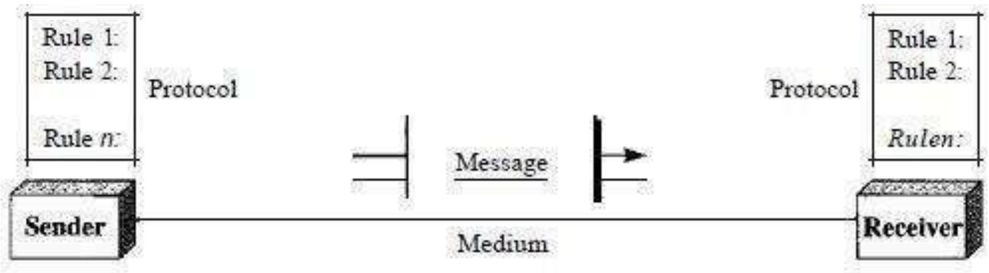
UNIT -I
Introduction to Computer Networks

1.1 Data Communication: When we communicate, we are sharing information. This sharing can be local or remote. Between individuals, local communication usually occurs face to face, while remote communication takes place over distance.

Computer Network: A computer network is a set of computers connected together for the purpose of sharing resources. The most common resource shared today is connection to the Internet. Other shared resources can include a printer or a file server. The Internet itself can be considered a computer network.

1.1.1 Components:

A data communications system has five components.



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

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

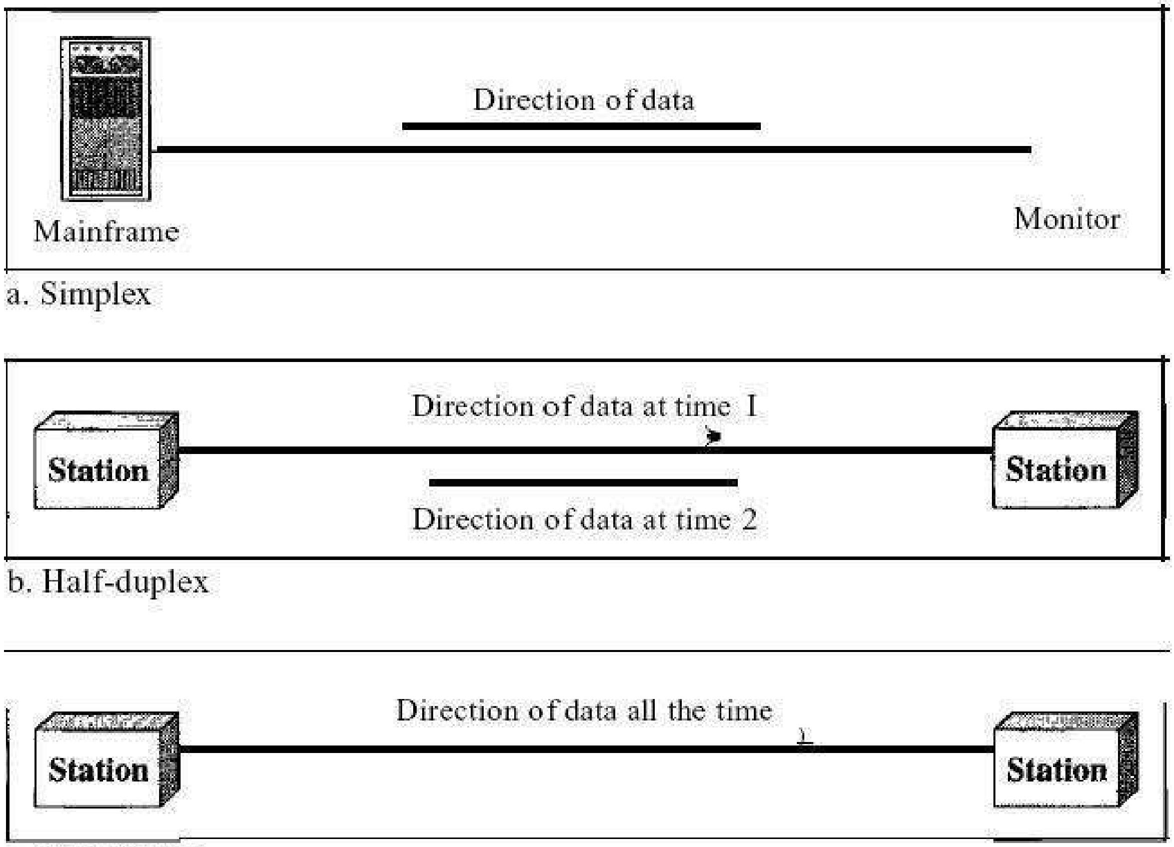
3. Receiver. The receiver is the device that receives the message. It can be a computer, workstation, telephone handset, television, and so on.

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

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

1.1.2 Data Flow

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


Direction of data
Mainframe
Monitor
Direction of data at time I
Station
Station
Direction of data at time 2
Direction of data all the time
Station
Station

a. Simplex

b. Half-duplex

c. Full·duplex

Simplex:

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

Half-Duplex:

In half-duplex mode, each station can both transmit and receive, but not at the same time. When one device is sending, the other can only receive, and vice versa The half-duplex mode is like a one-lane road with traffic allowed in both directions.

When cars are traveling in one direction, cars going the other way must wait. In a half-duplex transmission, the entire capacity of a channel is taken over by whichever of the two devices is transmitting at the time. Walkie-talkies and CB (citizens band) radios are both half-duplex systems.

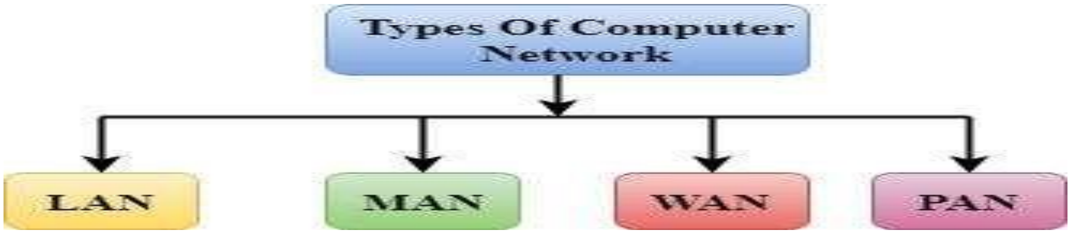
The half-duplex mode is used in cases where there is no need for communication in both directions at the same time; the entire capacity of the channel can be utilized for each direction.

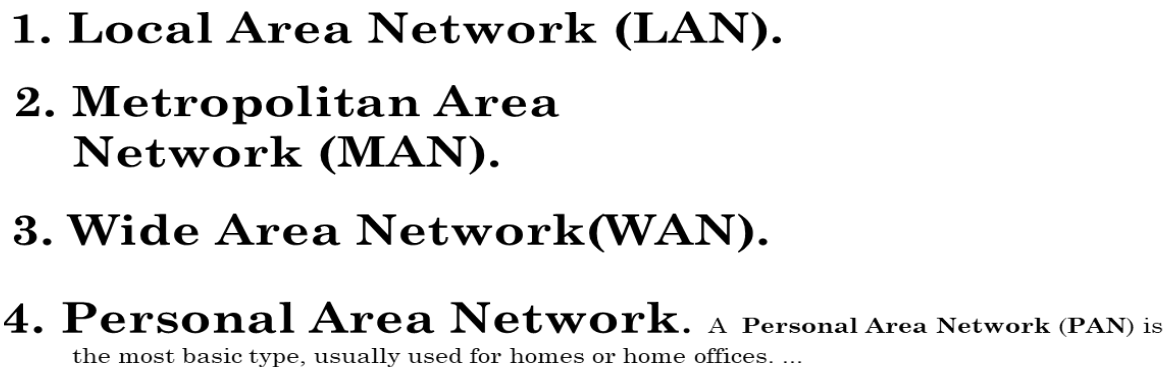
Full-Duplex:

In full-duplex both stations can transmit and receive simultaneously (see Figure c). The full-duplex mode is like a two-way street with traffic flowing in both directions at the same time. In full-duplex mode, signals going in one direction share the capacity of the link: with signals going in the other direction. This sharing can occur in two ways: Either the link must contain two physically separate transmission paths, one for sending and the other for receiving; or the capacity of the channel is divided between signals traveling in both directions. One common example of full-duplex communication is the telephone network. When two people are communicating by a telephone line, both can talk and listen at the same time. The full-duplex mode is used when communication in both directions is required all the time. The capacity of the channel, however, must be divided between the two directions.

Types of Computer Networks:

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





1. Local Area Network (LAN).

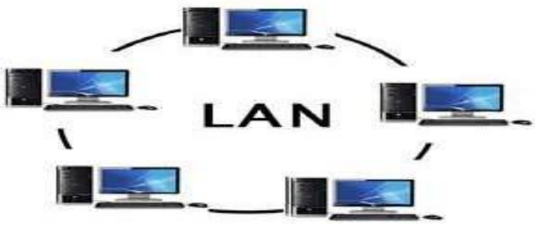
2. Metropolitan Area Network (MAN).

3. Wide Area Network(WAN).

4. Personal Area Network. A Personal Area Network (PAN) is the most basic type, usually used for homes or home offices. ...

Local Area Network (LAN) :

- Local Area Network is a group of computers connected to each other in a small area such as building, office.
- LAN is used for connecting two or more personal computers through a communication medium such a twisted pair, coaxial cable, etc.
- It is less costly as it is built with inexpensive hardware such as hubs, network adapters, and Ethernet cables.
- The data is transferred at an extremely faster rate in Local Area Network.
- Local Area Network provides higher security.



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.

* A metropolitan area network is a network that covers a larger geographic area by interconnecting a different LAN to form a larger network.
* Government agencies use MAN to connect to the citizens and private industries.
* In MAN, various LANs are connected to each other through a telephone exchange line.
* The most widely used protocols in MAN are RS-232, Frame Relay, ATM, ISDN, etc.
*It has a higher range than Local Area Network (LAN).

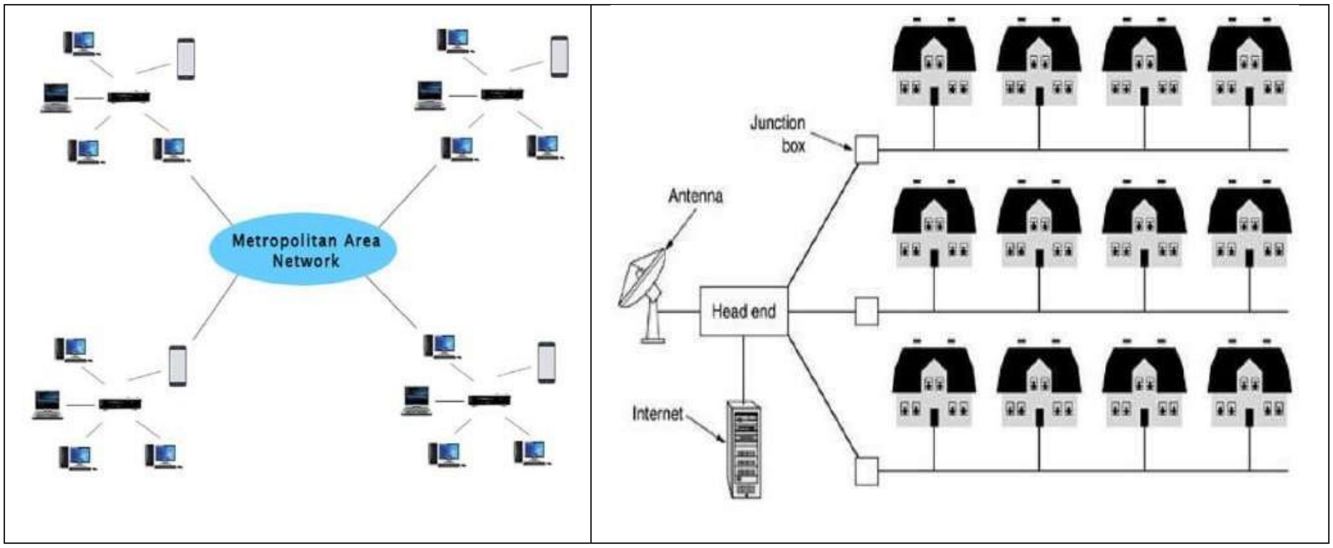


Fig.: Metropolitan area network based on cable TV.

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. 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 route 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.

