In these early systems, a large antenna was placed on top of a nearby hill and signal was then piped to the subscribers' houses.

At first, these were locally-designed, ad hoc systems. Then companies began jumping into the business, getting contracts from city governments to wire up an entire city. The next step was television programming and even entire channels designed for cable only. Often these channels were highly specialized, such as all news, all sports, all cooking, all gardening, and so on. But from their inception until the late 1990s, they were intended for television reception only.

To a first approximation, a MAN might look something like the system shown in Fig.2. In this figure both television signals and Internet are fed into the centralized head end for subsequent distribution to people's homes.

Cable television is not the only MAN. Recent developments in high-speed wireless Internet access resulted in another MAN, which has been standardized as IEEE 802.16.



A MAN is implemented by a standard called DQDB (Distributed Queue Dual Bus) or IEEE 802.16. DQDB has two unidirectional buses (or cables) to which all the computers are attached.

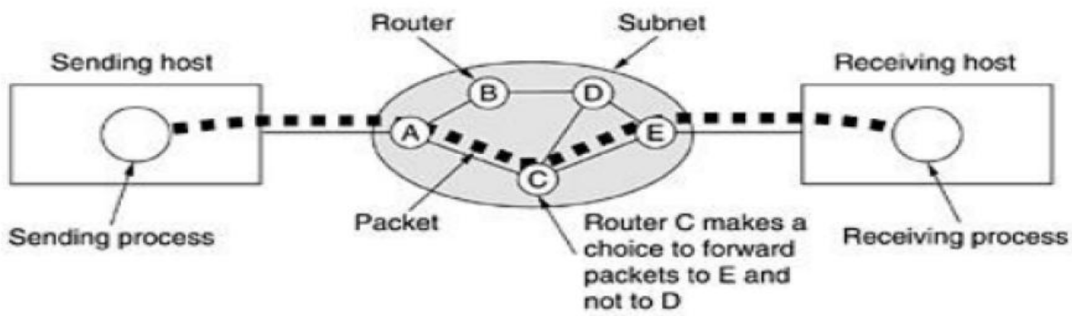
Wide Area Network (WAN).

A wide area network, or WAN, spans a large geographical area, often a country or continent. It contains a collection of machines intended for running user (i.e., application) programs. These machines are called as hosts. The hosts are connected by a communication subnet, or just subnet for short. The hosts are owned by the customers (e.g., people's personal computers), whereas the communication subnet is typically owned and operated by a telephone company or Internet service provider. The job of the subnet is to carry messages from host to host, just as the telephone system carries words from speaker to listener. Separation of the pure communication aspects of the network (the subnet) from the application aspects (the hosts), greatly simplifies the complete network design. In most wide
area networks, the subnet consists of two distinct components: transmission lines and switching elements. Transmission lines move bits between machines. They can be made of copper wire, optical fiber, or even radio links.

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

The principle of a packet-switched WAN is so important. Generally, when a process on some host has a message to be sent to a process on some other host, the sending host first cuts the message into packets, each one bearing its number in the sequence. These packets are then injected into the network one at a time in quick succession. The packets are transported individually over the network and deposited at the receiving host, where they are reassembled into the original message and delivered to the receiving process. A stream of packets resulting from some initial message is illustrated in Fig.3.1.

In this figure, all the packets follow the route ACE, rather than ABDE or ACDE. In some networks all packets from a given message must follow the same route; in others each packed is routed separately. Of course, if ACE is the best route, all packets may be sent along it, even if each packet is individually routed.



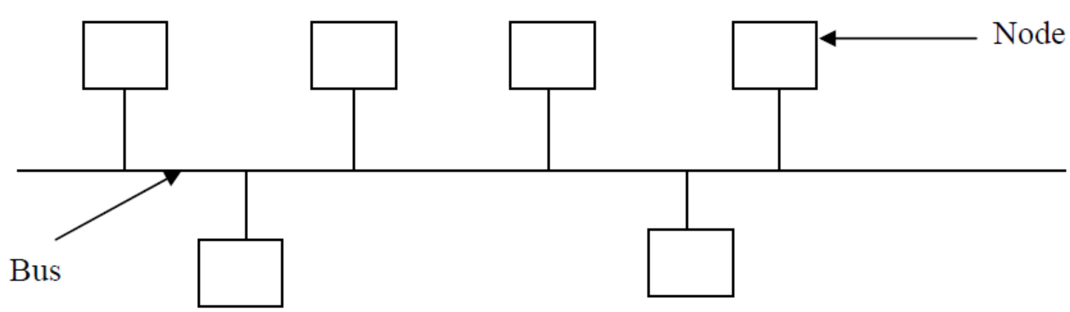
Not all WANs are packet switched. A second possibility for a WAN is a satellite system. Each router has an antenna through which it can send and receive. All routers can hear the output from the satellite, and in some cases they can also hear the upward transmissions of their fellow routers to the satellite as well. Sometimes the routers are connected to a substantial point-to-point subnet, with only some of them having a satellite antenna. Satellite networks are inherently broadcast and are most useful when the broadcast property is important.

Network topologies:

Network topology defined as the logical connection of various computers in the network.
The six basic network topologies are: bus, ring, star, tree, mesh and hybrid.

1. Bus Topology:

In bus topology all the computers are connected to a long cable called a bus. A node that wants to send data puts the data on the bus which carries it to the destination node. In this topology any computer can data over the bus at any time. Since, the bus is shared among all the computers. When two or more computers to send data at the same time, an arbitration mechanism is needed to prevent simultaneous access to the bus.

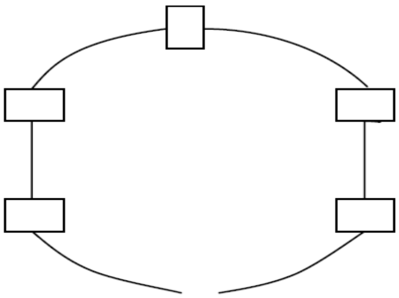


Bus Topology

A bus topology is easy to install but is not flexible i.e., it is difficult to add a new node to bus. In addition to this the bus stops functioning even if a portion of the bus breaks down. It is also very difficult to isolate fault.

2. Ring Topology:

In ring topology, the computers are connected in the form of a ring. Each node has exactly two adjacent neighbors. To send data to a distant node on a ring it passes through many intermediate nodes to reach to its ultimate destination.

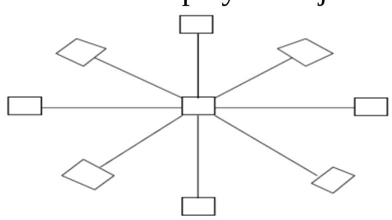


Ring Topology

A ring topology is as to install and reconfigure. In this topology, fault isolation is easy because a signal that circulates all the time in a ring helps in identifying a faulty node.
The data transmission takes place in only one direction. When a node fails in ring, it breaks down the whole ring. To overcome this drawback some ring topologies use dual rings.
The topology is not useful to connect large number of computers.

3. Star Topology:

In star topology all the nodes are connected to a central node called a hub. A node that wants to send some six data to some other node on the network, send data to a hub which in turn sends it the destination node. A hub plays a major role in such networks.

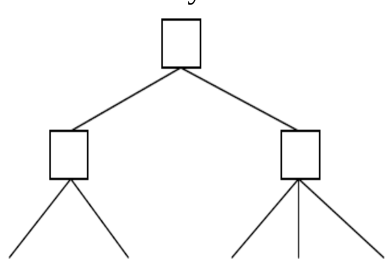


Star topology

Star topology is easy to install and reconfigure. If a link fails then it separates the node connected to link from the network and the network continues to function. However, if the hub goes down, the entire network collapses.

4. Tree Topology:

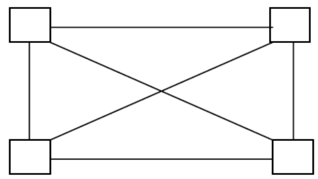
Tree topology is a hierarchy of various hubs. The entire nodes are connected to one hub or the other. There is a central hub to which only a few nodes are connected directly.



The central hub, also called active hub, looks at the incoming bits and regenerates them so that they can traverse over longer distances. The secondary hubs in tree topology may be active hubs or passive hubs. The failure of a transmission line separates a node from the network.

5. Mesh Topology:

A mesh topology is also called complete topology. In this topology, each node is connected directly to every oilier node in the network. That is if there are n nodes then there would be n(n — 1)/2 physical links in the network.



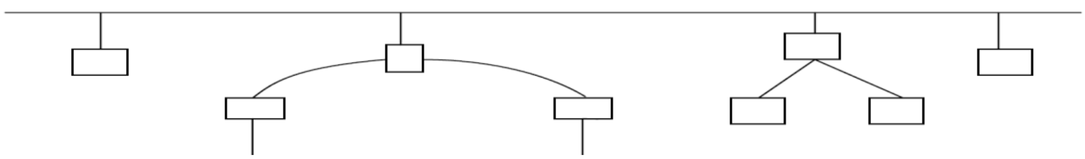
Mesh Topology

As there are dedicated links, the topology does not have congestion problems. Further it does not need a special Media Access Control (MAC) protocol to prevent simultaneous access to the transmission media since links are dedicated, not shared. The topology also provides data security.

The network can continue to function even in the failure of one of the links. Fault identification is also easy. The main disadvantage of mesh topology is the complexity of the network and the cost associated with the cable length. The mesh topology is not useful for medium to large networks.

6. Hybrid Topology:

Hybrid topology is formed by connecting two or more topologies together. For example, hybrid topology can be created by using the bus, star and ring topologies,



PROTOCOL LAYERING

We defined the term *protocol* in Chapter 1. In data communication and networking, aprotocol defines the rules that both the sender and receiver and all intermediate devices need to follow to be able to communicate effectively. When communication is simple, we may need only one simple protocol; when the communication is complex, we may need to divide the task between different layers, in which case we need a protocol at each layer, or protocol layering.

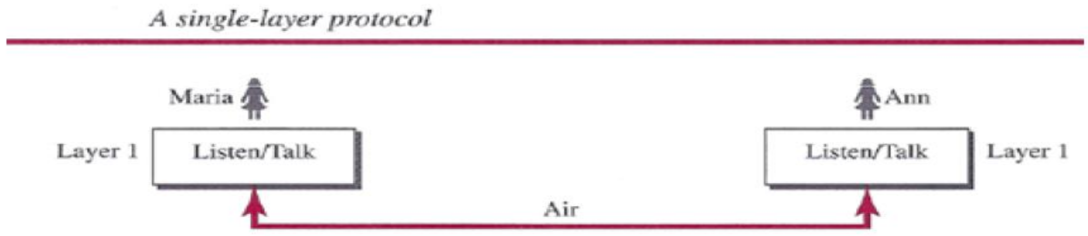
Scenarios

Let us develop two simple scenarios to better understand the need for protocol layering.

First Scenario

In the first scenario, communication is so simple that it can occur in only one layer. Assume Maria and Ann are neighbors with a lot of common ideas. Communication between Maria and Ann takes place in one layer, face to face, in the same language, as shown in

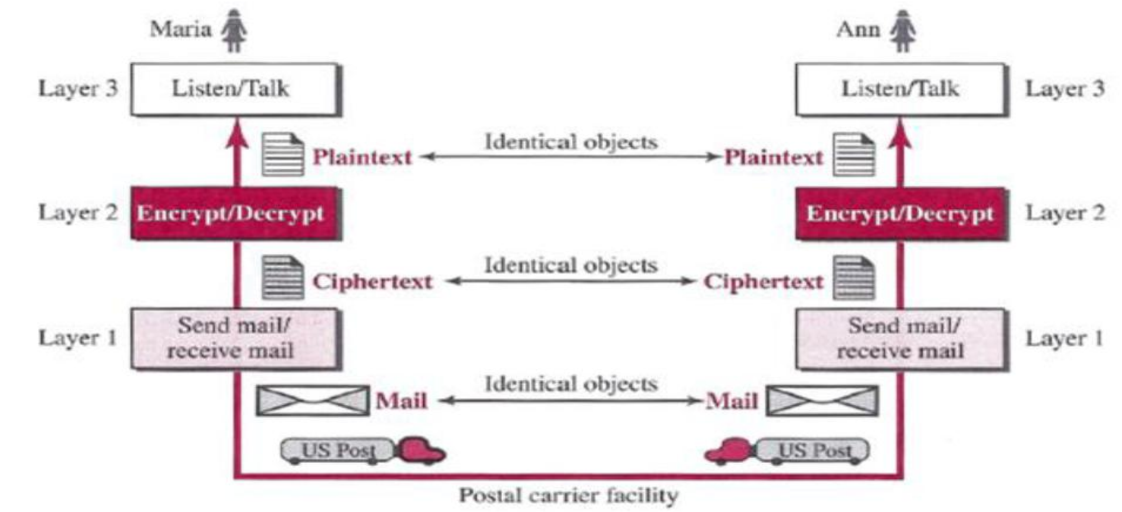
Figure.



Even in this simple scenario, we can see that a set of rules needs to be followed. First, Maria and Ann know that they should greet each other when they meet. Second, they know that they should confine their vocabulary to the level of their friendship. Third, each party knows that she should refrain from speaking when the other party is speaking. Fourth, each party knows that the conversation should be a dialog, not a monolog: both should have the opportunity to talk about the issue. Fifth, they should exchange some nice words when they leave. We can see that the protocol used by Maria and Ann is different from the communication between a professor and the students in a lecture hall. The communication in the second case is mostly monolog; the professor talks most of the time unless a student has a question, a situation in which the protocol dictates that she should raise her hand and wait for permission to speak. In this case, the communication is normally very formal and limited to the subject being taught.

Second Scenario

In the second scenario, we assume that Ann is offered a higher-level position in her company, but needs to move to another branch located in a city very far from Maria. The two friends still want to continue their communication and exchange ideas because they have come up with an innovative project to start a new business when they both retire. They decide to continue their conversation using regular mail through the post office. However, they do not want their ideas to be revealed by other people if the letters are intercepted. They agree on an encryption/decryption technique. The sender of the letter encrypts it to make it unreadable by an intruder; the receiver of the letter decrypts it to get the original letter. We discuss the encryption/decryption methods in but for the moment we assume that Maria and Ann use one technique that makes it hard to decrypt the letter if one does not have the key for doing so. Now we can say that the communication between Maria and Ann takes place in three layers, as shown in Figure. We assume that Ann and Maria each have three machines (or robots) that can perform the task at each layer.



Principles of Protocol Layering

Let us discuss two principles of protocol layering.