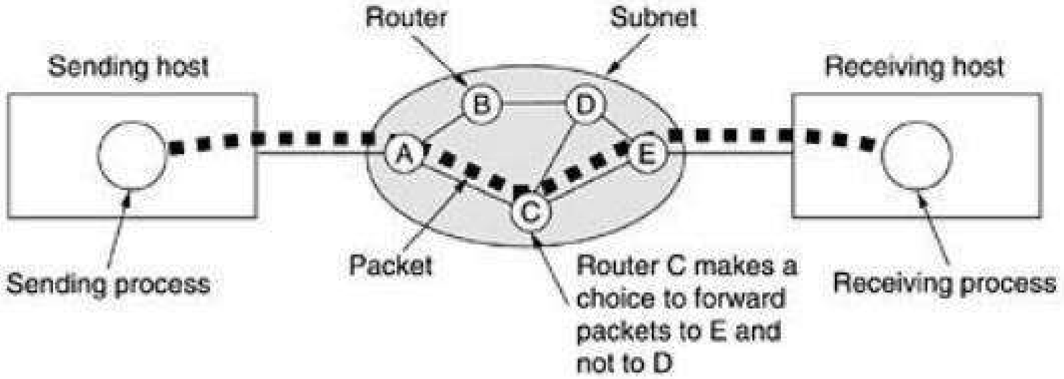
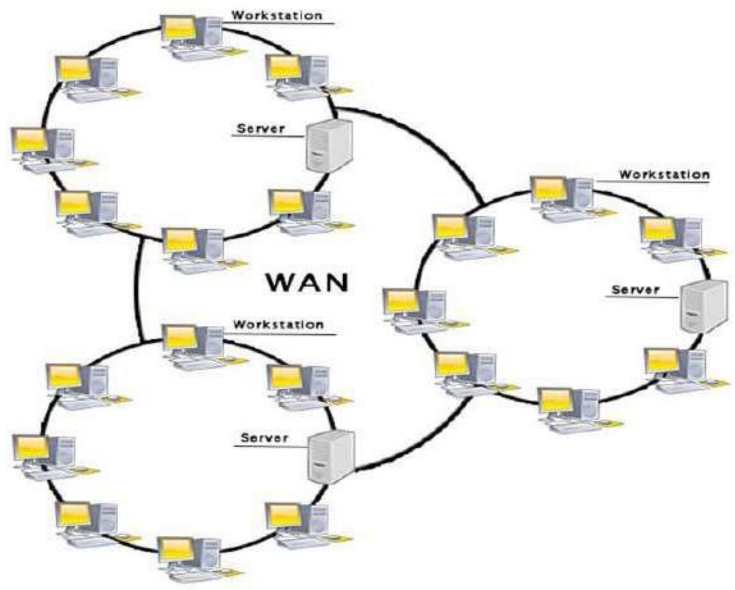


Fig.: A stream of packets from sender to receiver

* A Wide Area Network is a network that extends over a large geographical area such as states or countries.
* A Wide Area Network is quite bigger network than the LAN.
* A Wide Area Network is not limited to a single location, but it spans over a large geographical area through a telephone line, fibre optic cable or satellite links.
* The internet is one of the biggest WAN in the world.
* A Wide Area Network is widely used in the field of Business, government, and education.



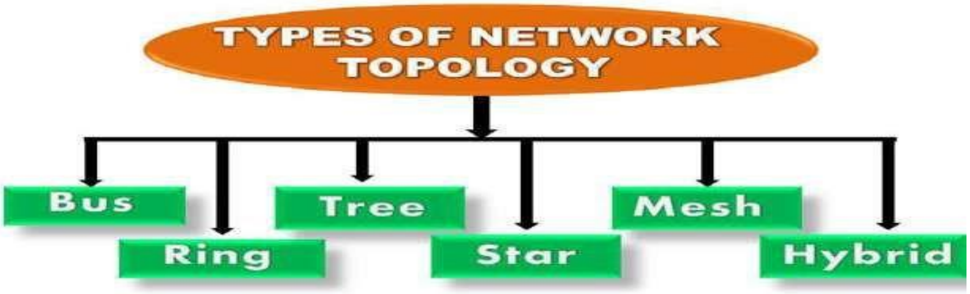
Personal Area Network (PAN):

- Personal Area Network is a network arranged within an individual person, typically within a range of 10 meters.
- Personal Area Network is used for connecting the computer devices of personal use is known as Personal Area Network.
- Thomas Zimmerman was the first research scientist to bring the idea of the Personal Area Network.
- Personal Area Network covers an area of 30 feet.
- Personal computer devices that are used to develop the personal area network are the aptop, mobile phones, media player and play stations.

Topologies

What is Topology? :

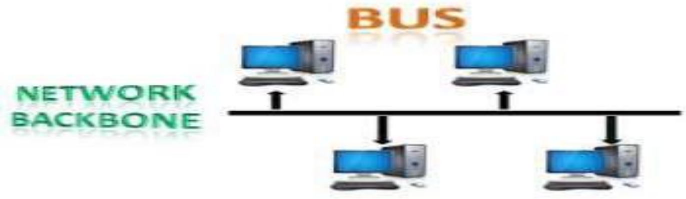
- Topology defines the structure of the network of how all the components are interconnected to each other.
- There are two types of topology: **physical and logical topology**.
- Physical topology is the geometric representation of all the nodes in a network.



1. **Bus Topology:**

- The bus topology is designed in such a way that all the stations are connected through a single cable known as a backbone cable.
- Each node is either connected to the backbone cable by drop cable or directly connected to the backbone cable
- When a node wants to send a message over the network, it puts a message over the network. All the stations available in the network will receive the message whether it has been addressed or not.
- The bus topology is mainly used in 802.3 (Ethernet) and 802.4 standard networks.
- The configuration of a bus topology is quite simpler as compared to other topologies.

- The backbone cable is considered as a **"single lane"** through which the message is broadcast to all the stations
- The most common access method of the bus topologies is **CSMA** (Carrier Sense Multiple Access).



Advantages and Disadvantages of Bus Topology:

Advantages of Bus Topology:

- It is cost effective.
- Cable required is least compared to another network topology.
- Used in small networks.
- It is easy to understand.
- Easy to expand joining two cables together.

Disadvantages of Bus Topology:

- Cables fails then whole network fails.
- If network traffic is heavy or nodes are more the performance of the network decreases.
- Cable has a limited length.

It is slower than the ring topology.

2. Ring Topology:

- Ring topology is like a bus topology, but with connected ends.
- The node that receives the message from the previous computer will retransmit to the next node.
- The data flows in one direction, i.e., it is unidirectional.
- The data flows in a single loop continuously known as an endless loop.
- It has no terminated ends, i.e., each node is connected to other node and having no termination point.
- The data in a ring topology flow in a clockwise direction.
- The most common access method of the ring topology is **token passing**.

Token passing: It is a network access method in which token is passed from one node to another node.

Token: It is a frame that circulates around the network

Working of Token passing:

- A token moves around the network, and it is passed from computer to computer until it reaches the destination
- The sender modifies the token by putting the address along with the data.
- The data is passed from one device to another device until the destination address matches. Once the token received by the destination device, then it sends the acknowledgment to the sender.
- In a ring topology, a token is used as a carrier.

Advantages and Disadvantages of Ring topology:

Advantages of Ring topology:

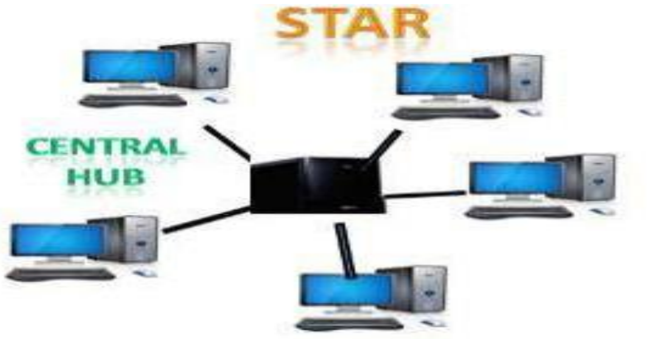
- **Network Management:** Faulty devices can be removed from the network without bringing the network down.
- **Product availability:** Many hardware and software tools for network operation and monitoring are available.
- **Cost:** Twisted pair cabling is inexpensive and easily available. Therefore, the installation cost is very low.
- **Reliable:** It is a more reliable network because the communication system is not dependent on the single host computer.

Disadvantages of Ring topology :

- **Difficult troubleshooting:** It requires specialized test equipment to determine the cable faults. If any fault occurs in the cable, then it would disrupt the communication for all the nodes.
- **Failure:** The breakdown in one station leads to the failure of the overall network.
- **Reconfiguration difficult:** Adding new devices to the network would slow down the network.
- **Delay:** Communication delay is directly proportional to the number of nodes. Adding new devices increases the communication delay.

3. Star Topology:

- Star topology is an arrangement of the network in which every node is connected to the central hub, switch or central computer.
- The central computer is known as a **server**, and the peripheral devices attached to the server are known as **clients**.
- Coaxial cable or RJ-45 cables are used to connect the computers.
- Hubs or Switches are mainly used as connection devices in a **physical star topology**.
- Star topology is the most popular topology in network implementation.



Advantages and Disadvantages of Star topology of Star topology:

Advantages of Star topology :

- **Efficient troubleshooting:** Troubleshooting is quite efficient in a star topology as compared to bus topology.In a bus topology, the manager has to inspect the kilometers of cable. In a star topology, all the stations are connected to the centralized network.
- **Limited failure:** As each station is connected to the central hub with its own cable, therefore failure in one cable will not affect the entire network.
- **Familiar technology:** Star topology is a familiar technology as its tools are cost-effective.
- **Easily expandable:** It is easily expandable as new stations can be added to the open ports on the hub.

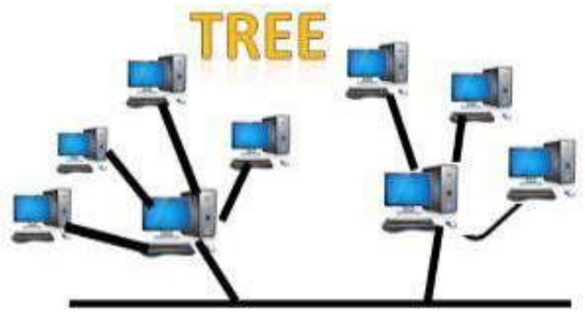
- **Cost effective:** Star topology networks are cost-effective as it uses inexpensive coaxial cable.
- **High data speeds:** It supports a bandwidth of approx 100Mbps. Ethernet 100BaseT is one of the most popular Star topology networks.

Disadvantages of Star topology :

- **A Central point of failure:** If the central hub or switch goes down, then all the connected nodes will not be able to communicate with each other.
- **Cable:** Sometimes cable routing becomes difficult when a significant amount of routing is required.

4. Tree topology

- Tree topology combines the characteristics of bus topology and star topology.
- A tree topology is a type of structure in which all the computers are connected with each other in hierarchical fashion.
- The top-most node in tree topology is known as a root node, and all other nodes are the descendants of the root node.
- There is only one path exists between two nodes for the data transmission. Thus, it forms a parent -child hierarch



Advantages and Disadvantages of Tree topology:

Advantages of Tree topology:

- **Support for broadband transmission:** Tree topology is mainly used to provide broadband transmission, i.e., signals are sent over long distances without being attenuated.
- **Easily expandable:** We can add the new device to the existing network. Therefore, we can say that tree topology is easily expandable.
- **Easily manageable:** In tree topology, the whole network is divided into segments known as star networks which can be easily managed and maintained.
- **Error detection:** Error detection and error correction are very easy in a tree topology.
- **Limited failure:** The breakdown in one station does not affect the entire network.
- **Point-to-point wiring:** It has point-to-point wiring for individual segments

.

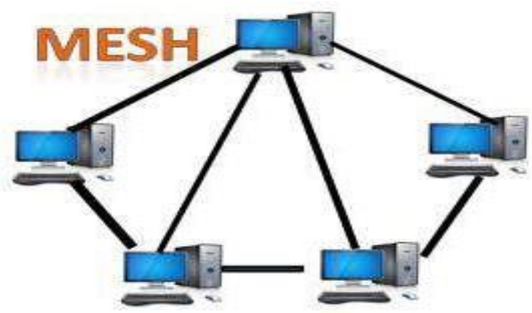
Disadvantages of Tree topology

- **Difficult troubleshooting:** If any fault occurs in the node, then it becomes difficult to troubleshoot the problem.
- **High cost:** Devices required for broadband transmission are very costly.
- **Failure:** A tree topology mainly relies on main bus cable and failure in main bus cable will damage the overall network.
- **Reconfiguration difficult:** If new devices are added, then it becomes difficult to rec

5. Mesh topology

- Mesh technology is an arrangement of the network in which computers are interconnected with each other through various redundant connections.
- There are multiple paths from one computer to another computer.
- It does not contain the switch, hub or any central computer which acts as a central point of communication.
- The Internet is an example of the mesh topology.
- Mesh topology is mainly used for WAN implementations where communication failures are a critical concern.
- Mesh topology is mainly used for wireless networks.
- Mesh topology can be formed by using the formula:

Number of cables = (n*(n-1))/2;
Where n is the number of nodes that represents the network.



Advantages and Disadvantages of Mesh topology :

Advantages of Mesh topology:

- **Reliable:** The mesh topology networks are very reliable as if any link breakdown will not affect the communication between connected computers.
- **Fast Communication:** Communication is very fast between the nodes.
- **Easier Reconfiguration:** Adding new devices would not disrupt the communication between other

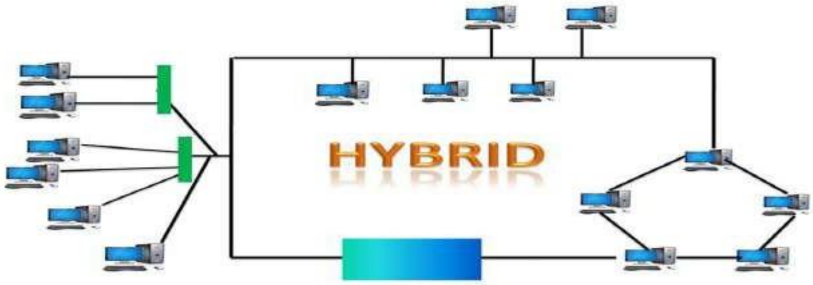
  Devices Disadvantages of Mesh topology

- **Cost:** A mesh topology contains a large number of connected devices such as a router and more transmission media than other topologies.