First Principle

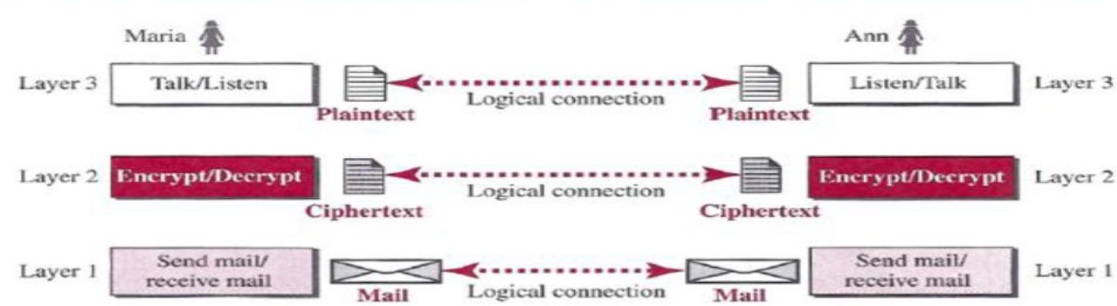
The first principle dictates that if we want bidirectional communication, we need to make each layer so that it is able to perform two opposite tasks, one in each direction. For example, the third layer task is to listen (in one direction) and *talk* (in the other direction). The second layer needs to be able to encrypt and decrypt. The first layer needs to send and receive mail.

Second Principle

The second principle that we need to follow in protocol layering is that the two objects under each layer at both sites should be identical. For example, the object under layer 3 at both sites should be a plaintext letter. both sites should be a cipher text letter. The object under layer 1 at both sites should be a piece of mail.

Logical Connections

After following the above two principles, we can think about logical connection between each layer as shown in below figure. This means that we have layer-to-layer communication. Maria and Ann can think that there is a logical (imaginary) connection at each layer through which they can send the object created from that layer. We will see that the concept of logical connection will help us better understand the task of layering. We encounter in data communication and networking.

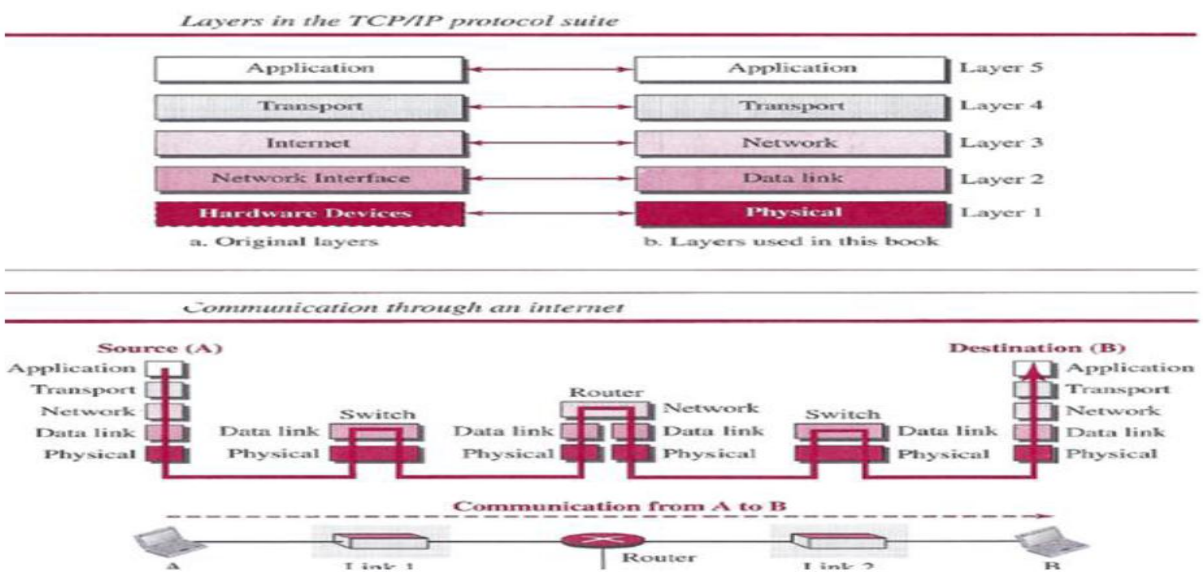


TCP/IP PROTOCOL SUITE

Now that we know about the concept of protocol layering and the logical communication between layers in our second scenario, we can introduce the *TCP/IP* (Transmission Control Protocol/Internet Protocol). *TCP/IP* is a protocol suite (a set of protocols organized in different layers) used in the Internet today. It is a hierarchical protocol made up of interactive modules, each of which provides a specific functionality. The term *hierarchical* means that each upper level protocol is supported by the services provided by one or more lower level protocols. The original *TCP/IP* protocol suite was defined as four software layers built upon the hardware. Today, however, *TCP/IP* is thought of as a five-layer model. Following figure shows both configurations.

Layered Architecture

To show how the layers in the *TCP/IP* protocol suite are involved in communication between two hosts, we assume that we want to use the suite in a small internet made up of three LANs (links), each with a link-layer switch. We also assume that the links are connected by one router, as shown in below Figure.



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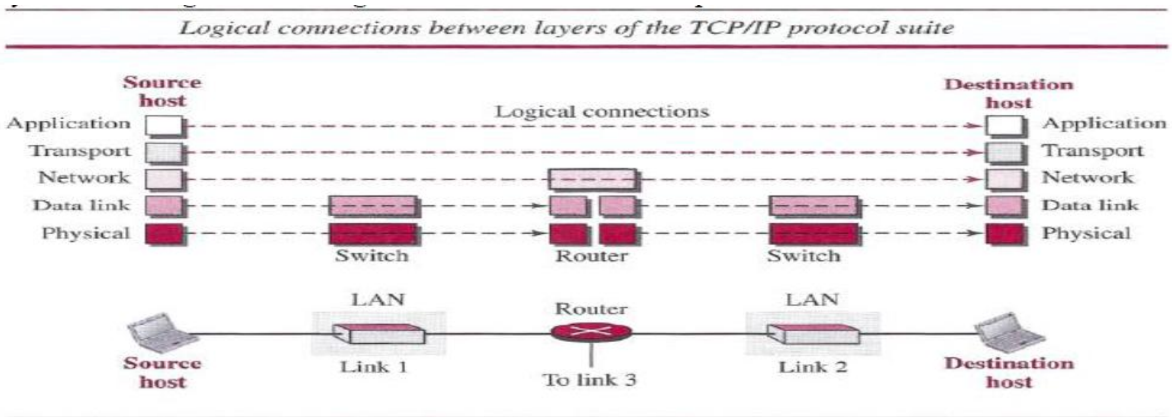
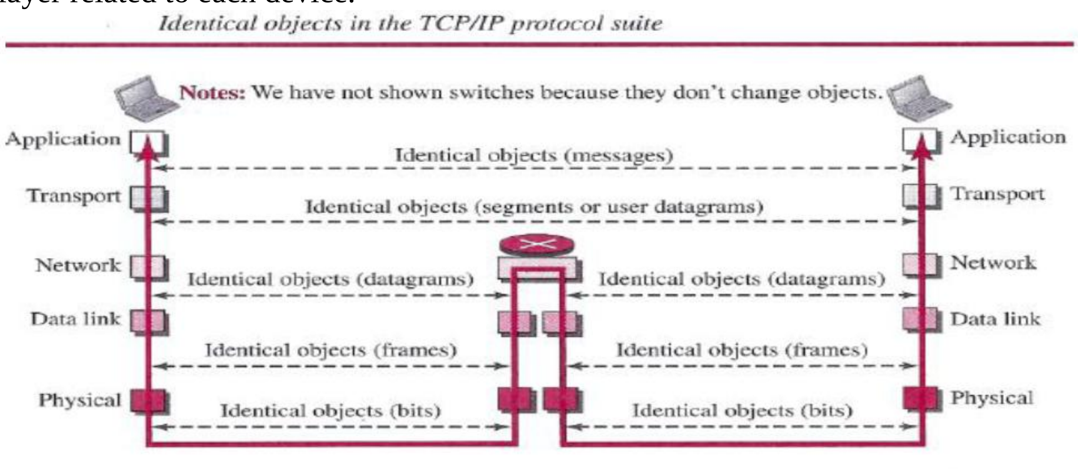
Layers in the TCP/IP Protocol Suite

After the above introduction, we briefly discuss the functions and duties of layers in the *TCP/IP* protocol suite. Each layer is discussed in detail in the next five parts of the book. To better understand the duties of each layer, we need to think about the logical connections between layers. Below figure shows logical connections in our simple internet.

Using logical connections makes it easier for us to think about the duty of each layer. As the figure shows, the duty of the application, transport, and network layers is end-to-end. However, the duty of the data-link and physical layers is hop-to-hop, in which a hop is a host or router. In our simple internet.



In other words, the domain of duty of the top three layers is the internet, and the domain of duty of the two lower layers is the link. Another way of thinking of the logical connections is to think about the data unit created from each layer. In the top three layers, the data unit (packets) should not be changed by any router or link-layer switch. In the bottom two layers, the packet created by the host is changed only by the routers, not by the link-layer switches. Below figure shows the second principle discussed previously for protocol layering. We show the identical objects below each layer related to each device.



Note that, although the logical connection at the network layer is between the two hosts, we can only say that identical objects exist between two hops in this case because a router may fragment the packet at the network layer and send more packets than received . Note that the link between two hops does not change the object.

Description of Each Layer

After understanding the concept of logical communication, we are ready to briefly discuss the duty of each layer.

Physical Layer

We can say that the physical layer is responsible for carrying individual bits in a frame across the link. Although the physical layer is the lowest level in the TCPIIP protocol suite, the communication between two devices at the physical layer is still a logical communication because there is another, hidden layer, the transmission media, under the physical layer. Two devices are connected by a transmission medium (cable or air). We need to know that the transmission medium does not carry bits; it carries electrical or optical signals. So the bits received in a frame from the data-link layer are transformed and sent through the transmission media, but we can think that the logical unit between two physical layers in two devices is a *bit.* There are several protocols that transform a bit to a signal.

Data-link Layer

We have seen that an internet is made up of several links (LANs and WANs) connected by routers. There may be several overlapping sets of links that a datagram can travel from the host to the destination. The routers are responsible for choosing the *best* links. However, when the next link to travel is determined by the router, the data-link layer is responsible for taking the datagram and moving it across the link. The link can be a wired LAN with a link-layer switch, a wireless LAN, a wired WAN, or a wireless WAN. We can also have different protocols used with any link type. In each case, the data-link layer is responsible for moving the packet through the link. *TCP/IP* does not define any specific protocol for the data-link layer. It supports all the standard and proprietary protocols. Any protocol that can take the datagram and carry it through the link suffices for the network layer. The data-link layer takes a datagram and encapsulates it in a packet called *«frame.* Each link-layer protocol may provide a different service. Some link-layer protocols provide complete error detection and correction, some provide only error correction.

Network Layer

The network layer is responsible for creating a connection between the source computer and the destination computer. The communication at the network layer is host-to-host. However, since there can be several routers from the source to the destination, the routers in the path are responsible for choosing the best route for each packet. We can say that the network layer is responsible for host-to-host communication and routing the packet through possible routes. Again, we may ask ourselves why we need the network layer. We could have added the routing duty to the transport layer and dropped this layer. One reason, as we said before, is the separation of different tasks between different layers. The second reason is that the routers do not need the application and transport layers.

Transport Layer

The logical connection at the transport layer is also end-to-end. The transport layer at the source host gets the message from the application layer, encapsulates it in a transport layer packet (called a *segment* or a *user datagram* in different protocols) and sends it, through the logical (imaginary) connection, to the transport layer at the destination host. In other words, the transport layer is responsible for giving services to the application layer: to get a message from an application program running on the source host and deliver it to the corresponding application program on the destination host. We may ask why we need an end-to-end transport layer when we already have an end-to-end application layer. The reason is the separation of tasks and duties, which we discussed earlier. The transport layer should be independent of the application layer. In addition, we will see that we have more than one protocol in the transport layer, which means that each application program can use the protocol that best matches its requirement.

Application Layer

The logical connection between the two application layers is end to-end. The two application layers exchange *messages* between each other as though there were a bridge between the two layers. However, we should know that the communication is done through all the layers. Communication at the application layer is between two *processes* (two programs running at this layer). To communicate, a process sends a request to the other process and receives a response. Process-to-process communication is the duty of the application layer. The application layer in the Internet includes many predefined protocols, but a user can also create a pair of processes to be run at the two hosts.

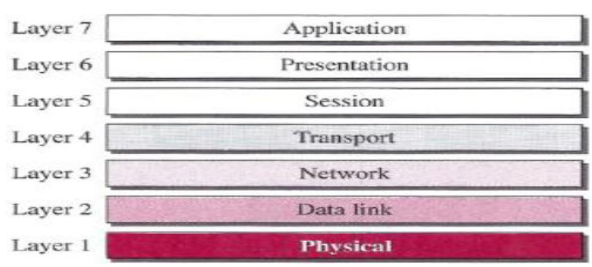
THE OSI MODEL

Although, when speaking of the Internet, everyone talks about the *TCP/IP* protocol suite, this suite is not the only suite of protocols defined. Established in 1947, the International Organization for Standardization (ISO) is a multinational body dedicated to worldwide agreement on international standards. Almost three-fourths of the countries in the world are represented in the ISO. An ISO standard that covers all aspects of network communications is the Open Systems Interconnection (OSI) model. It was first introduced in the late 1970s.

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 .

The OSI model



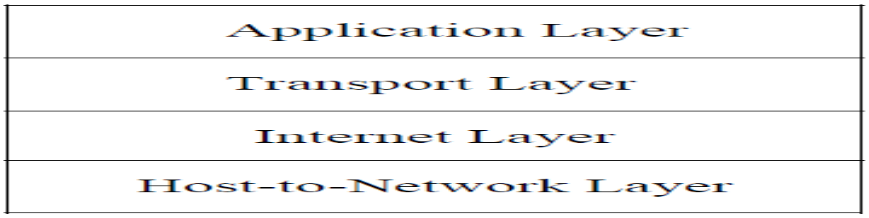
The TCP/IP Reference Model:

The TCP/IP reference model was developed prior to OSI model. The major design goals of this model were,

1. To connect multiple networks together so that they appear as a single network.
2. To survive after partial subnet hardware failures.
3. To provide a flexible architecture.

Unlike OSI reference model, TCP/IP reference model has only 4 layers. They are,

1.Host-to-Network Layer
2. Internet Layer
3. Transport Layer
4. Application Layer



TCP/IP Reference model

Host-to-Network Layer:

The TCP/IP reference model does not really say much about what happens here,except to point out that the host has to connect to the network using some protocol so it can send IP packets to it. This protocol is not defined and varies from host to host and network to network.