- **Management:** Mesh topology networks are very large and very difficult to maintain and manage. If the network is not monitored carefully, then the communication link failure goes undetected.
- **Efficiency:** In this topology, redundant connections are high that reduces the efficiency of the network.

6. Hybrid Topology

- The combination of various different topologies is known as **Hybrid topology**.
- A Hybrid topology is a connection between different links and nodes to transfer the data.
- When two or more different topologies are combined together is termed as Hybrid topology and if similar topologies are connected with each other will not result in Hybrid topology. For example, if there exist a ring topology in one branch of ICICI bank and bus topology in another branch of ICICI bank, connecting these two topologies will result in Hybrid topology.



THE INTERNET

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

A Brief History :

A network is a group of connected communicating devices such as computers and printers. An internet (note the lowercase letter i) is two or more networks that can communicate with each other. The most notable internet is called the Internet (uppercase letter I), a collaboration of more than hundreds of thousands of interconnected networks. The Advanced Research Projects Agency (ARPA) in the Department of Defense (DoD) was interested in finding a way to connect computers so that the researchers they funded could share their findings, thereby reducing costs and eliminating duplication of effort. In 1967, at an Association for Computing Machinery (ACM) meeting, ARPA presented its ideas for ARPANET, a small network of connected computers.

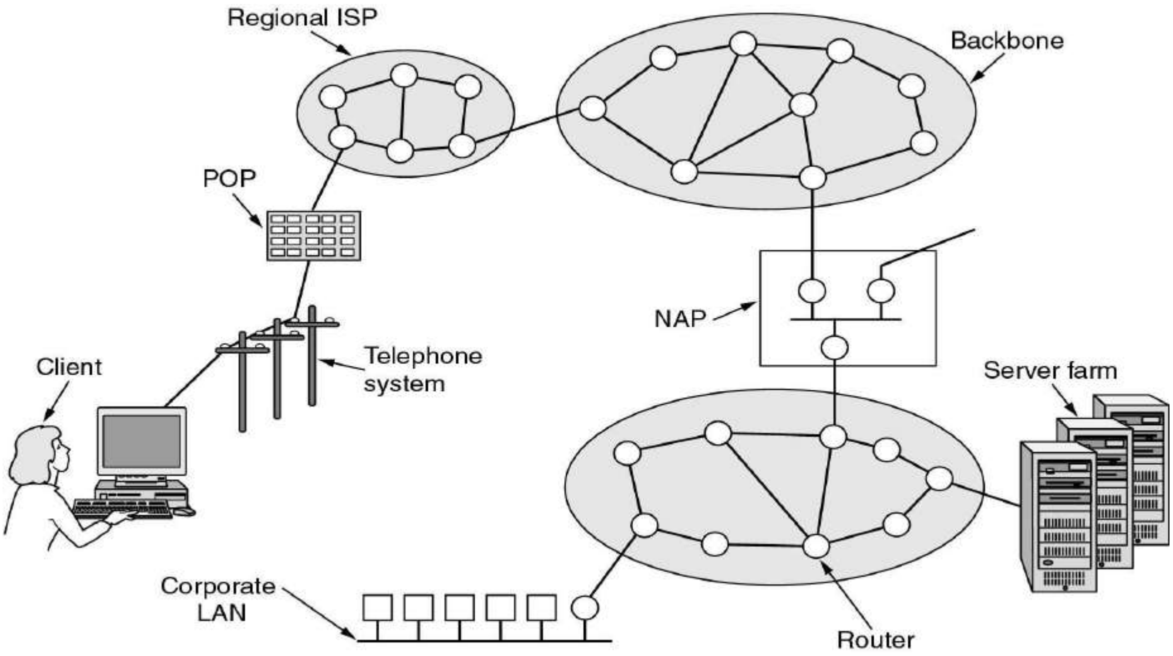
Transmission Control Protocol (TCP) and Internetworking Protocol (lP). IP would handle datagram routing while TCP would be responsible for higher-level functions such as segmentation, reassembly, and error detection. The internetworking protocol became known as TCPIIP.

The Internet Today

The Internet has come a long way since the 1960s. The Internet today is not a simple hierarchical structure. It is made up of many wide- and local-area networks joined by connecting devices and switching stations. It is difficult to give an accurate representation of the Internet because it is continually changing-new networks are being added, existing networks are adding addresses, and networks of defunct companies are being removed.

Internet service provider (ISP), company that provides Internet connections and services to individuals and organizations. In addition to providing access to the Internet, **ISPs** may also provide software packages (such as browsers), e-mail accounts, and a personal Web site or home page.

POP protocol is used in the **application layer protocol**, and it delivers best ability to fetch and receive all email by users.



***International Internet Service Providers**:*

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

***National Internet Service Providers**:*

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

Regional Internet Service Providers:

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

Local Internet Service Providers:

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

The OSI Reference Model:

The OSI model (minus the physical medium) is shown in Fig. This model is based on a proposal developed by the International Standards Organization (ISO) as a first step toward international standardization of the protocols used in the various layers (Day and Zimmermann, 1983). It was revised in 1995(Day, 1995). The model is called the ISO-OSI (Open Systems Interconnection) Reference Model because it deals with connecting open systems—that is, systems that are open for communication with other systems.

The OSI model has **seven layers**. The principles that were applied to arrive at the seven layers can be briefly summarized as follows:

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

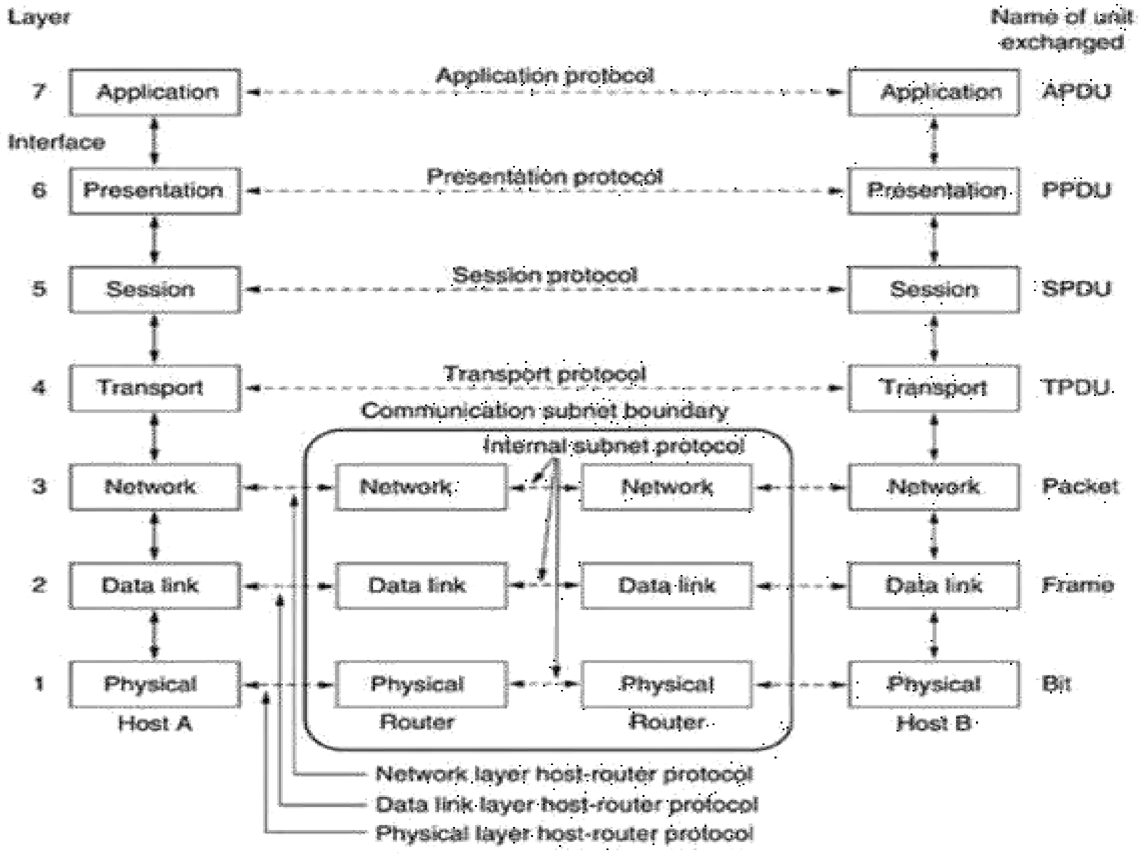


Fig.4: The OSI reference model

The Physical Layer:

The physical layer is concerned with transmitting raw bits over a communication channel. The design issues have to do with making sure that when one side sends a 1 bit, it is received by the other side as a 1 bit, not as a 0 bit.

The Data Link Layer:

The main task of the data link layer is to transform a raw transmission facility into a line that appears free of undetected transmission errors to the network layer. It accomplishes this task by having the sender break up the input data into data frames (typically a few hundred or a few thousand bytes) and transmits the frames sequentially. If the service is reliable, the receiver confirms correct receipt of each frame by sending back an acknowledgement frame.

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

The Network Layer:

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

If too many packets are present in the subnet at the same time, they will get in one another's way, forming bottlenecks. The control of such congestion also belongs to the network layer. More generally, the quality of service provided (delay, transit time, jitter, etc.) is also a network layer issue.

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

The Transport Layer:

The basic function of the transport layer is to accept data from above, split it up into smaller units if need be, pass these to the network layer, and ensure that the pieces all arrive correctly at the other end. Furthermore, all this must be done efficiently and in a way that isolates the upper layers from the inevitable changes in the hardware technology. The transport layer also determines what type of service to provide to the session layer, and, ultimately, to the users of the network. The most popular type of transport connection is an error-free point-to-point channel that delivers messages or bytes in the order in which they were sent. However, other possible kinds of transport service are the transporting of isolated messages, with no guarantee about the order of delivery, and the broadcasting of messages to multiple destinations. The type of service is determined when the connection is established.

The transport layer is a true end-to-end layer, all the way from the source to the destination. In other words, a program on the source machine carries on a conversation with a similar program on the destination machine, using the message headers and control messages.

The Session Layer:

The session layer allows users on different machines to establish sessions between them. Sessions offer various services, including dialog control (keeping track of whose turn it is to transmit), token management (preventing two parties from attempting the same critical operation at the same time), and synchronization (check pointing long transmissions to allow them to continue from where they were after a crash).

The Presentation Layer:

The presentation layer is concerned with the syntax and semantics of the information transmitted. In order to make it possible for computers with different data representations to communicate, the data structures to be exchanged can be defined in an abstract way, along with a standard encoding to be used "on the wire." The presentation layer manages these abstract data structures and allows higher-level data structures (e.g., banking records), to be defined and exchanged.

The Application Layer:

The application layer contains a variety of protocols that are commonly needed by users. One widely-used application protocol is HTTP (Hypertext Transfer Protocol), which is the basis for the World Wide Web. When a browser wants a Web page, it sends the name of the page it wants to the server using HTTP. The server then sends the page back. Other application protocols are used for file transfer, electronic mail, and network news.

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

Application Layer

Transport Layer

Internet Layer Host-to-

Network Layer

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. Fig. shows this correspondence.

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.

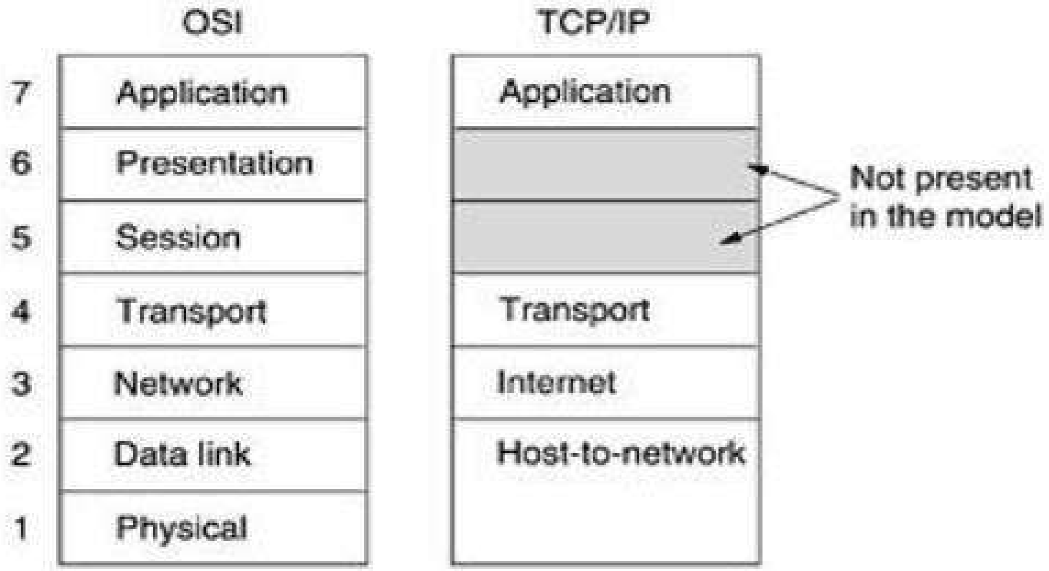


Fig.1: The TCP/IP reference model.

The second protocol in this layer, UDP (User Datagram Protocol), is an unreliable, connectionless protocol for applications that do not want TCP's sequencing or flow control and wish to provide their own. It is also widely used for one-shot, client-server-type request-reply queries and applications in which prompt delivery is more important than accurate delivery, such as transmitting speech or video. The relation of IP, TCP, and UDP is shown in Fig.2. Since the model was developed, IP has been implemented on many other networks.