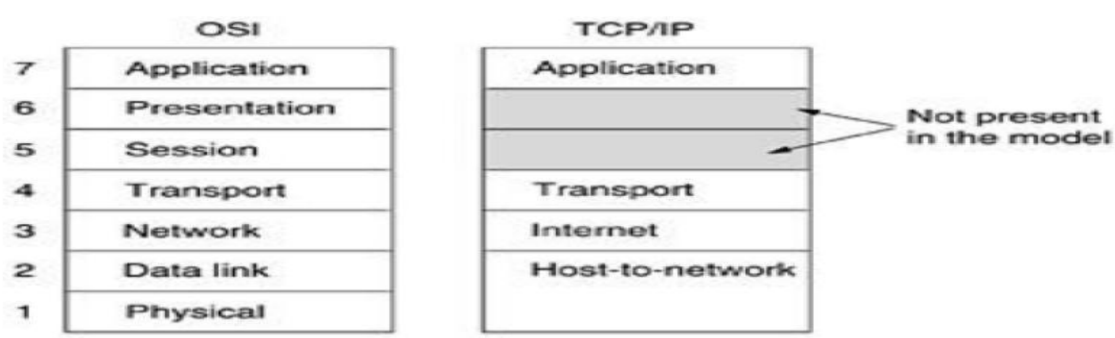
Internet Layer:

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

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

The Transport Layer:

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



The second protocol in this layer, UDP (User Datagram Protocol), is an unreliable,connectionless protocol for applications that do not want TCP's sequencing or flow control and wish to provide their own. It is also widely used for one-shot, client-server-type requestreply queries and applications in which prompt delivery is more important than accurate delivery, such as transmitting speech or video. The relation of IP, TCP, and UDP is shown.

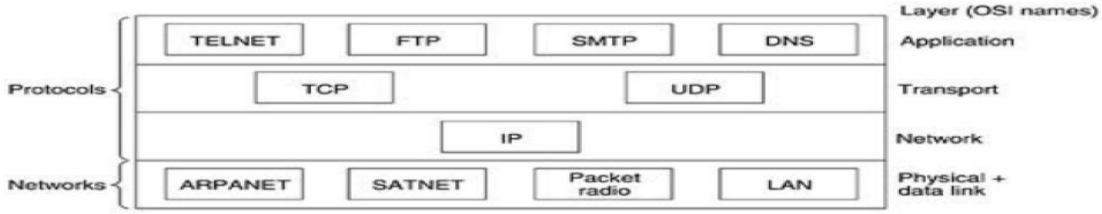


Fig.6.2: Protocols and networks in the TCP/IP model initially.

Application layer:

The TCP/IP model does not have session or presentation layers.On top of the transport layer is the application layer. It contains all the higher-level protocols. The early ones included virtual terminal (TELNET), file transfer (FTP), and electronic mail (SMTP), as shown in Fig.6.2. The virtual terminal protocol allows a user on one machine to log onto a distant machine and work there. The file transfer protocol provides a way to move data efficiently from one machine to another. Electronic mail was originally just a kind of file transfer, but later a specialized protocol (SMTP) was developed for it.

Many other protocols have been added to these over the years: the Domain Name System (DNS) for mapping host names onto their network addresses, NNTP, the protocol for moving USENET news articles around, and HTTP, the protocol for fetching pages on the World Wide Web, and many others.

Comparison of the OSI and TCP/IP Reference Models:

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

Despite these fundamental similarities, the two models also have many differences

Three concepts are central to the OSI model:

1.Services.

2. Interfaces.

3. Protocols.

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

A layer's interface tells the processes above it how to access it. It specifies what the parameters are and what results to expect. It, too, says nothing about how the layer works inside. Finally, the peer protocols used in a layer are the layer's own business. It can use any protocols it wants to, as

long as it gets the job done (i.e., provides the offered services). It can also change them at will without affecting software in higher layers.
The TCP/IP model did not originally clearly distinguish between service, interface, and protocol, although people have tried to retrofit it after the fact to make it more OSI-like. For example, the only real services offered by the internet layer are SEND IP PACKET and RECEIVE IP PACKET.
As a consequence, the protocols in the OSI model are better hidden than in the TCP/IP model and can be replaced relatively easily as the technology changes. Being able to make such changes is one of the main purposes of having layered protocols in the first place.
The OSI reference model was devised before the corresponding protocols were invented. This ordering means that the model was not biased toward one particular set of protocols, a fact that made it quite general. The downside of this ordering is that the designers did not have much experience with the subject and did not have a good idea of which functionality to put in which layer.
Another difference is in the area of connectionless versus connection-oriented communication. The OSI model supports both connectionless and connection-oriented communication in the network layer, but only connection-oriented communication in the transport layer, where it counts (because the transport service is visible to the users). The TCP/IP model has only one mode in the network layer (connectionless) but supports both modes in the transport layer, giving the users a choice. This choice is especially important for
simple request-response protocols.

Problems of the TCP/IP Reference Mode:

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

Second, the TCP/IP model is not at all general and is poorly suited to describing any protocol stack other than TCP/IP. Trying to use the TCP/IP model to describe Bluetooth, for example, is completely impossible.

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

Fourth, the TCP/IP model does not distinguish (or even mention) the physical and data link layers. These are completely different. The physical layer has to do with the transmission characteristics of copper wire, fiber optics, and wireless communication. The data link layer's job is to delimit the start and end of frames and get them from one side to the other with the desired degree of reliability. A proper model should include both as separate layers. The TCP/IP model does not do this. Finally, although the IP and TCP protocols were carefully thought out and well implemented, many of the other protocols were ad hoc, generally produced by a couple of graduate students hacking away until they got tired. The protocol implementations were then distributed free, which resulted in their becoming widely used, deeply entrenched, and thus hard to replace. Some of them are a bit of an embarrassment now. The virtual terminal protocol, TELNET, for example, was designed for a ten-character per second mechanical Teletype terminal. It knows nothing of graphical user interfaces and mice. Nevertheless, 25 years later, it is still in widespread use.

Problems of the OSI Model and Protocols:

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

1. Bad Timing:

The time at which a standard is established is absolutely critical to its success. David Clark of M.I.T. has a theory of standards that he calls the apocalypse of the two elephants, which is illustrated in Fig. This figure shows the amount of activity surrounding a new subject. When the subject is first discovered, there is a burst of research activity in the form of discussions, papers, and meetings. After a while this activity subsides, corporations discover the subject,
and the billion-dollar wave of investment hits. It is essential that the standards be written in the trough in between the two "elephants." If the standards are written too early, before the research is finished, the subject may still be poorly understood; the result is bad standards. If they are written too late, so many companies may have already made major investments in different ways of doing things that the standards are effectively ignored. If the interval between the two elephants is very short(because everyone is in a hurry to get started), the people developing the standards may get crushed.

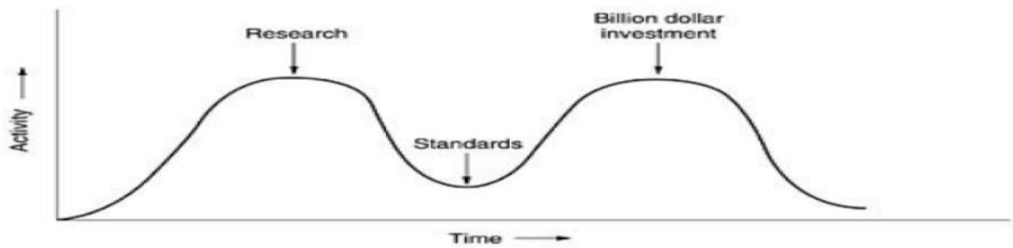


Fig.9: The apocalypse of the two elephants

2. Bad Technology:

The second reason that OSI never caught on is that both the model and the protocols are flawed. The choice of seven layers was more political than technical, and two of the layers (session and presentation) are nearly empty, whereas two other ones (data link and network) are overfull.

The OSI model, along with the associated service definitions and protocols, is extraordinarily complex. When piled up, the printed standards occupy a significant fraction of a meter of paper. They are also difficult to implement and inefficient in operation. In addition to being incomprehensible, another problem with OSI is that some functions, such as addressing, flow control, and error control, reappear again and again in each layer.

3. Bad Implementations:

Given the enormous complexity of the model and the protocols, it will come as no surprise that the initial implementations were huge, unwieldy, and slow. Everyone who tried them got burned. It did not take long for people to associate "OSI" with "poor quality." Although the products improved in the course of time, the image stuck.

4. Bad Politics:

On account of the initial implementation, many people, especially in academia, thought of TCP/IP as part of UNIX, and UNIX in the 1980s in academia was not unlike parenthood (then incorrectly called motherhood) and apple pie. OSI, on the other hand, was widely thought to be the creature of the European telecommunication ministries, the European Community, and later the U.S. Government. This belief was only partly true, but the very idea of a bunch of government bureaucrats trying to shove a technically inferior standard down the throats of the poor researchers and programmers down in the trenches actually developing computer networks did not help much. Some people viewed this development in the same light as IBM announcing in the 1960s that PL/I was the language of the future, or DoD correcting this later by announcing that it was actually Ada.

INTERNET HISTORY

Now that we have given an overview of the Internet, let us give a brief history of the internet. This brief history makes it clear how the Internet has evolved from a private network to a global one in less than 40 years.

Early History

There were some communication networks, such as telegraph and telephone networks, before 1960. These networks were suitable for constant-rate communication at that time, which means that after a connection was made between two users, the encoded message (telegraphy) or voice (telephony) could be exchanged.

ARPANET

In the mid-1960s, mainframe computers in research organizations were stand-alone 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 the Advanced Research Projects Agency Network (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 turn, would be connected to each other. Each IMP had to be able to communicate with other IMPs as well as with its own attached host.

Birth of the Internet

In 1972, Vint Cerf and Bob Kahn, both of whom were part of the core ARPANET group, collaborated on what they called the *Internet ting Project. TCPI/P* Cerf and Kahn's landmark 1973 paper outlined the protocols to achieve end-to-end delivery of data. This was a new version of NCP. This paper on transmission control protocol (TCP) included concepts such as encapsulation, the datagram, and the functions of a gateway. Transmission Control Protocol (TCP) and Internet Protocol (IP). IP would handle datagram routing while TCP would be responsible for higher level functions such as segmentation, reassembly, and error detection. The new combination became known as TCPIIP.

MILNET

In 1983, ARPANET split into two networks: Military Network (MILNET) for military users and ARPANET for non military users.

CSNET

Another milestone in Internet history was the creation of CSNET in 1981. Computer Science Network (CSNET) was a network sponsored by the National Science Foundation (NSF).

NSFNET

With the success of CSNET, the NSF in 1986 sponsored the National Science Foundation Network (NSFNET), a backbone that connected five supercomputer centers located throughout the United States.

ANSNET

In 1991, the U.S. government decided that NSFNET was not capable of supporting the rapidly increasing Internet traffic. Three companies, IBM, Merit, and Verizon, filled the void by forming a nonprofit organization called Advanced Network & Services (ANS) to build a new, high-speed Internet backbone called Advanced Network Services Network (ANSNET).

Internet Today

Today, we witness a rapid growth both in the infrastructure and new applications. The Internet today is a set of pier networks that provide services to the whole world. What has made the internet so popular is the invention of new applications.

World Wide Web

The 1990s saw the explosion of Internet applications due to the emergence of the World Wide Web (WWW). The Web was invented at CERN by Tim Berners-Lee. This invention has added the commercial applications to the Internet.

Multimedia

Recent developments in the multimedia applications such as voice over IP (telephony), video over IP (Skype), view sharing (YouTube), and television over IP (PPLive) has increased the number of users and the amount of time each user spends on the network.

Peer-to-Peer Applications

Peer-to-peer networking is also a new area of communication with a lot of potential.

STANDARDS AND ADMINISTRATION

In the discussion of the Internet and its protocol, we often see a reference to a standard or an administration entity. In this section, we introduce these standards and administration entities for those readers that are not familiar with them; the section can be skipped if the reader is familiar with them.

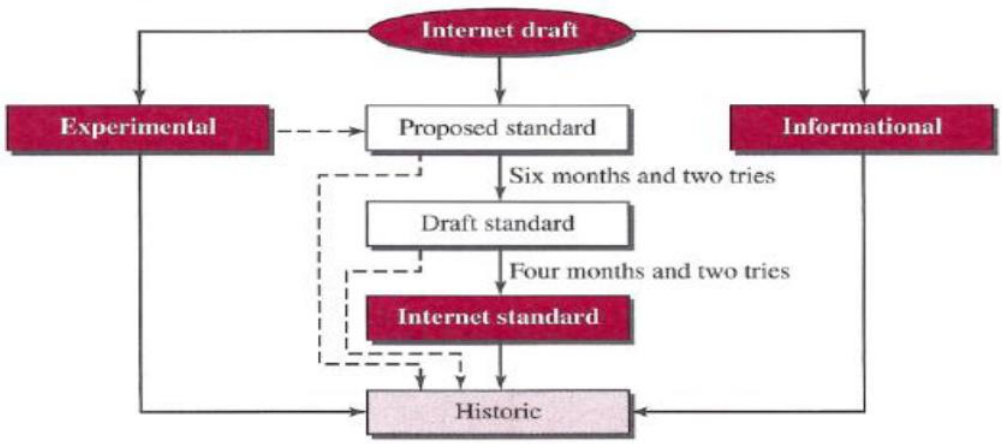
INTERNET STANDARDS

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 six-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.

Maturity Levels

An RFC, during its lifetime, falls into one of six *maturity levels:* proposed standard, draft standard, Internet standard, historic, experimental, and informational. *Proposed Standard.* A proposed standard is a specification that is stable, well understood, and of sufficient interest to the Internet community. At this level, the specification is usually tested and implemented by several different groups.

Maturity levels of an RFC



Draft Standard. A proposed standard is elevated to draft standard status after at least two successful independent and interoperable implementations. Barring difficulties, a draft standard, with modifications if specific problems are encountered, normally becomes an Internet standard.

Internet Standard. A draft standard reaches Internet standard status after demonstrations of successful implementation.

Historic The historic RFCs are significant from a historical perspective. They either have been superseded by later specifications or have never passed the necessary maturity levels to become an Internet standard.

Experimental An RFC classified as experimental describes work related to an experimental situation that does not affect the operation of the Internet. Such an RFC should not be implemented in any functional Internet service.

Informational An RFC classified as informational contains general, historical, or tutorial information related to the Internet. It is usually written by someone in a non-Internet organization, such as a vendor.

Requirement Levels

RFCs are classified into five *requirement levels:* required, recommended, elective, limited use, and not recommended.

Required An RFC is labeled *required* if it must be implemented by all Internets systems to achieve minimum conformance. For example, IF and ICMP are required protocols.

Recommended An RFC labeled recommended is not required for minimum conformance; it is recommended because of its usefulness. For example, FTP and TELNET are recommended protocols.