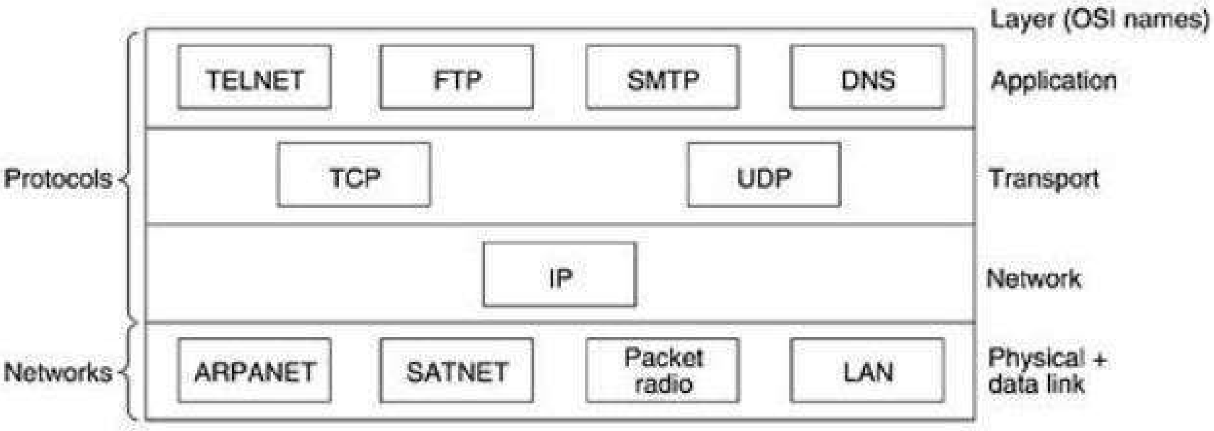


Fig.2: Protocols and networks in the TCP/IP model initially.

The 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.

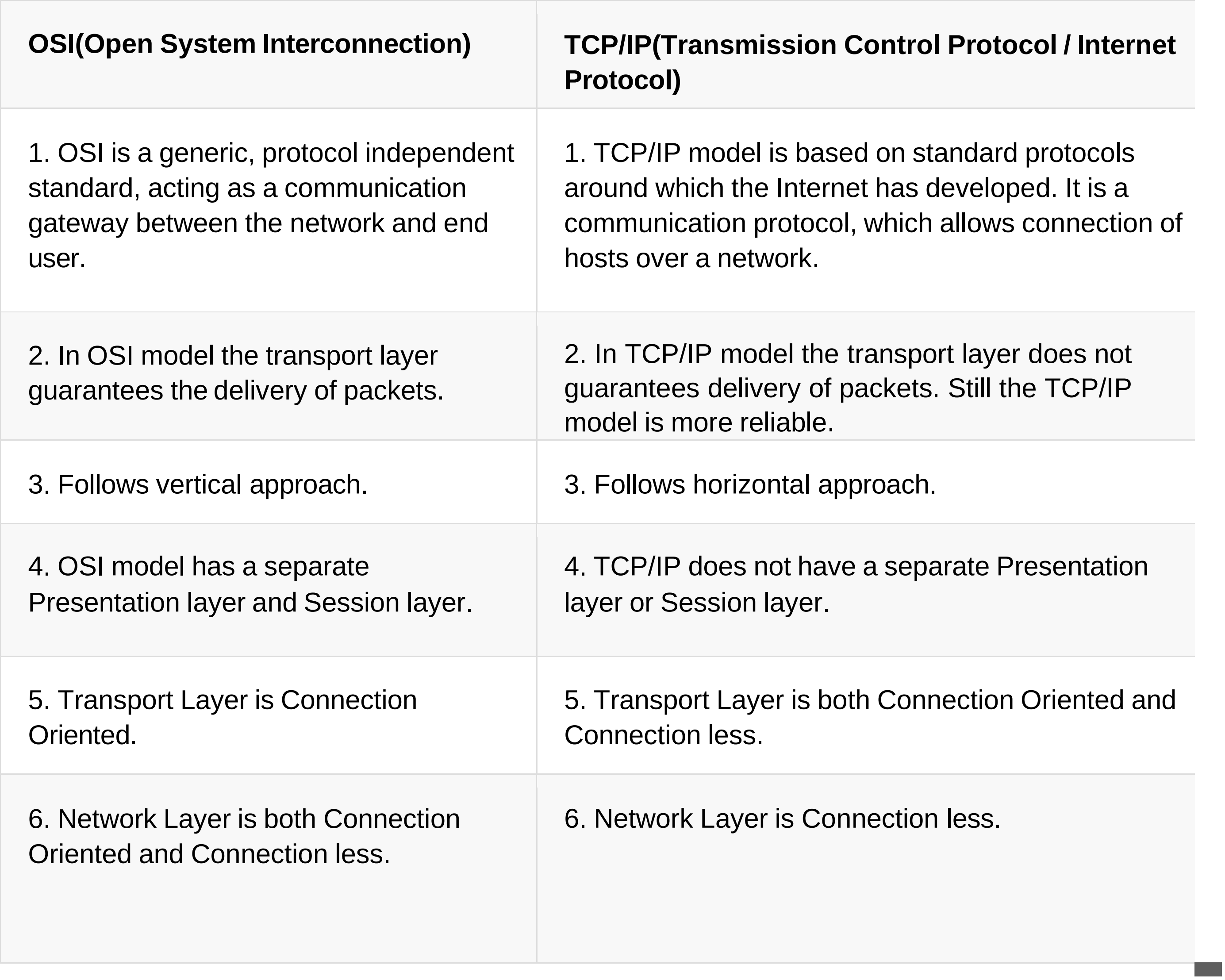
A layer's interface tells the processes above it how to access it. It specifies what the parameters are and what results to expect. It, too, says nothing about how the layer works inside.

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

.

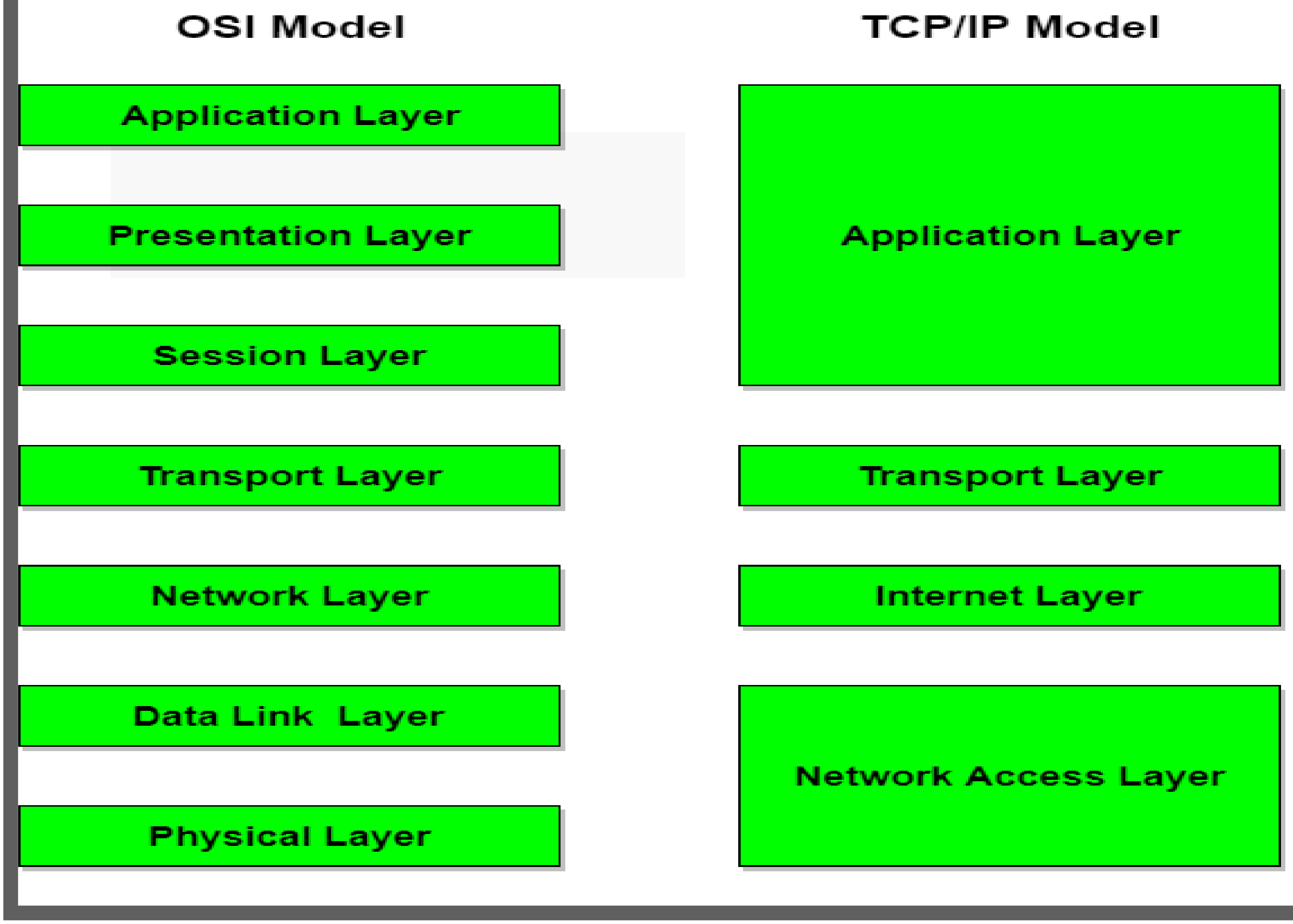
Comparison between OSI Reference Model and TCP/IP Reference Model



OSI(Open System Interconnection)	TCP/IP(Transmission Control Protocol / Internet Protocol)
1. OSI is a generic, protocol independent standard, acting as a communication gateway between the network and end user.	1. TCP/IP model is based on standard protocols around which the Internet has developed. It is a communication protocol, which allows connection of hosts over a network.
2. In OSI model the transport layer guarantees the delivery of packets.	2. In TCP/IP model the transport layer does not guarantees delivery of packets. Still the TCP/IP model is more reliable.
3. Follows vertical approach.	3. Follows horizontal approach.
4. OSI model has a separate Presentation layer and Session layer.	4. TCP/IP does not have a separate Presentation layer or Session layer.
5. Transport Layer is Connection Oriented.	5. Transport Layer is both Connection Oriented and Connection less.
6. Network Layer is both Connection Oriented and Connection less.	6. Network Layer is Connection less.

7. OSI is a reference model around which the networks are built. Generally it is used as a guidance tool.	7. TCP/IP model is, in a way implementation of the OSI model.
8. Network layer of OSI model provides both connection oriented and connectionless service.	8. The Network layer in TCP/IP model provides connectionless service.
9. OSI model has a problem of fitting the protocols into the model.	9. TCP/IP model does not fit any protocol
10. Protocols are hidden in OSI model and are easily replaced as the technology changes.	10. In TCP/IP replacing protocol is not easy.

Diagrammatic Comparison between OSI Reference Model and TCP/IP Reference Model



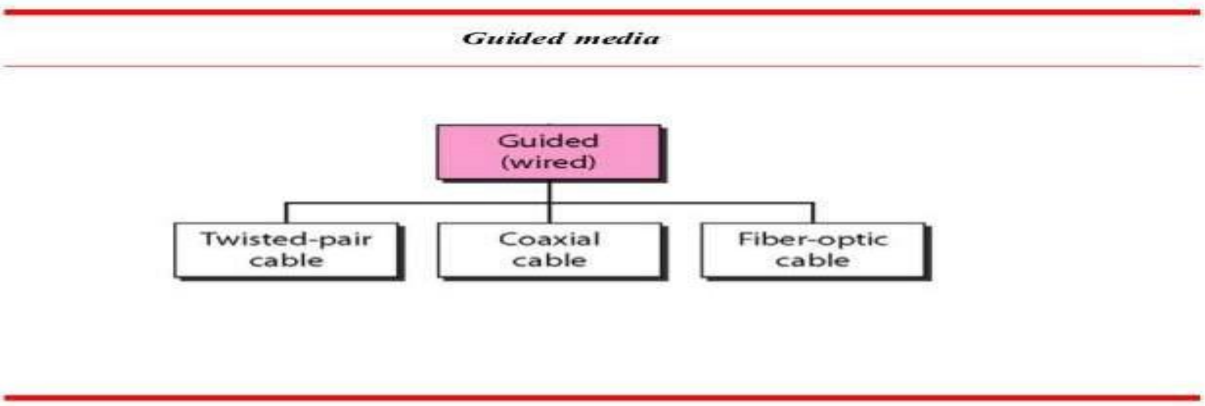
Types of Transmission Medias

1. **Guided Transmission Media**
2. **Unguided Transmission Media**

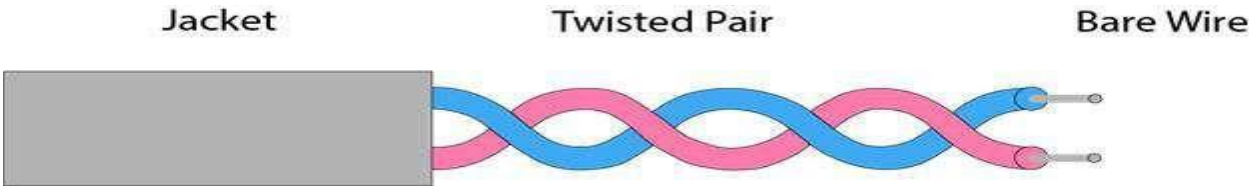
Guided Transmission Media :

It is defined as the physical medium through which the signals are transmitted. It is also known as Bounded media

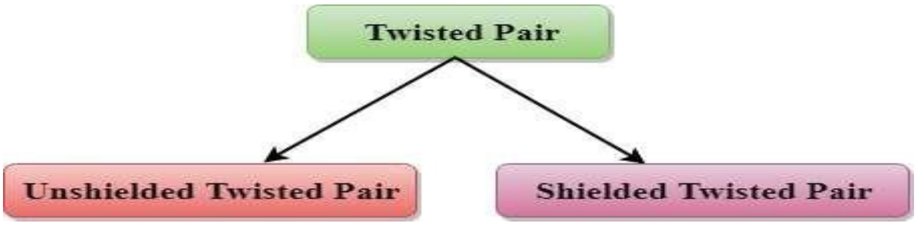
There are several types of cable which are commonly used with LANs. In some cases, a network will utilize only one type of cable, other networks will use a variety of cable types. The type of cable chosen for a network is related to the network’s topology, protocol, and size. Understanding the characteristics of different types of cable and how they relate to other aspects of a network is necessary for the development of a successful network.



1. **Twisted Pair Cable:** Twisted pair is a physical media made up of a pair of cables twisted with each other. A twisted pair cable is cheap as compared to other transmission media. Installation of the twisted pair cable is easy, and it is a lightweight cable. The frequency range for twisted pair cable is from 0 to 3.5KHz.



Types of Twisted pair:



- Unshielded Twisted Pair (UTP) Cable
- Shielded Twisted Pair (STP) Cable

Unshielded Twisted Pair (UTP) Cable

Twisted pair cabling comes in two varieties: shielded and unshielded. Unshielded twisted pair (UTP) is the most popular and is generally the best option for school networks.

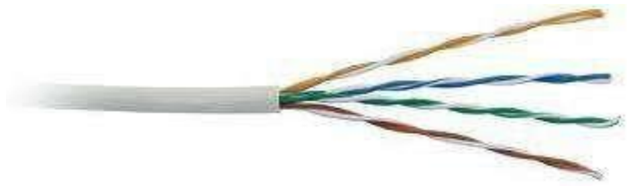


Figure 2.1 Unshielded Twisted Pair

The quality of UTP may vary from telephone-grade wire to extremely high-speed cable. The cable has four pairs of wires inside the jacket. Each pair is twisted with a different number of twists per inch to help eliminate interference from adjacent pairs and other electrical devices. The tighter the twisting, the higher the supported transmission rate and the greater the cost per foot. The EIA/TIA (Electronic Industry Association / Telecommunication Industry Association) has established standards of UTP and rated five categories of wire.

Type	**Use**
Category 1	Voice Only (Telephone Wire)
Category 2	Data to 4 Mbps (LocalTalk)
Category 3	Data to 10 Mbps (Ethernet)
Category 4	Data to 20 Mbps (16 Mbps Token Ring)
Category 5	Data to 100 Mbps (Fast Ethernet)

Table Categories of Unshielded Twisted Pair

Advantages Of Unshielded Twisted Pair:

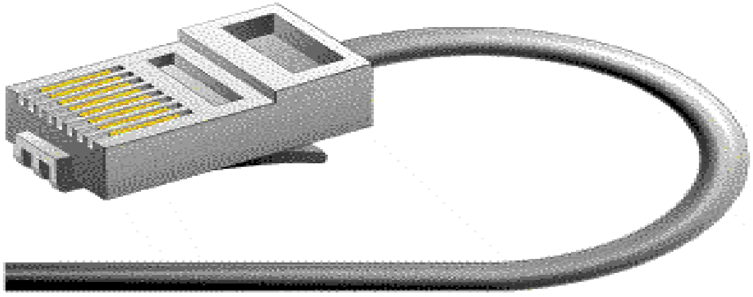
- It is cheap.
- Installation of the unshielded twisted pair is easy.
- It can be used for high-speed LAN.

Disadvantage:

- This cable can only be used for shorter distances because of attenuation.

Unshielded Twisted Pair Connector

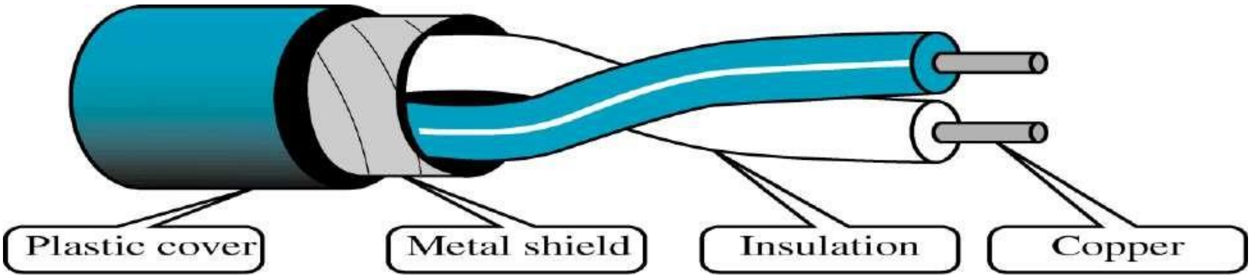
The standard connector for unshielded twisted pair cabling is an RJ-45 connector. This is a plastic connector that looks like a large telephone-style connector (See fig. 2.2). a slot allows the RJ-45 to be inserted only one way. RJ stands for Registered Jack, implying that the connector follows a standard borrowed from the telephone industry. This standard designates which wire goes with each pin inside the connector.



Shielded Twisted Pair (STP) Cable :

A shielded twisted pair is a cable that contains the mesh surrounding the wire that allows the higher transmission rate.

A disadvantage of UTP is that it may be susceptible to radio and electrical frequency interference. Shielded twisted pair |(STP) is suitable for environments with electrical interference; however, the extra shielding can make the cables quite bulky. Shielded twisted pair is often used on networks using Token Ring topology.



Characteristics Of Shielded Twisted Pair:

- **The cost of the shielded twisted pair cable is not very high and not very low.**
- **An installation of STP is easy.**
- **It has higher capacity as compared to unshielded twisted pair cable.**
- **It has a higher attenuation.**
- **It is shielded that provides the higher data transmission rate.**

Disadvantages

- **It is more expensive as compared to UTP and coaxial cable.**
- **It has a higher attenuation rate.**

2. Coaxial Cable

Coaxial cable is very commonly used transmission media, for example, TV wire is usually a coaxial cable.

Coaxial cabling has a single copper conductor at its center. A plastic layer provides insulation between the center conductor and the braided metal shield (See fig. 3). The metal shield helps to block any outside interference from fluorescent lights, motors, and other computers.

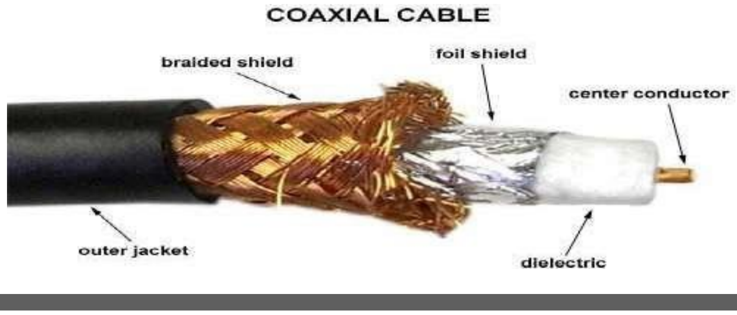
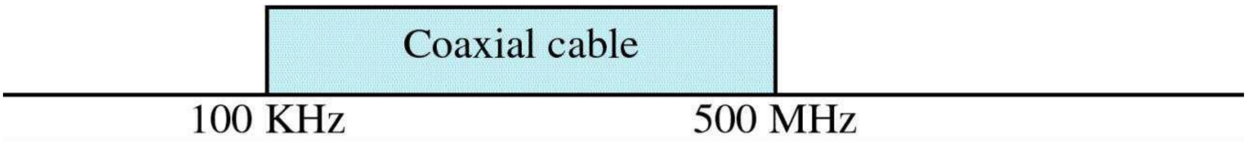


Fig. 2.3 Coaxial cable



- The name of the cable is coaxial as it contains two conductors parallel to each other.
- It has a higher frequency as compared to Twisted pair cable.
- The inner conductor of the coaxial cable is made up of copper, and the outer conductor is made up of copper mesh. The middle core is made up of non-conductive cover that separates the inner conductor from the outer conductor.

- Thin coaxial cable is also referred to as thinnet. 10base2 refers to the specifications for thin coaxial cable carrying Ethernet signals. The 2 refers to the approximate maximum segment length being 200 meters. In actual fact the maximum segment length is 185 meters. Thin coaxial cable is popular in school networks, especially linear bus networks.

- Thick coaxial cable is also referred to as thicknet. 10base refers to the specifications for thick coaxial cable carrying Ethernet signals. The 5 refers to the maximum segment length being 500 meters. Thick coaxial cable has an extra protective plastic cover that helps keep moisture away from the center conductor. This makes thick coaxial a great choice when running longer lengths in a linear bus network. One disadvantage of thick coaxial is that it does not bend easily and is difficult to install.

Types of Coaxial cable

Baseband transmission: It is defined as the process of transmitting a single signal at high speed.
Broadband transmission: It is defined as the process of transmitting multiple signals simultaneously.

Advantages Of Coaxial cable:

- The data can be transmitted at high speed.
- It has better shielding as compared to twisted pair cable.
- It provides higher bandwidth.

Disadvantages Of Coaxial cable:

- It is more expensive as compared to twisted pair cable.

If any fault occurs in the cable causes the failure in the entire network.

Fiber Optic Cable :

- Fiber optic cable is a cable that uses electrical signals for communication.

- Fiber optic is a cable that holds the optical fibers coated in plastic that are used to send the data by pulses of light.

- The plastic coating protects the optical fibers from heat, cold, electromagnetic interference from other types of wiring.

Fiber optics provide faster data transmission than copper wires

Fiber optic cabling consists of a center glass core surrounded by several layers of protective materials. It transmits light rather than electronic signals eliminating the problem of electrical interference.

This makes it ideal for certain environments that contain a large amount of electrical interference. It has also made it the standard for connecting networks between buildings, due to its immunity to the effects of moisture and lighting.

Fiber optic cable has the ability to transmit signals over much longer distances than coaxial and twisted pair. It also has made it the standard for connecting networks between buildings, due to its immunity to the effects of moisture and lighting.

Fiber optic cable has the ability to transmit signals over mush longer distances than coaxial and twisted pair. It also has the capability to carry information at vastly greater speeds. This capacity broadens communication possibilities to include services such as video conferencing and interactive services. The cost of fiber optic cabling is comparable to copper cabling; however it is more difficult to install and modify. 10BaseF refers to the specifications for fiber optic cable carrying Ethernet signals.

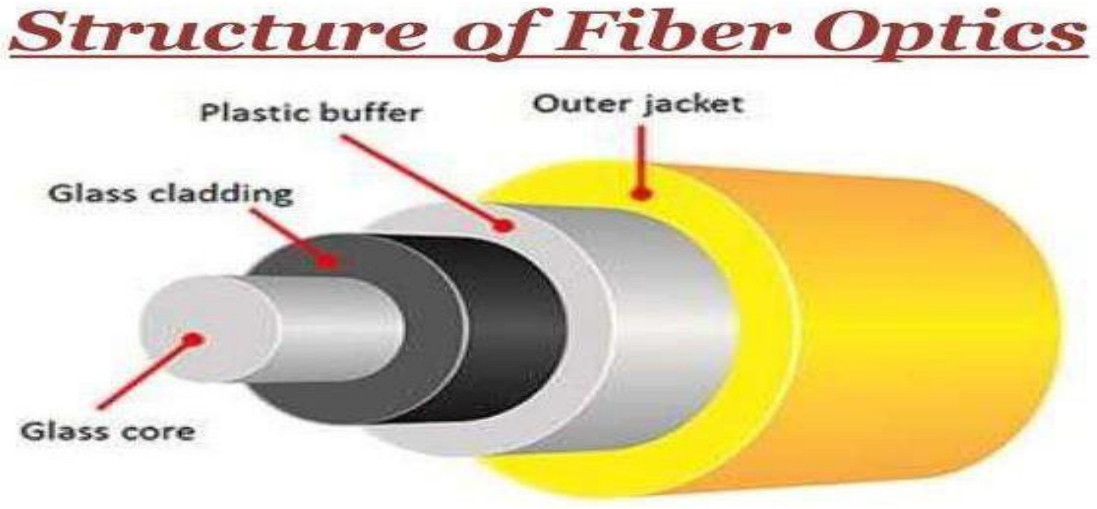


Fig-1. Fiber Optic Cable

Facts about fiber optic cables:

- Outer insulating jacket is made of Teflon or PVC.
- Kevlar fiber helps to strengthen the cable and prevent breakage.
- A plastic coating is used to cushion the fiber center.
- Center (core) is made of glass or plastic fibers.

Fiber Optic Connector

The most common connector used with fiber optic cable is an ST connector. It is barrel shaped, similar to a BNC connector. A newer connector, the SC, is becoming more popular. It has a squared face and is easier to connect in a confined space.