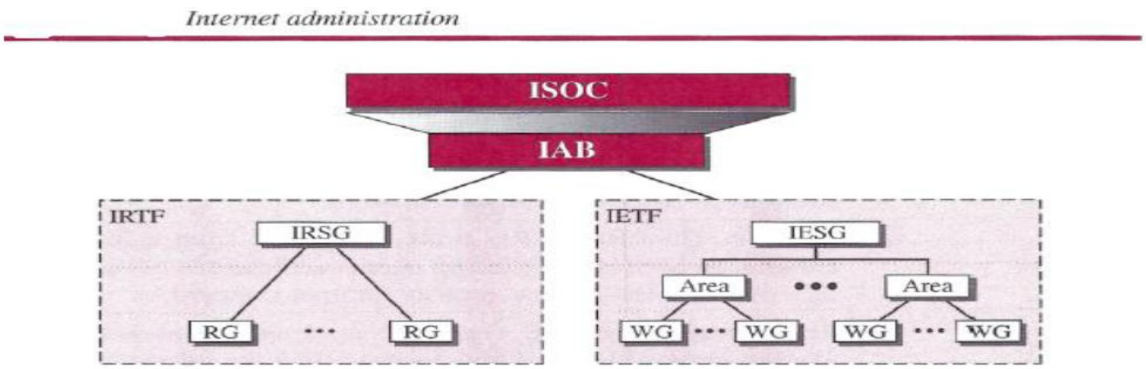
Elective An RFC labeled elective is not required and not recommended. However, a system can use it for its own benefit.

Limited Use An RFC labeled limited use should be used only in limited situations. Most of the experimental RFCs fall under this category.

Not Recommended *An* RFC labeled not recommended is inappropriate for general use. Normally a historic (deprecated) RFC may fall under this category.

INTERNET ADMINISTRATION

The Internet, with its roots primarily in the research domain, has evolved and gained a broader user base with significant commercial activity. Various groups that coordinate Internet issues have guided this growth and development. Appendix G gives the addresses, e-rnail addresses, and telephone numbers for some of these groups. Shows the general organization of Internet administration. E-rnail addresses and telephone numbers for some of these groups. Below figure
shows the general organization of Internet administration.



Isoc

The Internet Society (ISOC) is an international, nonprofit organization formed in 1992 to provide support for the Internet standards process. ISOC accomplishes this through maintaining and supporting other Internet administrative bodies such as lAB, IETF,IRTF, and IANA (see the following sections). ISOC also promotes research and other scholarly activities relating to the Internet.

lAB

The Internet Architecture Board (lAB) is the technical advisor to the ISOC. The main purposes of the lAB are to oversee the continuing development of the *TCP/IP* Protocol Suite and to serve in a technical advisory capacity to research members of the Internet community. lAB accomplishes this through its two primary components, the Internet Engineering Task Force (IETF) and the Internet Research Task Force (IRTF). Another responsibility of the lAB is the editorial management of the RFCs, described earlier. lAB is also the external liaison between the Internet and other standards organizations and forums.

JETF

The Internet Engineering Task Force (IETF) is a forum of working groups managed by the Internet Engineering Steering Group (IESG). IETF is responsible for identifying operational problems and proposing solutions to these problems. IETF also develops and reviews specifications intended as Internet standards. The working groups are collected into areas, and each area concentrates on a specific topic. Currently nine areas have been defined. The areas include applications, protocols, routing, network management next generation (lPng), and security.

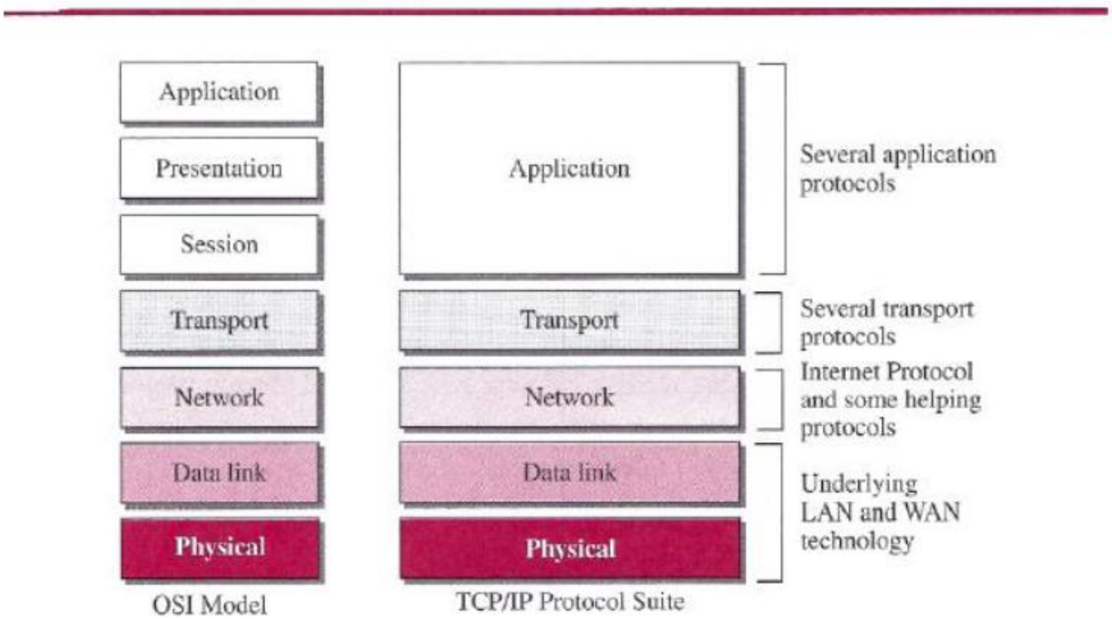
JRTF

The Internet Research Task Force (IRTF) is a forum of working groups managed by the Internet Research Steering Group (IRSG). IRTF focuses on long-term research topics related to Internet protocols, applications, architecture, and technology.

COMPARISION OF OSI AND TCP/IP REFERENCE MODEL

When we compare the two models, we find that two layers, session and presentation, are missing from the *TCP/IP* protocol suite. These two layers were not added to the *TCP/IP* protocol suite after the publication of the OSI model. The application layer in the suite is usually considered to be the combination of three layers in the OSI model.

TCP/IP and OSI model



Two reasons were mentioned for this decision. First, *TCP/IP* has more than one transport-layer protocol. Some of the functionalities of the session layer are available in some of the transport-layer protocols. Second, the application layer is not only one piece of software. Many Applications can be developed at this layer. If some of the functionalities mentioned in the session and presentation layers are needed for a particular application, they can be included in the development of that piece of software.

Lack of OSI Model's Success

The OSI model appeared after the *TCP/IP* protocol suite. Most experts were at first excited and thought that the *TCP/IP* protocol would be fully replaced by the OSI model. This did not happen for several reasons, but we describe only three, which are agreed upon by all experts in the field. First, OSI was completed when *TCP/IP* was fully in place and a lot of time and money had been spent on the suite; changing it would cost a lot. Second, some layers in the OSI model were never fully defined. For example, although the services provided by the presentation and the session layers were listed in the document, actual protocols for these two layers were not fully defined, nor were they fully described, and the corresponding software was not fully developed. Third, when OSI was implemented by an organization in a different application, it did not show a high enough level of performance to entice the Internet authority to switch from the *TCP/IP* protocol suite to the OSI model.

PHYSICAL LAYER

One of the major functions of the physical layer is to move data in the form of electromagnetic signals across a transmission medium. Whether you are collecting numerical statistics from another computer, sending animated pictures from a design workstation, or causing a bell to ring at a distant control center, you are working with the transmission of **data** across network connections. Generally, the data usable to a person or application are not in a form that can be transmitted over a network. For example, a photograph must first be changed to a form that transmission media can accept. Transmission media work by conducting energy along a physical path. For transmission, data needs to be changed to **signals.**

COMPUTER NETWORK SECURITY

During initial days of internet, its use was limited to military and universities for research and development purpose. Later when all networks merged together and formed internet, the data used to travel through public transit network. Common people may send the data that can be highly sensitive such as their bank credentials, username and passwords, personal documents, online shopping details, or confidential documents.

All security threats are intentional i.e. they occur only if intentionally triggered. Security threats can be divided into the following categories:

Interruption

Interruption is a security threat in which availability of resources is attacked. For example, a user is unable to access its web-server or the web-server is hijacked.

Privacy-Breach

In this threat, the privacy of a user is compromised. Someone, who is not the authorized person is accessing or intercepting data sent or received by the original authenticated user.

Integrity

This type of threat includes any alteration or modification in the original context of communication. The attacker intercepts and receives the data sent by the sender and the attacker then either modifies or generates false data and sends to the receiver. The receiver receives the data assuming that it is being sent by the original Sender.

Authenticity

This threat occurs when an attacker or a security violator poses as a genuine person and accesses the resources or communicates with other genuine users.

No technique in the present world can provide 100% security. But steps can be taken to secure data while it travels in unsecured network or internet. The most widely used technique is Cryptography.

Cryptography is a technique to encrypt the plain-text data which makes it difficult to understand and interpret. There are several cryptographic algorithms available present day as described below:

- Secret Key
- Public Key
- Message Digest

Secret Key Encryption

Both sender and receiver have one secret key. This secret key is used to encrypt the data at sender's end. After the data is encrypted, it is sent on the public domain to the receiver. Because the receiver knows and has the Secret Key, the encrypted data packets can easily be decrypted.
Example of secret key encryption is Data Encryption Standard (DES). In Secret Key encryption, it is required to have a separate key for each host on the network making it difficult to manage.

Public Key Encryption

In this encryption system, every user has its own Secret Key and it is not in the shared domain. The secret key is never revealed on public domain. Along with secret key, every user has its own but public key. Public key is always made public and is used by Senders to encrypt the data. When the user receives the encrypted data, he can easily decrypt it by using its own Secret Key.
Example of public key encryption is Rivest-Shamir-Adleman (RSA)..

Message Digest

In this method, actual data is not sent; instead a hash value is calculated and sent. The other end user, computes its own hash value and compares with the one just received. If both hash values are matched, then it is accepted; otherwise rejected.
Example of Message Digest is MD5 hashing. It is mostly used in authentication where user password is cross checked with the one saved on the server.

Signals

When data is sent over physical medium, it needs to be first converted into electromagnetic signals. Data itself can be analog such as human voice, or digital such as file on the disk. Both analog and digital data can be represented in digital or analog signals.

Digital Signals

Digital signals are discrete in nature and represent sequence of voltage pulses. Digital signals are used within the circuitry of a computer system.

Analog Signals

Analog signals are in continuous wave form in nature and represented by continuous electromagnetic waves.

Transmission Impairment

When signals travel through the medium, they tend to deteriorate. This may have many reasons as given:

Attenuation

For the receiver to interpret the data accurately, the signal must be sufficiently strong. When the signal passes through the medium, it tends to get weaker. As it covers distance, it loses strength.

Dispersion

As signal travels through the media, it tends to spread and overlaps. The amount of dispersion depends upon the frequency used.

Delay distortion

Signals are sent over media with pre-defined speed and frequency. If the signal speed and frequency do not match, there are possibilities that signal reaches destination in arbitrary fashion. In digital media, this is very critical that some bits reach earlier than the previously sent ones.

Noise

Random disturbance or fluctuation in analog or digital signal is said to be Noise in signal, which may distort the actual information being carried. Noise can be characterized in one of the following class:

Thermal Noise

Heat agitates the electronic conductors of a medium which may introduce noise in the media. Up to a certain level, thermal noise is unavoidable.

Intermodulation

When multiple frequencies share a medium, their interference can cause noise in the medium. Intermodulation noise occurs if two different frequencies are sharing a medium and one of them has excessive strength or the component itself is not functioning properly, then the resultant frequency may not be delivered as expected.

Crosstalk

This sort of noise happens when a foreign signal enters into the media. This is because signal in one medium affects the signal of second medium.

Impulse

This noise is introduced because of irregular disturbances such as lightening, electricity, short-circuit, or faulty components. Digital data is mostly affected by this sort of noise.

Channel Capacity

The speed of transmission of information is said to be the channel capacity. We count it as data rate in digital world. It depends on numerous factors such as:

Bandwidth: The physical limitation of underlying media.

- Error-rate: Incorrect reception of information because of noise.
- Encoding: The number of levels used for signaling.

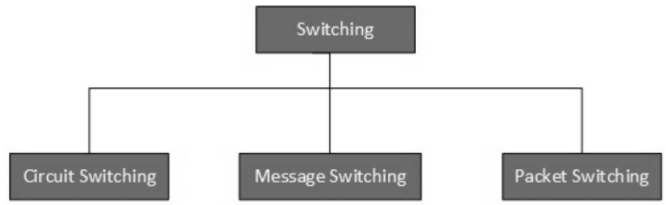
Multiplexing

Multiplexing is a technique to mix and send multiple data streams over a single medium. This technique requires system hardware called multiplexer (MUX) for multiplexing the streams and sending them on a medium, and de-multiplexer (DMUX) which takes information from the medium and distributes to different destinations.

Switching

Switching is a mechanism by which data/information sent from source towards destination which are not directly connected. Networks have interconnecting devices, which receives data from directly connected sources, stores data, analyze it and then forwards to the next interconnecting device closest to the destination.

Switching can be categorized as:



DIGITAL TRANSMISSION

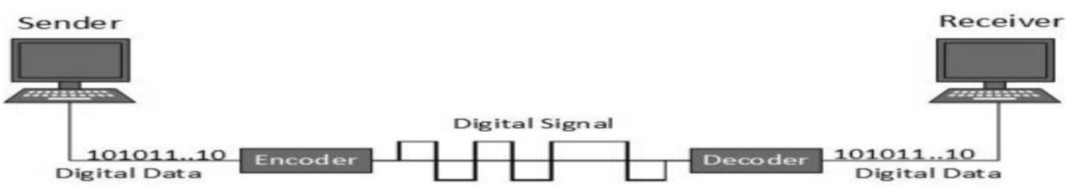
Data or information can be stored in two ways, analog and digital. For a computer to use the data, it must be in discrete digital form. Similar to data, signals can also be in analog and digital form. To transmit data digitally, it needs to be first converted to digital form.

Digital-to-Digital Conversion

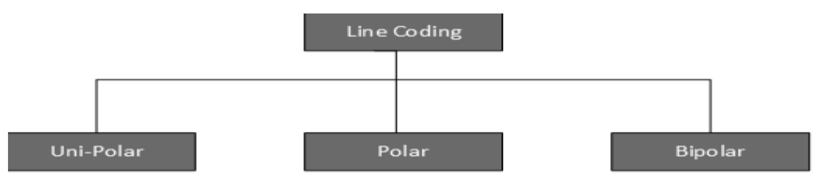
This section explains how to convert digital data into digital signals. It can be done in two ways, line coding and block coding. For all communications, line coding is necessary whereas block coding is optional.

Line Coding

The process for converting digital data into digital signal is said to be Line Coding. Digital data is found in binary format. It is represented (stored) internally as series of 1s and 0s.