Specification	Cable Type	Maximum length
10BaseT	**Unshielded Twisted Pair**	**100 meters**
10Base2	**Thin Coaxial**	**185 meters**
10Base5	**Thick Coaxial**	**500 meters**
10BaseF	**Fiber Optic**	**2000 meters**
100BaseT	**Unshielded Twisted Pair**	**100 meters**
100BaseTX	**Unshielded Twisted Pair**	**220 meters**

Table : Ethernet Cable Summary

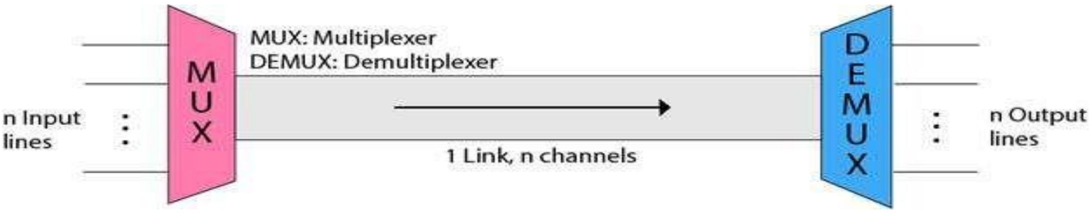
Unguided Transmission Media:

Unguided transmission media is data signals that flow through the air. They are not guided or bound to a channel to follow.

Unguided media transport electromagnetic waves without using a physical conductor. This type of communication is often referred to as wireless communication. Signals are normally broadcast through free space and thus are available to anyone who has a device receiving them. Unguided signals can travel from the source to destination in several ways: ground propagation, sky propagation, and line-of-sight propagation.

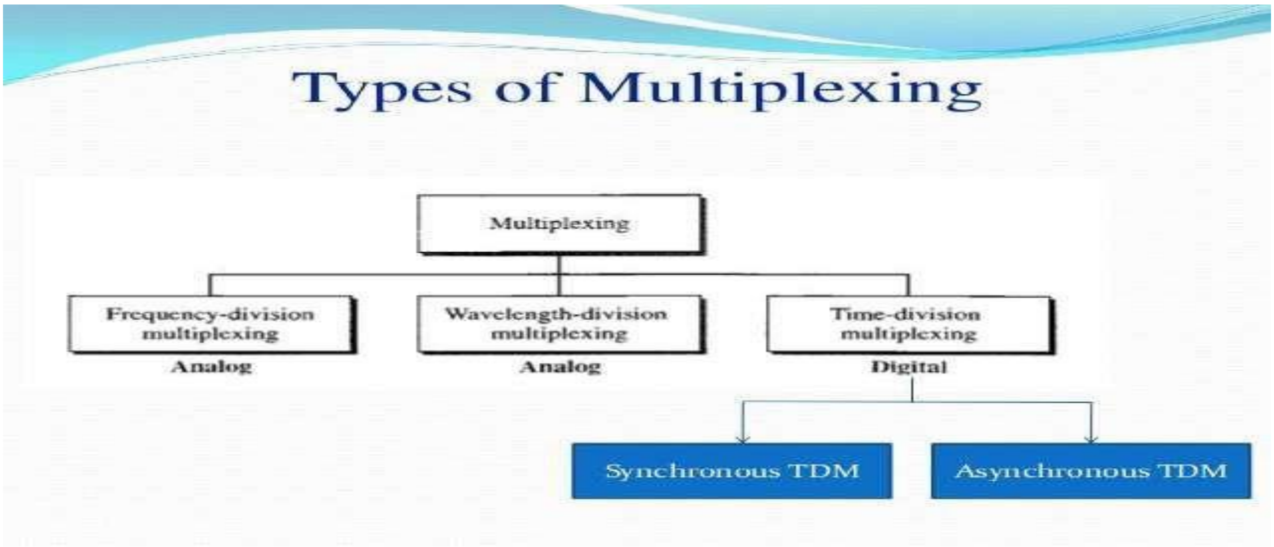
Multiplexing

- It is the set of techniques that allows the simultaneous transmission of multiple signals across a single data link.
- Multiplexing is done using a device called Multiplexer (MUX) that combine *n* input lines to generate one output line i.e. (*many to one)*.
- At the receiving end a device called De-multiplexer (DEMUX) is used that separate signal into its component signals i.e. one input and several outputs *(one to many)*.



Advantages of Multiplexing:

- Effective use of the bandwidth of medium
- More than one signals can be sent over single medium or link



Types of Multiplexing

Multiplexing

Frequency-division multiplexing

Analog

Wavelength-division multiplexing

Analog

Time-division multiplexing

Digital

Synchronous TDM

Asynchronous TDM

1. Frequency Division Multiplexing:

- It is an analog technique.
- Signals of different frequencies are combined into a composite signal and is transmitted on the single link.
- Bandwidth of a link should be greater than the combined bandwidths of the various channels.
- Each signal is having different frequency.
- Channels are separated by the strips of unused bandwidth called *Guard Bands* (to prevent overlapping).

When the carrier is frequency, FDM is used. FDM is an analog technology. FDM divides the spectrum or carrier bandwidth in logical channels and allocates one user to each channel. Each user can use the channel frequency independently and has exclusive access of it. All channels are divided in such a way that they do not overlap with each other. Channels are separated by guard bands. Guard band is a frequency which is not used by either channel.

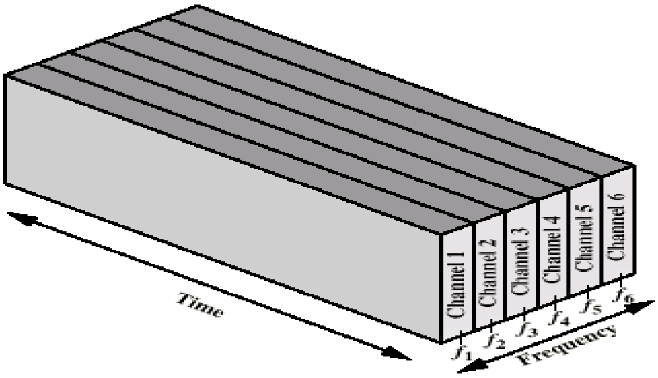
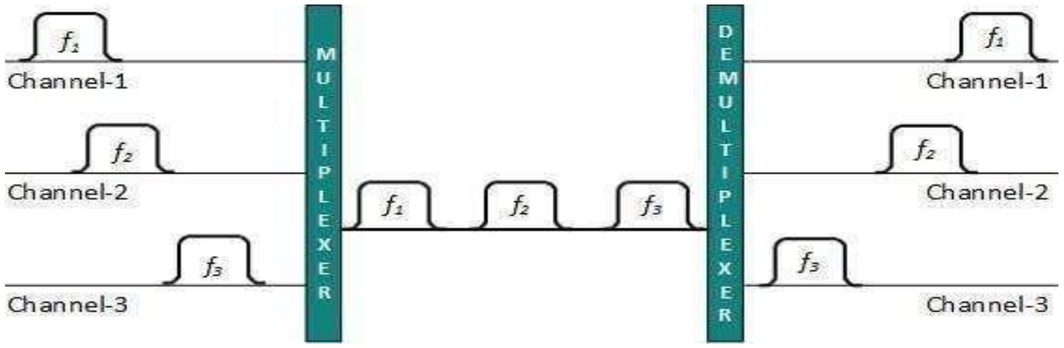
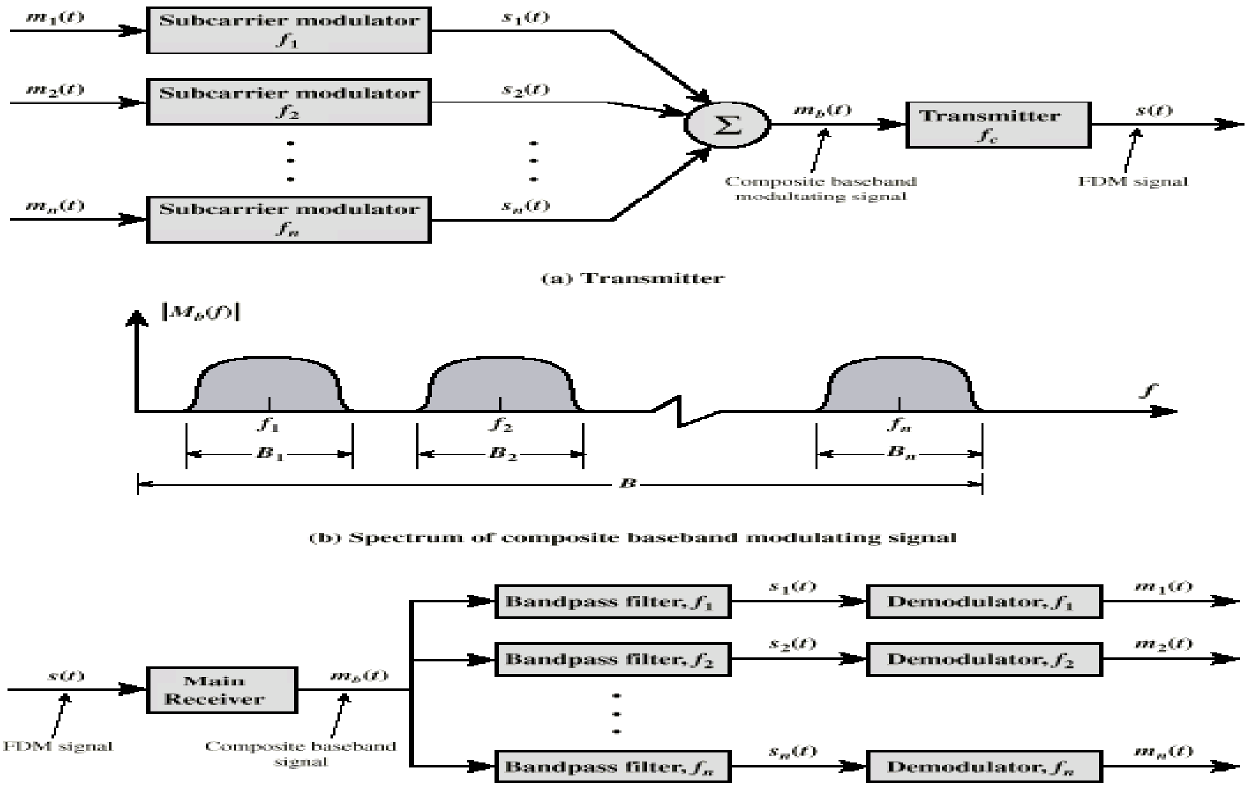


Fig: FDM System





(a) Transmitter

(b) Spectrum of composite baseband modulating signal

(c) Receiver

2. Wavelength division multiplexing:

- WDM is an analog multiplexing technique.
- Working is same as FDM.
- In WDM different signals are *optical or light* signals that are transmitted through optical fiber.
- Various light waves from different sources are combined to form a composite light signal that is transmitted across the channel to the receiver.
- At the receiver side, this composite light signal is broken into different light waves by Demultiplexer.
- This Combining and the Splitting of light waves is done by using a PRISM. Prism bends beam of light based on the angle of incidence and the frequency of light wave.

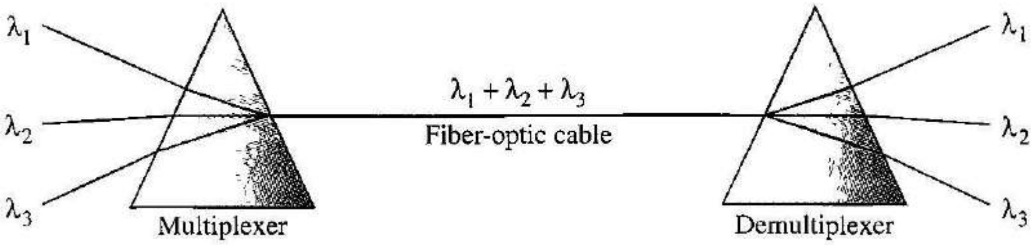
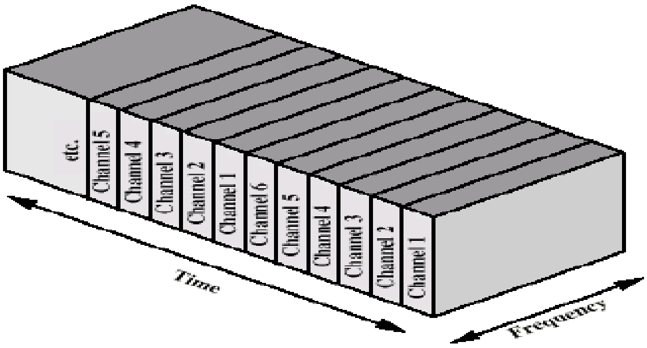


Fig : Wave Division Multiplexing

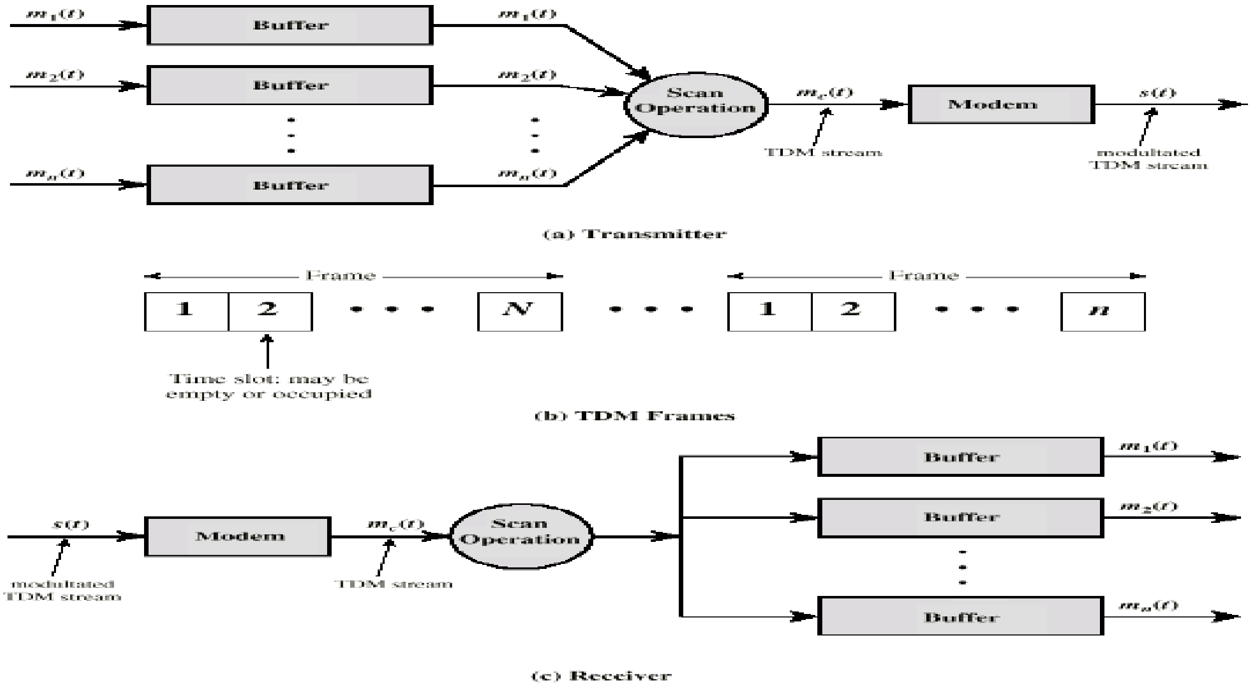
- Number of sources generating laser beams at different frequencies.
- Multiplexer consolidates sources for transmission over single fiber.
- Optical amplifiers amplify all wavelengths.
  -Typically tens of km apart
- Demux separates channels at the destination
- Mostly 1550nm wavelength range
- Same general architecture as other FDM
- Was 200MHz per channel
- Now 50GHz

Time Division Multiplexing:

- It is the digital multiplexing technique.
- Channel/Link is not divided on the basis of *frequency* but on the *basis of time.*
- Total time available in the channel is divided between several users.
- Each user is allotted a particular time interval called *time slot* or *slice.*
- In TDM the data rate capacity of the transmission medium should be greater than the data rate required by sending of receiving devices



TDM is applied primarily on digital signals but can be applied on analog signals as well. In TDM the shared channel is divided among its user by means of time slot. Each user can transmit data within the provided time slot only. Digital signals are divided in frames, equivalent to time slot i.e. frame of an optimal size which can be transmitted in given time slot.TDM works in synchronized mode. Both ends, i.e. Multiplexer and De-multiplexer are timely synchronized and both switch to next channel simultaneously.



(a) Transmitter

(b) TDM Frames

(c) Receiver

<u>Types of TDM</u> :

- **Synchronous TDM**
- **Asynchronous TDM**

<u>Synchronous TDM</u> :

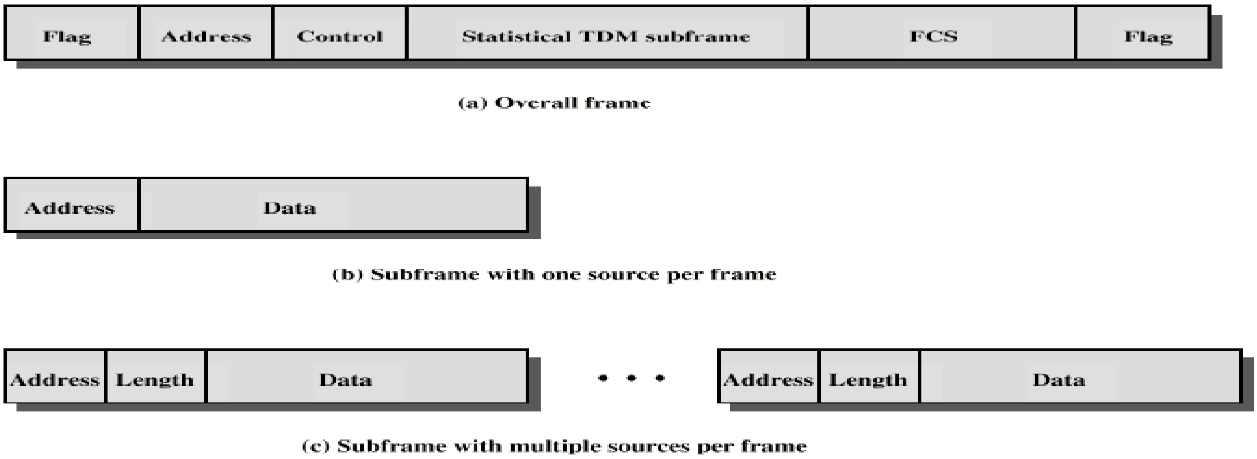
- Each device is given same Time Slot to transmit the data over the link, whether the device has any data to transmit or not.
- Each device places its data onto the link when its *Time Slot* arrives, each device is given the possession of line turn by turn.
- If any device does not have data to send then its time slot remains empty.
- Time slots are organized into *Frames* and each frame consists of one or more time slots.
- If there are *n* sending devices there will be *n* slots in frame.

<u>Asynchronous TDM (or) Statistical TDM</u>:

The channel capacity cannot be fully utilized. Some of the slots go empty in certain frames

Statistical TDM:

- In Synchronous TDM many slots are wasted
- Statistical TDM allocates time slots dynamically based on demand
- Multiplexer scans input lines and collects data until frame full
- Data rate on line lower than aggregate rates of input lines

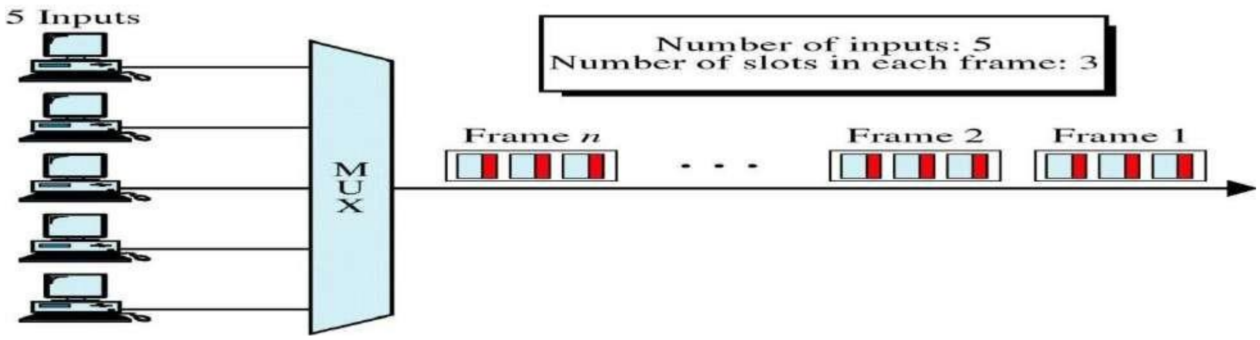


(a) Overall frame

(b) Subframe with one source per frame

(c) Subframe with multiple sources per frame

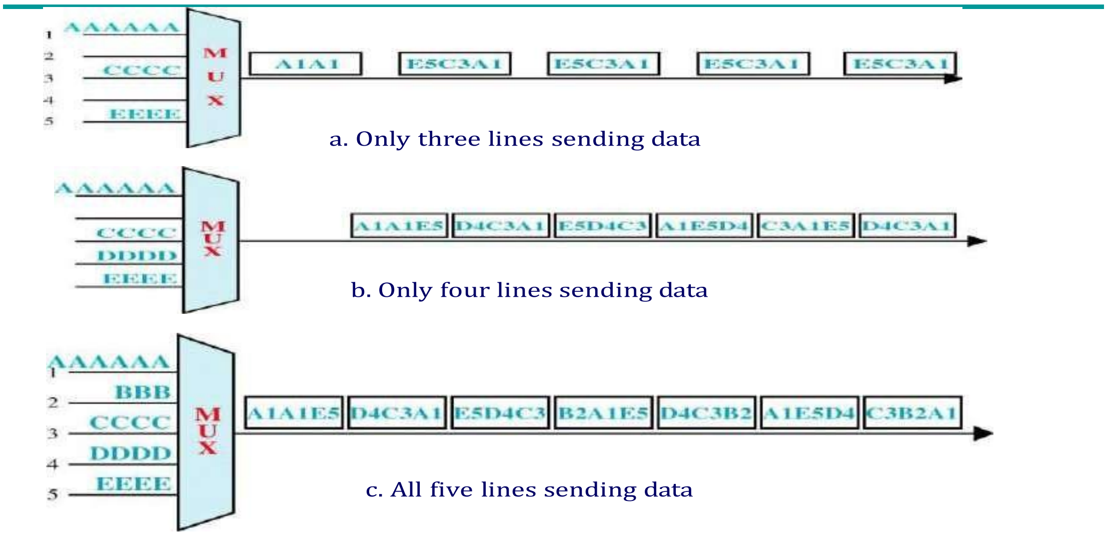
Fig: Statistical TDM Frame Formats



Asynchronous TDM

- Also known as Statistical Time Division *multiplexing*
- In Asynchronous TDM time slots are not *Fixed* i.e. slots are Flexible.
- Total speed of the input lines can be greater than the capacity of the path.
- In ASTDM we have *n* input lines and *m* slots i.e. *m* less than *n (m<n*).
- Slots are not predefined rather slots are allocated to any of the device that has data to send.

Frames and Addresses



a. Only three lines sending data

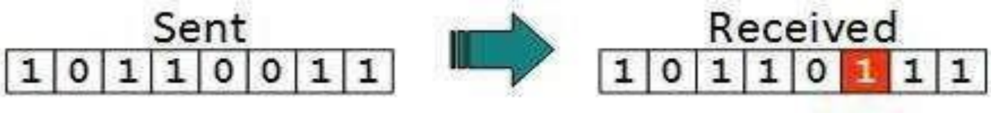
b. Only four lines sending data

c. All five lines sending data

Error Detection and Correction :

Types of Errors : There may be three types of errors

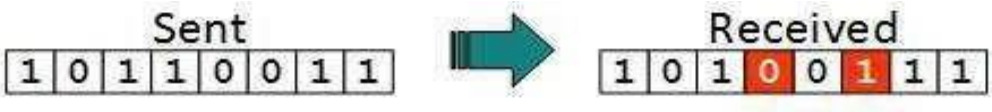
- **Single bit error**



In a frame, there is only one bit, anywhere though, which is corrupt

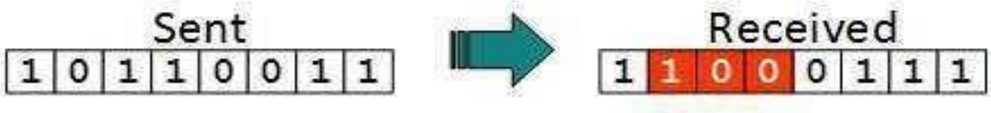
.

- **Multiple bits error**



Frame is received with more than one bits in corrupted state.

- **Burst error**



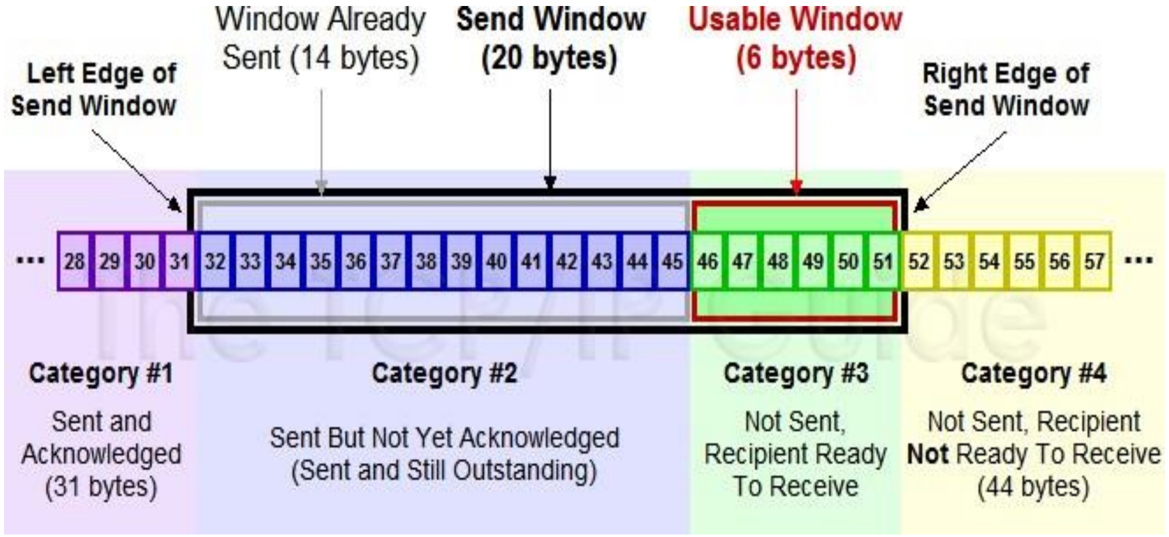
Frame contains more than1 consecutive bits corrupted.