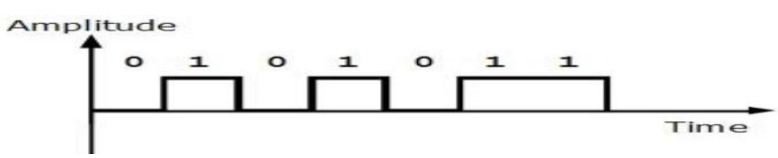


Digital signal is denoted by discreet signal, which represents digital data. There are three types of line coding schemes available:



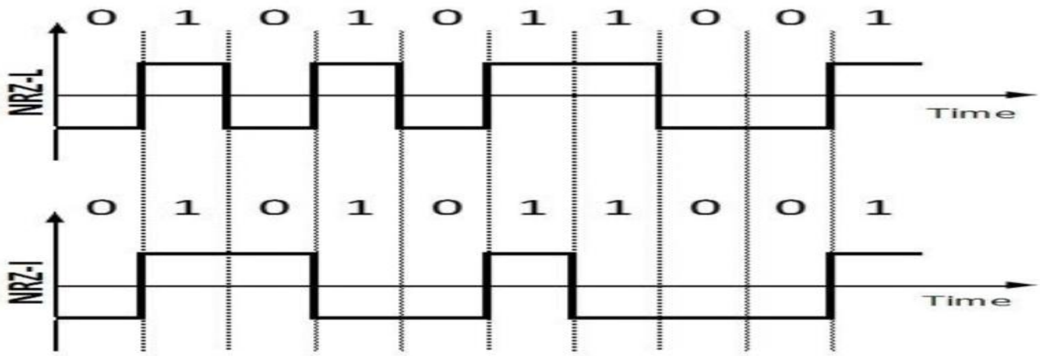
Unipolar Encoding

Unipolar encoding schemes use single voltage level to represent data. In this case, to represent binary 1, high voltage is transmitted and to represent 0, no voltage is transmitted. It is also called Unipolar-Non-return-to-zero, because there is no rest condition i.e. it either represents 1 or 0.



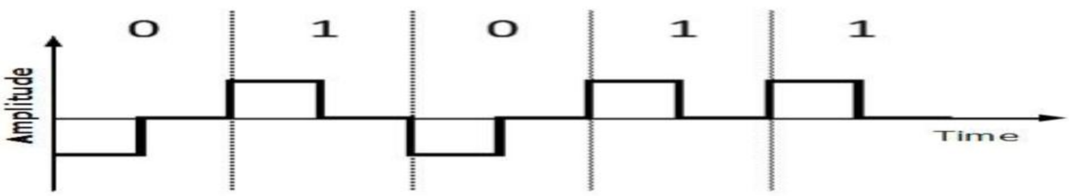
Polar Encoding

Polar encoding scheme uses multiple voltage levels to represent binary values. Polar encodings is available in four types:



Polar Non Return to Zero (Polar NRZ)

It uses two different voltage levels to represent binary values. Generally, positive voltage represents 1 and negative value represents 0. It is also NRZ because there is no rest condition. NRZ scheme has two variants: NRZ-L and NRZ-I.



NRZ-L changes voltage level at when a different bit is encountered whereas NRZ-I changes voltage when a 1 is encountered.

Return to Zero (RZ)

Problem with NRZ is that the receiver cannot conclude when a bit ended and when the next bit is started, in case when sender and receiver's clock are not synchronized.

RZ uses three voltage levels, positive voltage to represent 1, negative voltage to represent 0 and zero voltage for none. Signals change during bits not between bits.

Manchester

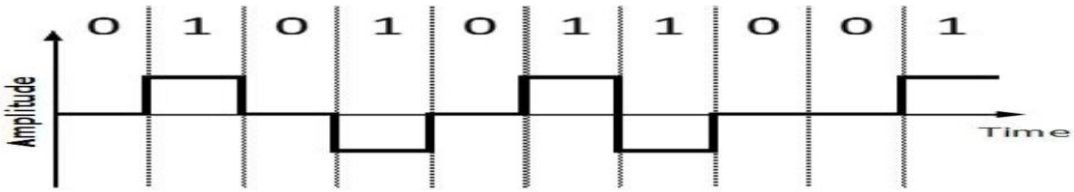
This encoding scheme is a combination of RZ and NRZ-L. Bit time is divided into two halves. It transits in the middle of the bit and changes phase when a different bit is encountered.

Differential Manchester

This encoding scheme is a combination of RZ and NRZ-I. It also transits at the middle of the bit but changes phase only when 1 is encountered.

Bipolar Encoding

Bipolar encoding uses three voltage levels, positive, negative, and zero. Zero voltage represents binary 0 and bit 1 is represented by altering positive and negative voltages.



Block Coding

To ensure accuracy of the received data frame, redundant bits are used. For example, in even-parity, one parity bit is added to make the count of 1s in the frame even. This way the original number of bits is increased. It is called Block Coding.

Block coding is represented by slash notation, mB/nB. Means, m-bit block is substituted with n-bit block where n > m. Block coding involves three steps:

1. Division
2. Substitution
3. Combination.

Analog-to-Digital Conversion

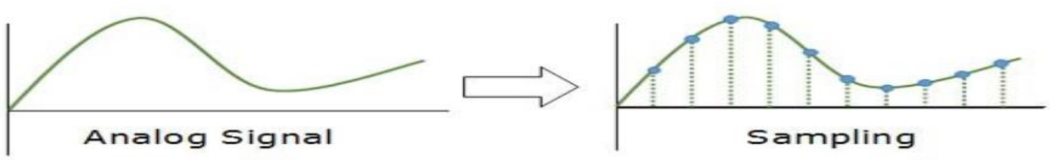
Microphones create analog voice and camera creates analog videos, which are treated is analog data. To transmit this analog data over digital signals, we need analog to digital conversion.

Analog data is a continuous stream of data in the wave form whereas digital data is discrete. To convert analog wave into digital data, we use Pulse Code Modulation (PCM).

PCM is one of the most commonly used method to convert analog data into digital form. It involves three steps:

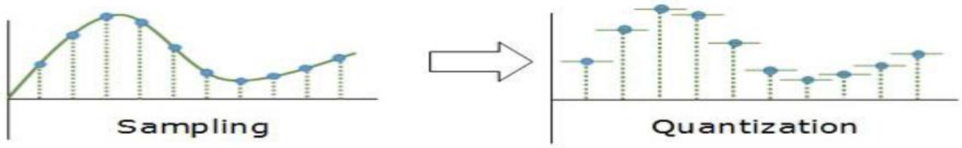
- Sampling
- Quantization
- Encoding.

Sampling



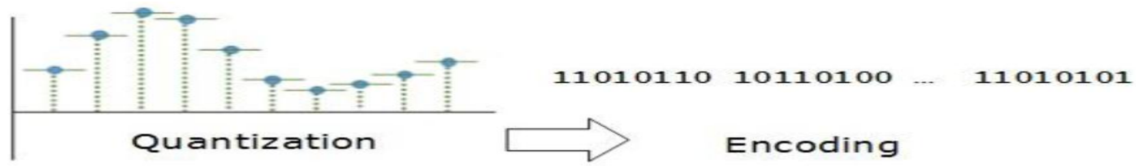
The analog signal is sampled every T interval. Most important factor in sampling is the rate at which analog signal is sampled. According to Nyquist Theorem, the sampling rate must be at least two times of the highest frequency of the signal.

Quantization:



Sampling yields discrete form of continuous analog signal. Every discrete pattern shows the amplitude of the analog signal at that instance. The quantization is done between the maximum amplitude value and the minimum amplitude value. Quantization is approximation of the instantaneous analog value.

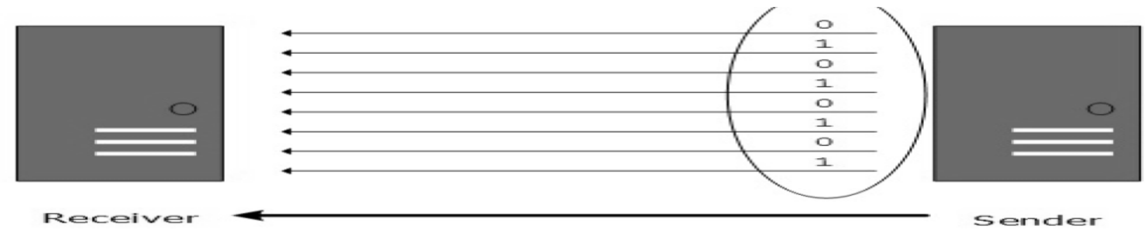
Enoding:



Transmission Modes

The transmission mode decides how data is transmitted between two computers. The binary data in the form of 1s and 0s can be sent in two different modes: Parallel and Serial.

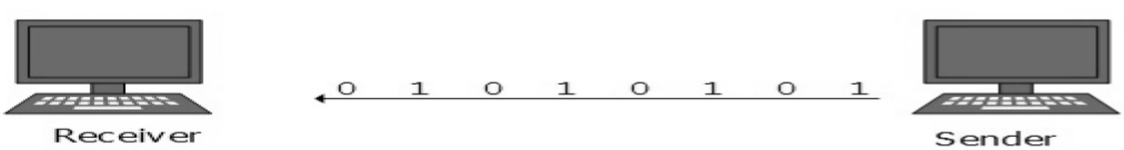
Parallel Transmission



The binary bits are organized into groups of fixed length. Both sender and receiver are connected in parallel with the equal number of data lines. Both computers distinguish between high order and low order data lines. The sender sends all the bits at once on all lines. Because the data lines are equal to the number of bits in a group or data frame, a complete group of bits (data frame) is sent in one go. Advantage of Parallel transmission is high speed and disadvantage is the cost of wires, as it is equal to the number of bits sent in parallel.

Serial Transmission

In serial transmission, bits are sent one after another in a queue manner. Serial transmission requires only one communication channel. Serial transmission can be either asynchronous or synchronous.



Asynchronous Serial Transmission

It is named so because there is no importance of timing. Data-bits have specific pattern and they help receiver recognize the start and end data bits. For example, a 0 is prefixed on every data byte and one or more 1s are added at the end.

Two continuous data-frames (bytes) may have a gap between them.

Synchronous Serial Transmission

Timing in synchronous transmission has importance as there is no mechanism followed to recognize start and end data bits. There is no pattern or prefix/suffix method. Data bits are sent in burst mode without maintaining gap between bytes (8-bits). Single burst of data bits may contain a number of bytes. Therefore, timing becomes very important.

It is up to the receiver to recognize and separate bits into bytes. The advantage of synchronous transmission is high speed, and it has no overhead of extra header and footer bits as in asynchronous transmission.

ANALOG TRANSMISSION

To send the digital data over an analog media, it needs to be converted into analog signal. There can be two cases according to data formatting.

Bandpass: The filters are used to filter and pass frequencies of interest. A bandpass is a band of frequencies which can pass the filter.

Low-pass: Low-pass is a filter that passes low frequencies signals.

When digital data is converted into a bandpass analog signal, it is called digital-to-analog conversion. When low-pass analog signal is converted into bandpass analog signal, it is called analog-to-analog conversion.

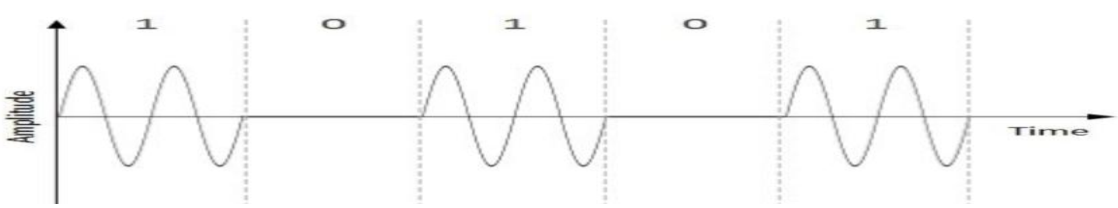
Digital-to-Analog Conversion

When data from one computer is sent to another via some analog carrier, it is first converted into analog signals. Analog signals are modified to reflect digital data.

An analog signal is characterized by its amplitude, frequency, and phase. There are three kinds of digital-to-analog conversions:

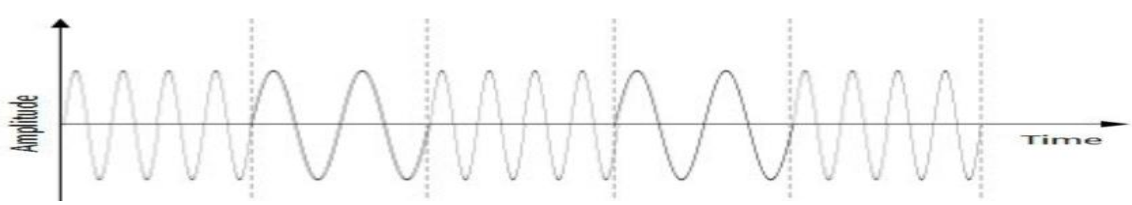
Amplitude Shift Keying

In this conversion technique, the amplitude of analog carrier signal is modified to reflect binary data.



When binary data represents digit 1, the amplitude is held; otherwise it is set to 0. Both frequency and phase remain same as in the original carrier signal.

Frequency Shift Keying: In this conversion technique, the frequency of the analog carrier signal is modified to reflect binary data

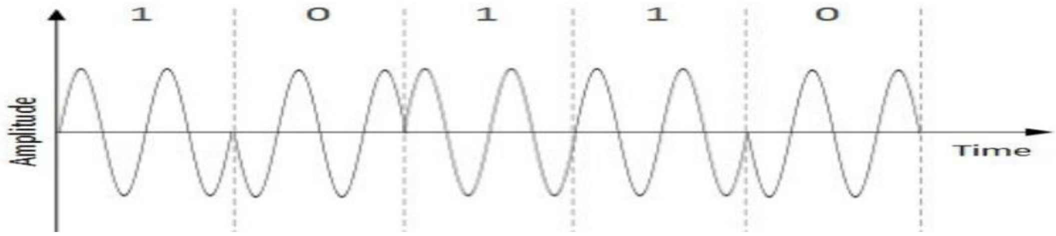


.

This technique uses two frequencies, f1 and f2. One of them, for example f1, is chosen to represent binary digit 1 and the other one is used to represent binary digit 0. Both amplitude and phase of the carrier wave are kept intact.

Phase Shift Keying

In this conversion scheme, the phase of the original carrier signal is altered to reflect the binary data.



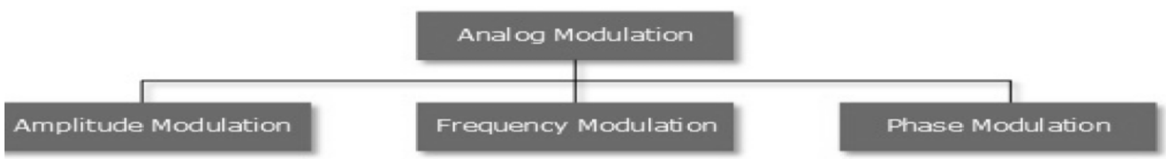
When a new binary symbol is encountered, the phase of the signal is altered. Amplitude and frequency of the original carrier signal is kept intact.

Quadrature Phase Shift Keying

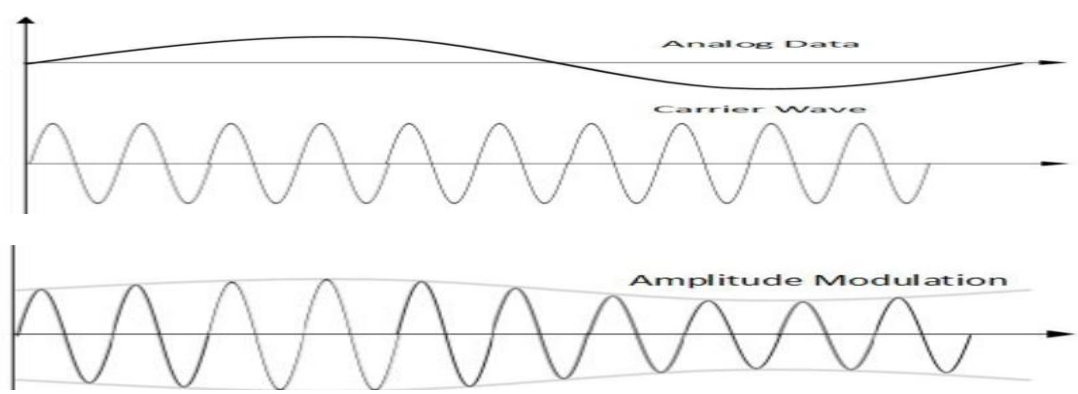
QPSK alters the phase to reflect two binary digits at once. This is done in two different phases. The main stream of binary data is divided equally into two sub-streams. The serial data is converted in to parallel in both sub-streams and then each stream is converted to digital signal using NRZ technique. Later, both the digital signals are merged together.

Analog-to-Analog Conversion

Analog signals are modified to represent analog data. This conversion is also known as Analog Modulation. Analog modulation is required when bandpass is used. Analog to analog conversion can be done in three ways:



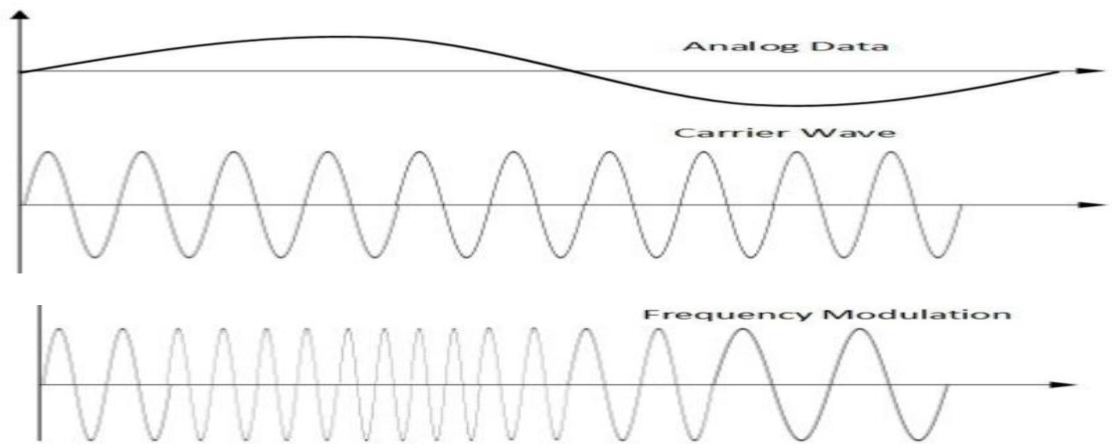
Amplitude Modulation : In this modulation, the amplitude of the carrier signal is modified to reflect the analog data.



Amplitude modulation is implemented by means of a multiplier. The amplitude of modulating signal (analog data) is multiplied by the amplitude of carrier frequency, which then reflects analog data. The frequency and phase of carrier signal remain unchanged.

Frequency Modulation

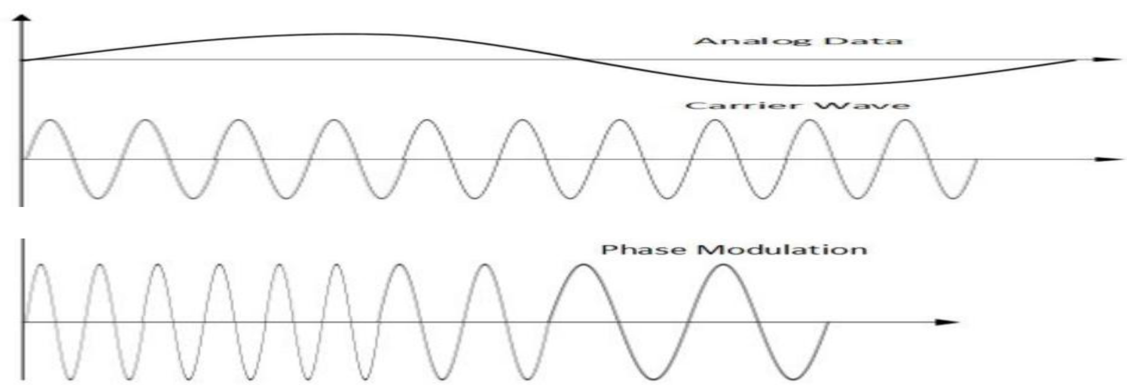
In this modulation technique, the frequency of the carrier signal is modified to reflect the change in the voltage levels of the modulating signal (analog data).



The amplitude and phase of the carrier signal are not altered.

Phase Modulation

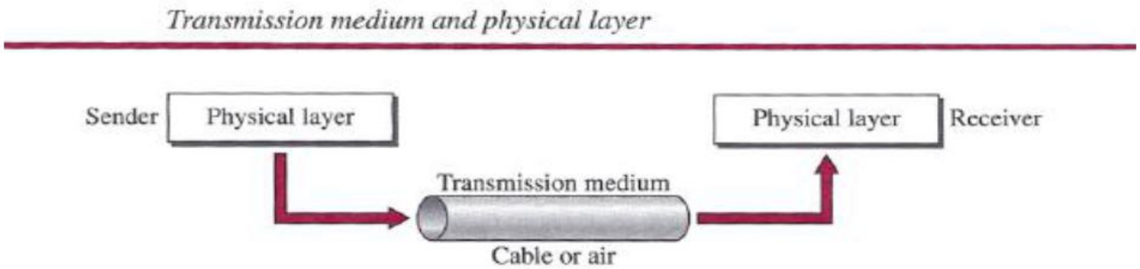
In the modulation technique, the phase of carrier signal is modulated in order to reflect the change in voltage (amplitude) of analog data signal.



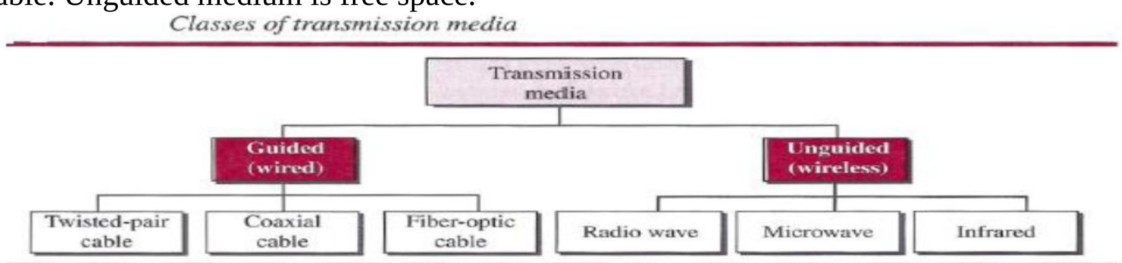
Phase modulation is practically similar to Frequency Modulation, but in Phase modulation frequency of the carrier signal is not increased. Frequency of carrier is signal is changed (made dense and sparse) to reflect voltage change in the amplitude of modulating signal.

TRANSMISSION MEDIA

Transmission media are actually located below the physical layer and are directly controlled by the physical layer. We could say that transmission media belong to layer zero. Below figure shows the position of transmission media in relation to the physical layer.



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.

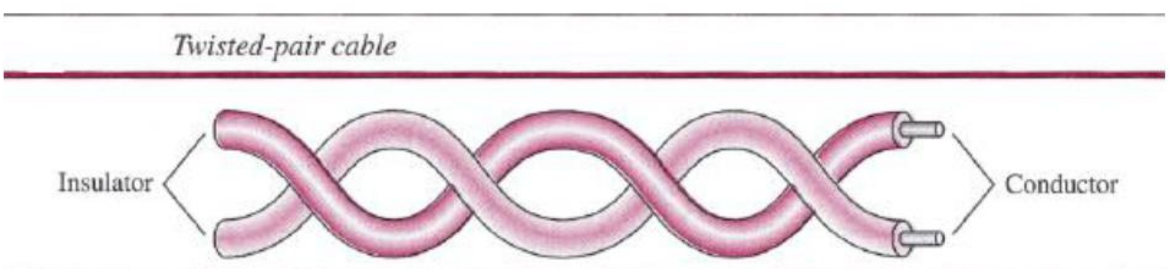


GUIDED MEDIA

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.

Twisted-Pair Cable

A twisted pair consists of two conductors (normally copper), each with its own plastic insulation, twisted together, as shown in following figure.

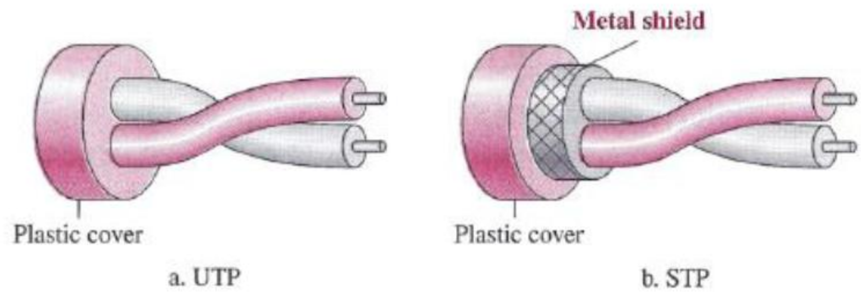


One of the wires is used to carry signals to the receiver, and the other is used only as a ground reference. The receiver uses the difference between the two.In addition to the signal sent by the sender on one of the wires, interference (noise) and crosstalk may affect both wires and create unwanted signals. If the two wires are parallel, the effect of these unwanted signals is not the same in both wires because they are at different locations relative to the noise or crosstalk sources (e.g., one is closer and the other is farther). This results in a difference at the receiver. By twisting the pairs, a balance is maintained. For example, suppose in one twist, one wire is closer to the noise source and the other is farther; in the next twist, the reverse is true. Twisting makes it probable that both wires are equally affected by external influences (noise or crosstalk). This means that the receiver, which calculates the difference between the two, receives no unwanted signals. The unwanted signals are mostly canceled out. From the above discussion, it is clear that the number of twists per unit of length (e.g., inch) has some effect on the quality of the cable.

Unshielded Versus Shielded Twisted-Pair Cable

The most common twisted-pair cable used in communications is referred to as *unshielded twisted-pair* (UTP). IBM has also produced a version of twisted-pair cable for its use, called *shielded twisted-pair* (STP). STP cable has a metal foil or braided mesh covering that encases each pair of insulated conductors. Although metal casing improves the quality of cable by preventing the penetration of noise or crosstalk, it is bulkier and more expensive. Below figure

UTP and STP cables



Categories

The Electronic Industries Association (EIA) has developed standards to classify unshielded twisted-pair cable into seven categories. Categories are determined by cable quality, with 1 as the lowest and 7 as the highest. Each EIA category is suitable for specific uses. Table below shows these categories.

Categories of unshielded twisted-pair cables

Category	*Specification*	*Data Rate (Mbps)*	*Use*
1	Unshielded twisted-pair used in telephone	< 0.1	Telephone
2	Unshielded twisted-pair originally used in T lines	2	T-1 lines
3	Improved CAT 2 used in LANs	10	LANs
4	Improved CAT 3 used in Token Ring networks	20	LANs
5	Cable wire is normally 24 AWG with a jacket and outside sheath	100	LANs

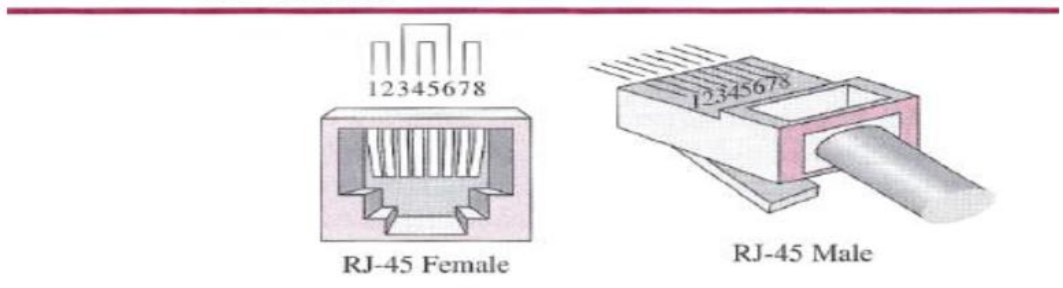
Categories of unshielded twisted-pair cables (continued)

Category	*Specification*	*Data Rate (Mbps)*	*Use*
5E	An extension to category 5 that includes extra features to minimize the crosstalk and electromagnetic interference	125	LANs
6	A new category with matched components coming from the same manufacturer. The cable must be tested at a 200-Mbps data rate.	200	LANs
7	Sometimes called *SSTP (shielded screen twisted-pair)*. Each pair is individually wrapped in a helical metallic foil followed by a metallic foil shield in addition to the outside sheath. The shield decreases the effect of crosstalk and increases the data rate.	600	LANs

Connectors

The most common UTP connector is **RJ45** (RJ stands for registered jack), as shown in below figure. The RJ45 is a keyed connector, meaning the connector can be inserted in only one way.

UTP connector



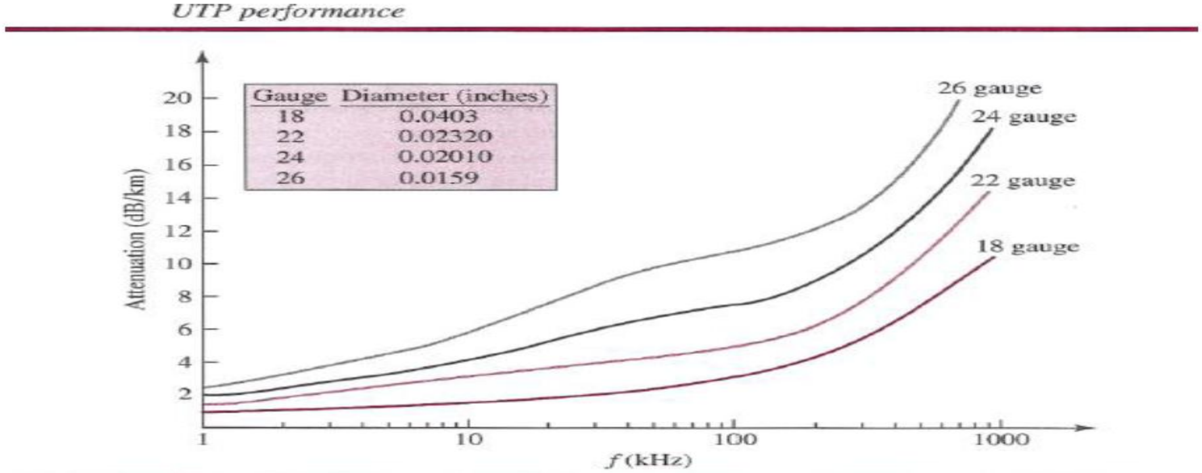
Performance

One way to measure the performance of twisted-pair cable is to compare attenuation versus frequency and distance. A twisted-pair cable can pass a wide range of frequencies. However, below figure shows that with increasing frequency, the attenuation, measured in decibels per kilometer (dB/km), sharply increases with frequencies above 100 kHz. Note that *gauge* is a measure of the thickness of the wire.

Applications

Twisted-pair cables are used in telephone lines to provide voice and data channels. The local loop-the line that connects subscribers to the central telephone office commonly consists of unshielded twisted-pair cables.

UTP performance

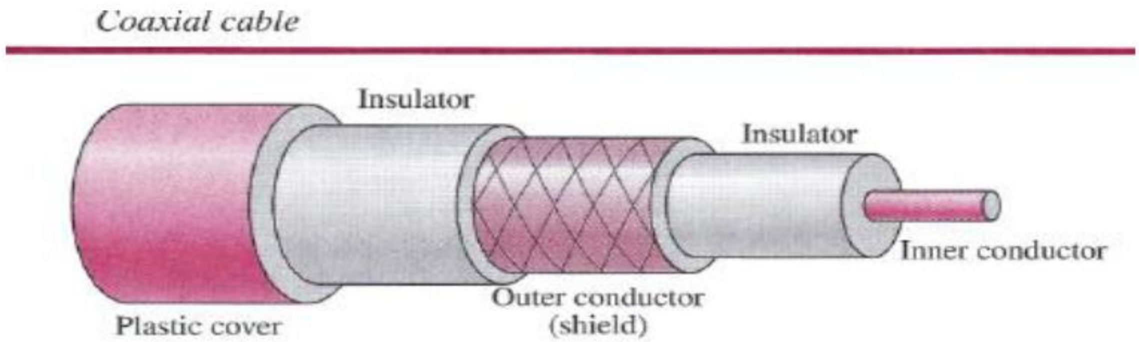


The DSL lines that are used by the telephone companies to provide high-data-rate connections also use the high-bandwidth capability of unshielded twisted-pair cables.

Local-area networks, such as lOBase-T and lOOBase-T, also use twisted-pair cables.

Coaxial Cable

Coaxial cable (or *coax)* carries signals of higher frequency ranges than those in twisted pair cable, in part because the two media are constructed quite differently. Instead of having two wires, coax has a central core conductor of solid or stranded wire (usually copper) enclosed in an insulating sheath, which is, in turn, encased in an outer conductor of metal foil, braid, or a combination of the two. The outer metallic wrapping serves both as a shield against noise and as the second conductor, which completes the circuit. This outer conductor is also enclosed in an insulating sheath, and the whole cable is protected by a plastic cover.



Coaxial Cable Standards

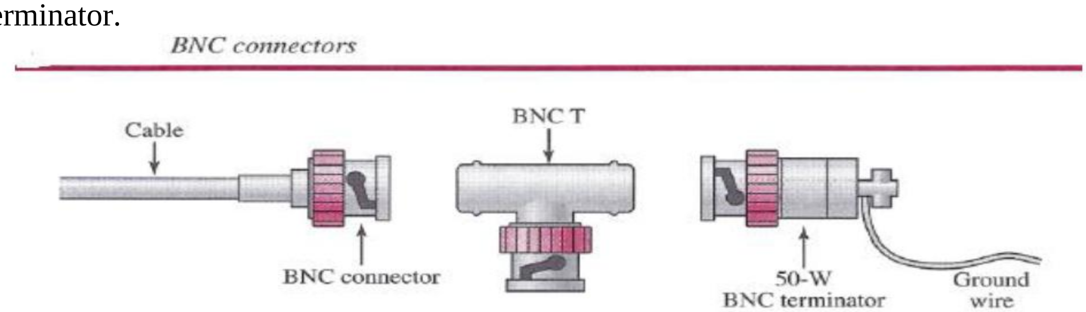
Coaxial cables are categorized by their Radio Government (RG) ratings. Each RG number denotes a unique set of physical specifications, including the wire gauge of the inner conductor, the thickness and type of the inner insulator, the construction of the shield, and the size and type of the outer casing. Each cable defined by an RG rating is adapted for a specialized function, as shown in below table.

Categories of coaxial cables

Category	*Impedance*	*Use*
RG-59	75 Ω	Cable TV
RG-58	50 Ω	Thin Ethernet
RG-11	50 Ω	Thick Ethernet

Coaxial Cable Connectors

To connect coaxial cable to devices, we need coaxial connectors. The most common type of connector used today is the Bayonet Neill-Concelman (BNC) connector. Below figure shows three popular types of these connectors: the BNC connector, the BNC T connector, and the BNC terminator.



The BNC connector is used to connect the end of the cable to a device, such as a TV set. The BNC T connector is used in Ethernet networks (see Chapter 13) to branch out to a connection to a computer or other device. The BNC terminator is used at the end of the cable to prevent the reflection of the signal.

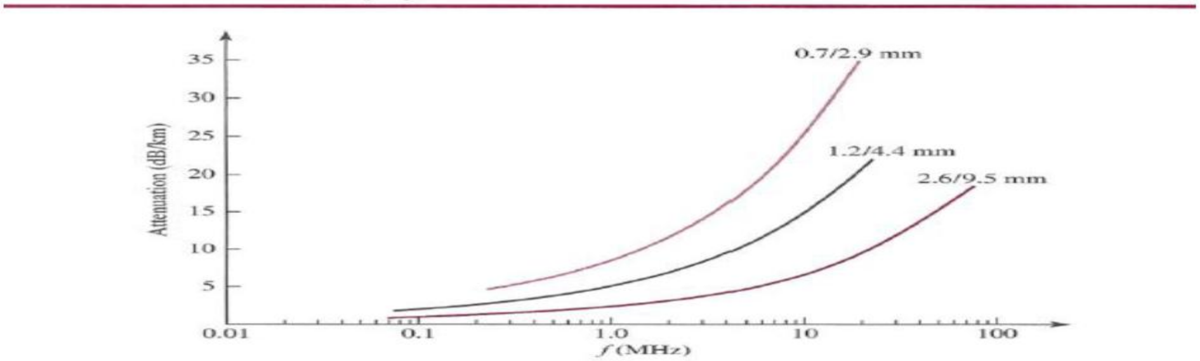
Performance

As we did with twisted-pair cable, we can measure the performance of a coaxial cable. We notice in Figure 7.9 that the attenuation is much higher in coaxial cable than in twisted-pair cable. In other words, although coaxial cable has a much higher bandwidth, the signal weakens rapidly and requires the frequent use of repeaters.

Applications

Coaxial cable was widely used in analog telephone networks where a single coaxial network could carry 10,000 voice signals. Later it was used in digital telephone networks where a single coaxial cable could carry digital data up to 600 Mbps. However, coaxial cable in telephone networks has largely been replaced today with fiber optic cable.

Coaxial cable performance

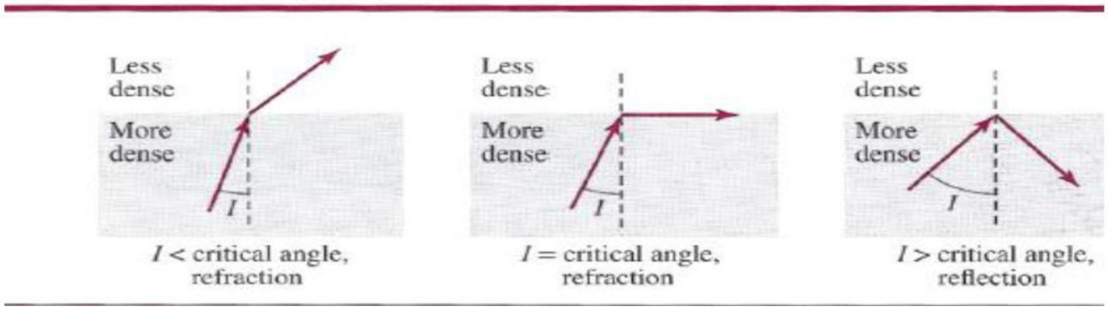


Cable TV networks also use coaxial cables. In the traditional cable TV network, the entire network used coaxial cable. Later, however, cable TV providers replaced most of the media with fiber-optic cable; hybrid networks use coaxial cable only at the network boundaries, near the consumer premises. Cable TV uses RG-59 coaxial cable. Another common application of coaxial cable is in traditional Ethernet LANs (see Because of its high bandwidth, and consequently high data rate, coaxial cable was chosen for digital transmission in early Ethernet LANs. The lOBase-2, or Thin Ethernet, uses RG-58 coaxial cable with BNC connectors to transmit data at 10 Mbps with a range of 185 m. The lOBase5, or Thick Ethernet, uses RG-ll (thick coaxial cable) to transmit 10 Mbps with a range of 5000 m. Thick Ethernet has specialized connectors.

Fiber-Optic Cable

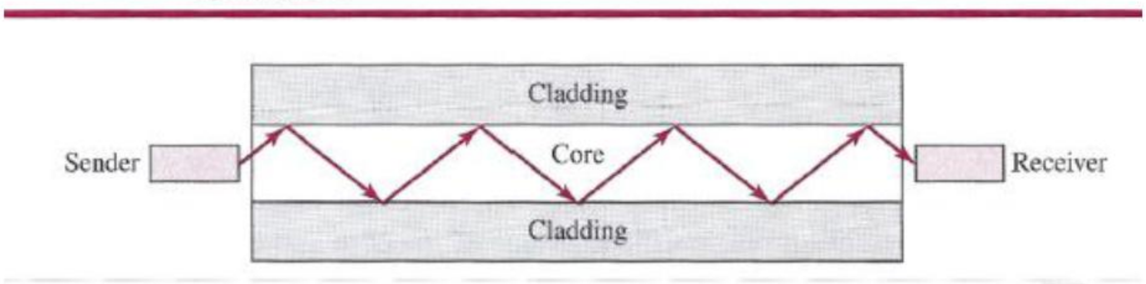
A fiber-optic cable is made of glass or plastic and transmits signals in the form of light. To understand optical fiber, we first need to explore several aspects of the nature of light. Light travels in a straight line as long as it is moving through a single uniform substance. If a ray of light traveling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

Bending of light ray



substance . Note that 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. The difference in density of the two materials must be such that a beam of light moving through the core is reflected off the cladding instead of being refracted into it. See below figure.

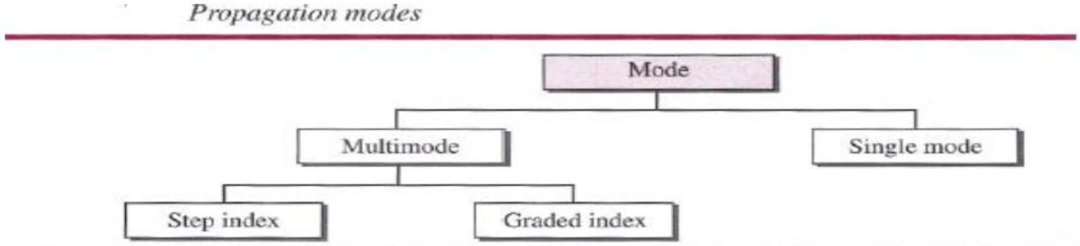
Optical fiber



Propagation Modes

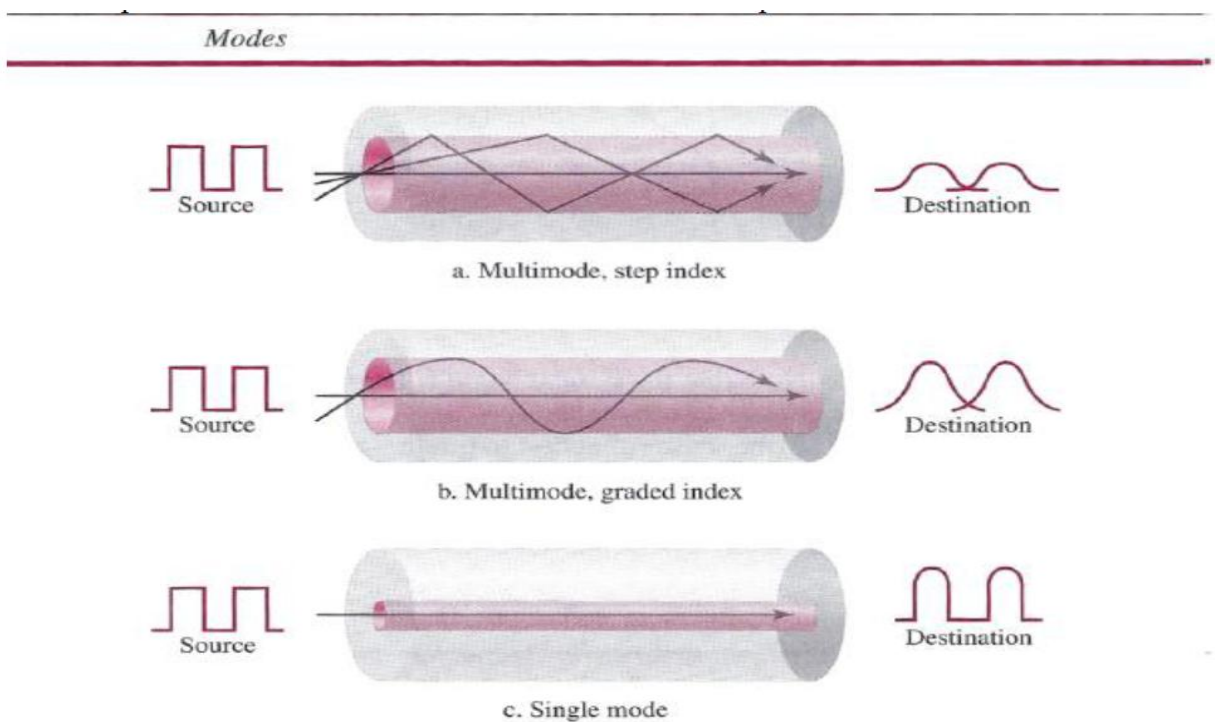
Current technology supports two modes (multimode and single mode) for propagating light along optical channels, each requiring fiber with different physical characteristics. Multimode can be implemented in two forms: step-index or graded-index .

Propagation modes



Multimode

Multimode is so named because multiple beams from a light source move through the core in different paths. How these beams move within the cable depends on the structure of the core.



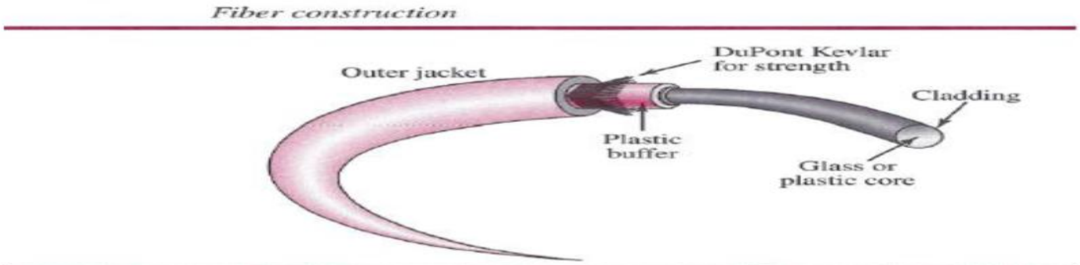
In **multimode step-index fiber,** the density of the core remains constant from the center to the edges. A beam of light moves through this constant density in a straight line until it reaches the interface of the core and the cladding. A second type of fiber, called **multimode graded-index fiber,** decreases this distortion of the signal through the cable. The word *index* here refers to the index of refraction. As we saw above, the index of refraction is related to density. *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 lowers 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.

Fiber Sizes

Optical fibers are defined by the ratio of the diameter of their core to the diameter of their cladding, both expressed in micrometers. The common sizes are shown in below table. Note that the last size listed is for single-mode only.

Cable Composition

Following figure shows the composition of a typical fiber-optic cable. The outer jacket is made of either pvc or Teflon. Inside the jacket are Kevlar strands to strengthen the cable. Kevlar is a strong material used in the fabrication of bulletproof vests. Below the Kevlar is another plastic coating to cushion the fiber. The fiber is at the center of the cable, and it consists of cladding and core.

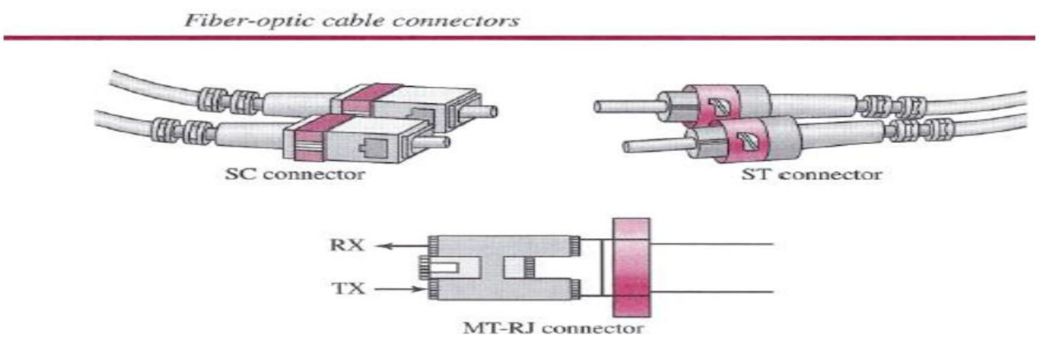


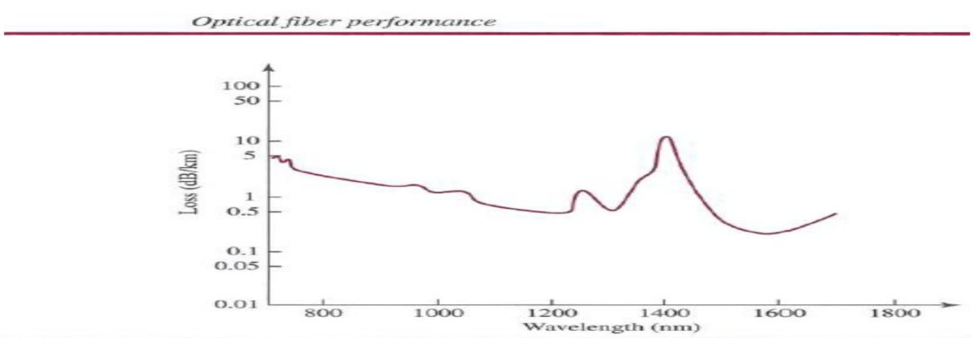
Fiber-Optic Cable Connectors

There are three types of connectors for fiber-optic cables, as shown in below figure. The subscriber channel (SC) connector is used for cable TV. It uses a push/pull locking system. The straight-tip (ST) connector is used for connecting cable to networking devices. It uses a bayonet locking system and is more reliable than sc. MT-RJ is a connector that is the same size as RJ45.

Performance

The plot of attenuation versus wavelength in Figure 7.16 shows a very interesting phenomenon in fiber-optic cable. Attenuation is flatter than in the case of twisted-pair cable and coaxial cable. The performance is such that we need fewer (actually one tenth as many) repeaters when we use fiber-optic cable.





Applications

Fiber-optic cable is often found in backbone networks because its wide bandwidth is cost-effective. Today, with wavelength-division multiplexing (WDM), we can transfer data at a rate of 1600 Gbps. The SONET network that we discuss in Chapter 14 provides such a backbone.

Some cable TV companies use a combination of optical fiber and coaxial cable, thus creating a hybrid network. Optical fiber provides the backbone structure while coaxial cable provides the

connection to the user premises. This is a cost-effective configuration since the narrow bandwidth requirement at the user end does not justify the use of optical fiber.
Local-area networks such as 100Base-FX network (Fast Ethernet) and 1000Base-X also use fiber-optic cable.

Advantages and Disadvantages of Optical Fiber

Advantages

Fiber-optic cable has several advantages over metallic cable (twisted-pair or coaxial).

- Higher bandwidth. Fiber-optic cable can support dramatically higher bandwidths (and hence data rates) than either twisted-pair or coaxial cable. Currently, data rates and bandwidth utilization over fiber-optic cable are limited not by the medium but by the signal generation and reception technology available.
- Less signal attenuation. Fiber-optic transmission distance is significantly greater than that of other guided media. A signal can run for 50 km without requiring regeneration. We need repeaters every 5 km for coaxial or twisted-pair cable.
- Immunity to electromagnetic interference. Electromagnetic noise cannot affect fiber-optic cables.
- Resistance to corrosive materials. Glass is more resistant to corrosive materials than copper.
- Light weight. Fiber-optic cables are much lighter than copper cables.
- Greater immunity to tapping. Fiber-optic cables are more immune to tapping than copper cables. Copper cables create antenna effects that can easily be tapped.

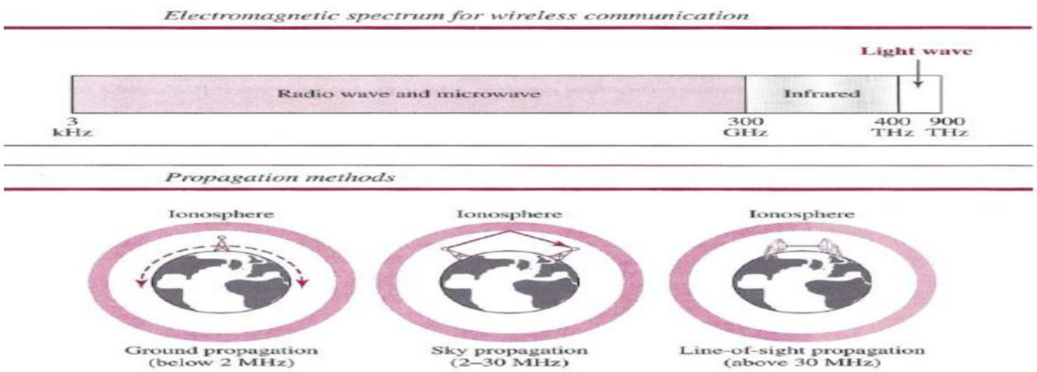
Disadvantages

There are some disadvantages in the use of optical fiber.

- Installation and maintenance. Fiber-optic cable is a relatively new technology. Its installation and maintenance require expertise that is not yet available everywhere. o Unidirectional light propagation. Propagation of light is unidirectional. If we need bidirectional communication, two fibers are needed.
- Cost. The cable and the interfaces are relatively more expensive than those of other guided media. If the demand for bandwidth is not high, often the use of optical fiber cannot be justified.

UNGUIDED MEDIA: WIRELESS

Unguided medium 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. Below 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 the destination in several ways: ground propagation, sky propagation, and line-of-sight propagation, as shown in below figure.



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. **In 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-of-sight propagation,** very high-frequency signals are transmitted in straight lines directly from antenna to antenna.

The section of the electromagnetic spectrum defined as radio waves and microwaves is divided into eight ranges, called *bands,* each regulated by government authorities. These bands are rated from *very low frequency* (VLF) to *extremely high frequency* (EHF). Below table lists these bands, their ranges, propagation methods, and some applications .

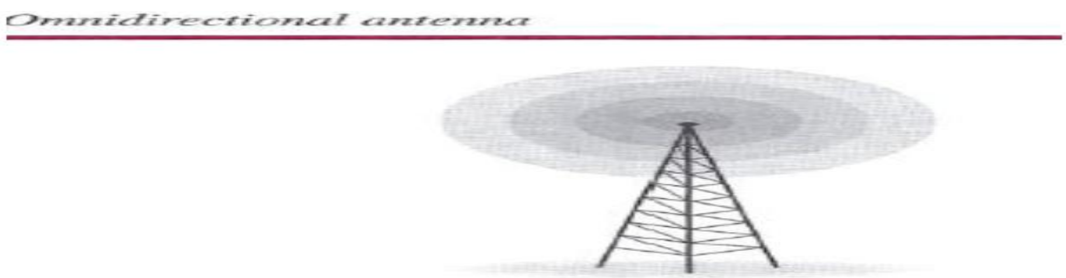
Band	*Range*	*Propagation*	*Application*
middle frequency (MF)	300 kHz–3 MHz	Sky	AM radio
high frequency (HF)	3–30 MHz	Sky	Citizens band (CB), ship/aircraft
very high frequency (VHF)	30–300 MHz	Sky and line-of-sight	VHF TV, FM radio
ultrahigh frequency (UHF)	300 MHz–3 GHz	Line-of-sight	UHF TV, cellular phones, paging, satellite
superhigh frequency (SF)	3–30 GHz	Line-of-sight	Satellite
extremely high frequency (EHF)	30–300 GHz	Line-of-sight	Radar, satellite
Band	*Range*	*Propagation*	*Application*
very low frequency (VLF)	3–30 kHz	Ground	Long-range radio navigation
low frequency (LF)	30–300 kHz	Ground	Radio beacons and navigational locators

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 I and 300 GHz are called microwaves. However, the behavior of the waves, rather than the frequencies, is a better criterion for classification. Radio waves, for the most part, are omnidirectional. When an antenna transmits radio waves, 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, too. The radio waves transmitted by one antenna are susceptible to interference by another antenna that may send signals using the same frequency or band. Radio waves, particularly those waves that propagate in the sky mode, can travel long distances. This makes radio waves a good candidate for long-distance broadcasting such as AM radio.

Omni directional Antenna

Radio waves use omni directional antennas that send out signals in all directions. Based on the wavelength, strength, and the purpose of transmission, we can have several types of antennas. Below Figure shows an omni directional antenna.



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, maritime radio, cordless phones, and paging are examples of multicasting.

Radio waves are used for multicast communications, such as radio and television, and paging systems.

Microwaves

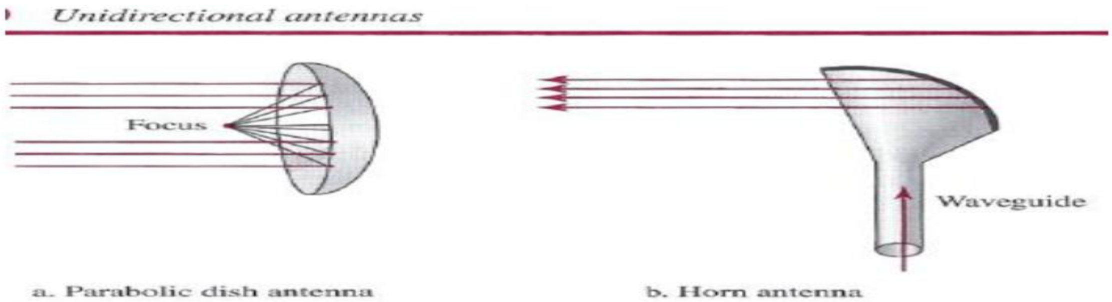
Electromagnetic waves having frequencies between 1 and 300 GHz are called microwaves. Microwaves are unidirectional. When an antenna transmits microwaves, they can be narrowly focused. This means that the sending and receiving antennas need to be aligned. The unidirectional property has an obvious advantage. 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. The curvature of the earth as well as other blocking obstacles does not allow two short towers to communicate by using microwaves. 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 subbands can be assigned, and a high data rate is possible.
- Use of certain portions of the band requires permission from authorities.

Unidirectional Antenna

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 .



A parabolic dish antenna is based on the geometry of a parabola: Every line parallel to the line of symmetry (line of sight) reflects off the curve at angles such that all the lines intersect in a common point called the focus. The parabolic dish works as a funnel, catching a wide range of waves and directing them to a common point. In this way, more of the signal is recovered than would be possible with a single-point receiver.

Outgoing transmissions are broadcast through a horn aimed at the dish. The microwaves hit the dish and are deflected outward in a reversal of the receipt path. A horn antenna looks like a gigantic scoop. Outgoing transmissions are broadcast up a stem (resembling a handle) and deflected outward in a series of narrow parallel beams by the curved head. Received transmissions are collected by the scooped shape of the horn, in a manner similar to the parabolic dish, and are deflected down into the stem.

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 phone, satellite networks, and wireless LANs

Microwaves are used for unicast communication such as cellular telephones, satellite networks, and wireless LANs.

Infrared

Infrared waves, with frequencies from 300 GHz to 400 THz (wavelengths from 1 mm to 770 nrn), 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. In addition, we cannot use infrared waves outside a building because the sun's rays contain infrared waves that can interfere with the communication.

Applications

The infrared band, almost 400 THz, has an excellent potential for data transmission. Such a wide bandwidth can be used to transmit digital data with a very high data rate. The *Infrared Data Association* elrDA), an association for sponsoring the use of infrared waves, has established standards for using these signals for communication between devices such as keyboards, mice, PCs, and printers. For example, some manufacturers provide a special port called the IrDA port that allows a wireless keyboard to communicate with a PC. The standard originally defined a data rate of 75 kbps for a distance up to 8 m. The recent standard defines a data rate of 4 Mbps.

Infrared signals defined by IrDA transmit through line of sight; the IrDA port on the keyboard needs to point to the PC for transmission to occur. Infrared signals can be used for short-range communication in a closed area using line-of-sight propagation.

1. Discuss about ISO/OSI reference model with neat sketch
2. Explain in detail about circuit switching and datagram switching with diagram
3. Briefly explain the different types of packet switching techniques with suitable networks.
4. Compare OSI model and Internet Model
5. Explain about Network Topology
6. Explain about Categories of networks
7. Explain about Cable Networks