Error control mechanism may involve two possible ways:

- **Error detection**
- **Error correction**

Sliding Window Protocols:

- **Sliding window protocols** are data link layer **protocols** for reliable and sequential delivery of data frames.
- The **sliding window** is also used in Transmission Control **Protocol**. In this **protocol**, multiple frames can be sent by a sender at a time before receiving an acknowledgment from the receiver


Window Already Sent (14 bytes)
Send Window (20 bytes)
Usable Window (6 bytes)
Left Edge of Send Window
Right Edge of Send Window
... 28 29 30 31 32 33 34 35 36 37 38 39 40 41 42 43 44 45 46 47 48 49 50 51 52 53 54 55 56 57 ...
Category #1
Sent and Acknowledged (31 bytes)
Category #2
Sent But Not Yet Acknowledged (Sent and Still Outstanding)
Category #3
Not Sent, Recipient Ready To Receive
Category #4
Not Sent, Recipient Not Ready To Receive (44 bytes)

What is DLL (Data Link Layer)?

The Data Link Layer is the second layer in the OSI model, above the Physical Layer, which ensures that the error free data is transferred between the adjacent nodes in the network. It breaks the datagram passed down by above layers and converts them into frames ready for transfer. This is called **Framing.**

It provides two main functionalities

- Reliable data transfer service between two peer network layers
- Flow Control mechanism which regulates the flow of frames such that data congestion is not there at slow receivers due to fast senders.

THE DATA LINK LAYER DESIGN ISSUES

FUNCTIONS

- Providing a well-defined service interface to the network layer.
- Dealing with transmission errors.
- Regulating the flow of data so that slow receivers are not swamped by fast senders –flow control.

The two main functions of the data link layer are:

1. **Data Link Control (DLC)**: It deals with the design and procedures for communication b/w nodes: node-to-node communication.

2. **Media Access Control (MAC)**: It explains how to share the link.

1. DATA LINK CONTROL (DLC):

Data link control functions includes

(1) Framing.

(2) Error Control.

(3) Flow Control.

(1) <u>FRAMING</u>

The frame contains

1. Frame header
2. Payload field for holding packet
3. Frame trailer

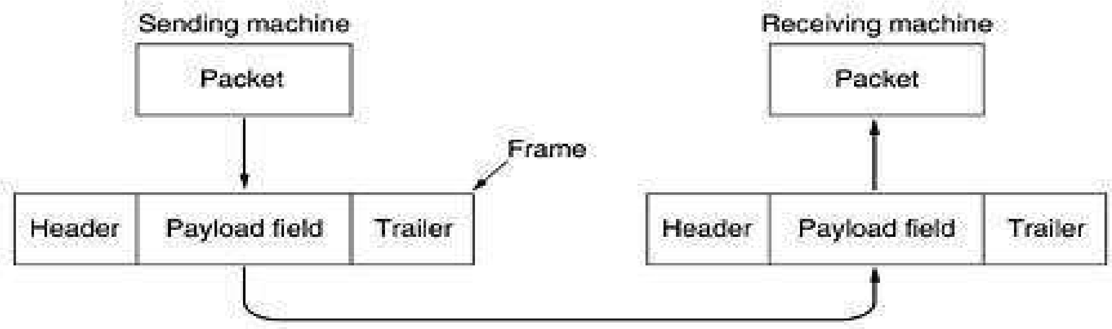


Figure 1.1 Relationships between Packets and Frames

Services provided to the network layer

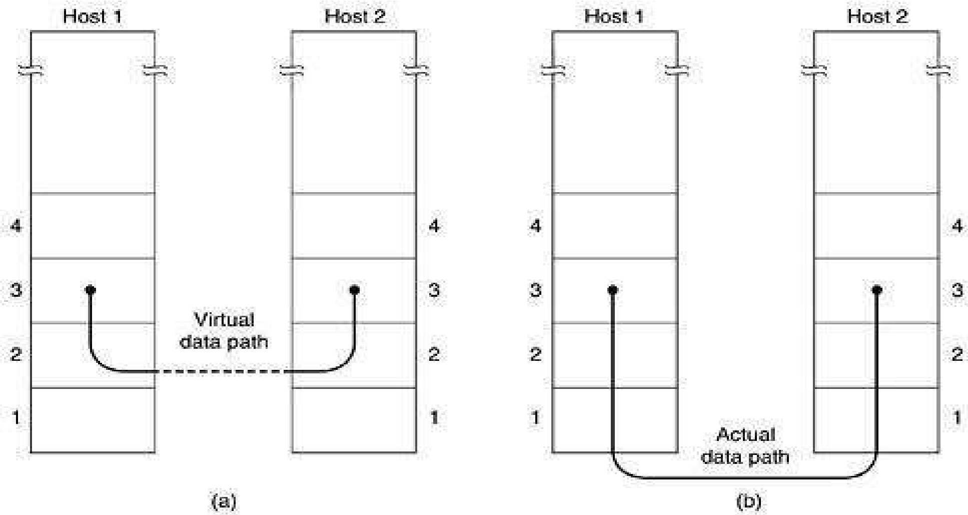


Figure 1.2 (a) Virtual communication. (b) Actual communication.

Transferring data from the network layer on the source machine to the network layer on the destination machine. The data link layer can be designed to offer various services. The actual services offered can vary from system to system. Three reasonable possibilities that are commonly provided are

1. **Unacknowledged connectionless service**

   - Source machine sends independent frames to destination machine having destination machine acknowledge them
   - No logical connection
   - Used when error rate is very low
   - Good for real-time traffic (voice)

2. **Acknowledged connectionless service**

   - No logical connection
   - Each frame sent is individually acknowledged
   - Useful over unreliable channels (i.e. wireless systems)

3. **Acknowledged connection-oriented service**

   - Source and destination machines establish a connection before any data are transferred
   - Each frame is numbered
   - DLL guarantees that...
     - Each frame is received
     - Each frame is received exactly once
     - Each frame is received in the right order

3 PHASES

When connection-oriented service is used, transfers go through three distinct phases

1. Connection established
2. Frames are transmitted
3. Connection released

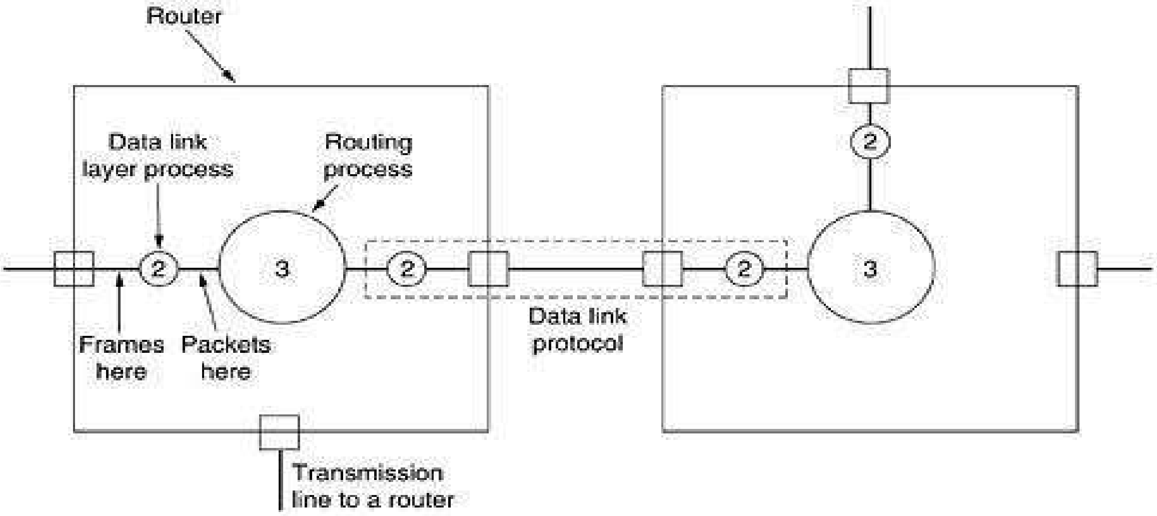


Figure 1.3 Placement of the data link Protocol

- Consider a typical example: a WAN subnet consisting of routers connected by point-to-point leased telephone lines.
- When a frame arrives at a router, the hardware checks it for errors, and then passes the frame to the data link layer software.
- The data link layer software checks to see if this is the frame expected, and if so, gives the packet contained in the payload field to the routing software.
- The routing software then chooses the appropriate outgoing line and passes the packet back down to the data link layer software, which then transmits it. The flow over two routers is shown in **Fig. 1-3.**

(1). FRAMING

Breaking the bit stream up into frames is more difficult than it at first appears. One way to achieve this framing is to insert time gaps between frames, much like the spaces between words in ordinary text. However, networks rarely make any guarantees about timing, so it is possible these gaps might be squeezed out or other gaps might be inserted during transmission.

There are four methods:

1. Character count.

2. Flag bytes with byte stuffing.

3. Starting and ending flags, with bit stuffing.

4. Physical layer coding violations.

1. Character count:

The first framing method uses a field in the header to specify the number of characters in the frame. When the data link layer at the destination sees the character count, it knows how many characters follow and hence where the end of the frame is. This technique is shown in Fig. 3-4(a) for four frames of sizes 5, 5, 8, and 8 characters, respectively.

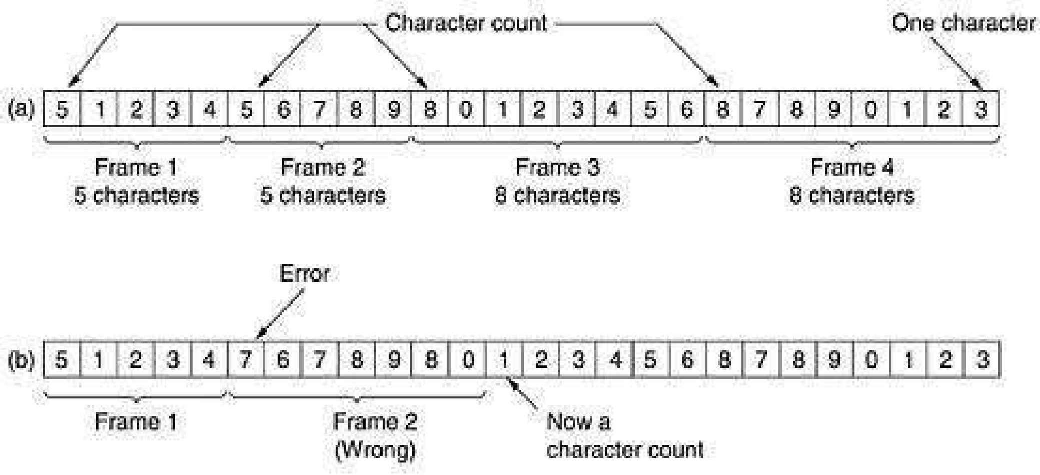


Figure 3-4. A character stream. (a) Without errors. (b) With one error.

Explanation (Figure 3-4.(a) A character stream Without errors.)

- The first framing method uses a field in the header to specify the number of characters in the frame.
- When the data link layer at the destination sees the character count, it knows how many characters follow and hence where the end of the frame is.
- This technique is shown in Fig. 3-4(a) for four frames of sizes 5, 5, 8, and 8 characters, respectively.
- The trouble with this algorithm is that the count can be garbled by a transmission error.

Explanation (Figure 3-4.(b) A character stream with errors.)

- For example, if the character count of 5 in the second frame of Fig. 3-4(b) becomes a 7, the destination will get out of synchronization and will be unable to locate the start of the next frame.
- Even if the checksum is incorrect so the destination knows that the frame is bad, it still has no way of telling where the next frame starts.
- Sending a frame back to the source asking for a retransmission does not help either, since the destination does not know how many characters to skip over to get to the start of the retransmission. For this reason, the character count method is rarely used anymore.

2. Flag bytes with byte stuffing:

Character-oriented framing approach

- In a character-oriented approach, data to be carried are 8-bit characters.
- The header, which normally carries the source and destination addresses and other control information.
- Trailer carries error detection or error correction redundant bits, are also multiples of 8 bits.
- To separate one frame from the next, an 8-bit (1-byte) flag is added at the beginning and the end of a frame.
- The flag, composed of protocol-dependent special characters, signals the start or end of a frame.

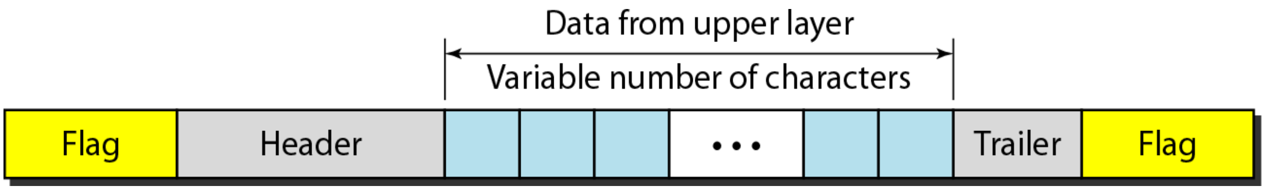


Figure: shows the format of a frame in a character-oriented protocol

Advantage:

1. Simple framing method.
2. Character-oriented framing was popular when only text was exchanged by the data Link layers.
3. The flag could be selected to be any character not used for text communication.

Disadvantage:

1. Even if with checksum, the receiver knows that the frame is bad there is no way to tell where the next frame starts.
2. Asking for retransmission doesn't help either because the start of the retransmitted frame is not known.
3. Hence No longer used.

3. Starting and ending character with byte stuffing

Byte stuffing is the process of adding 1 extra byte whenever there is a flag or escape character in the text.

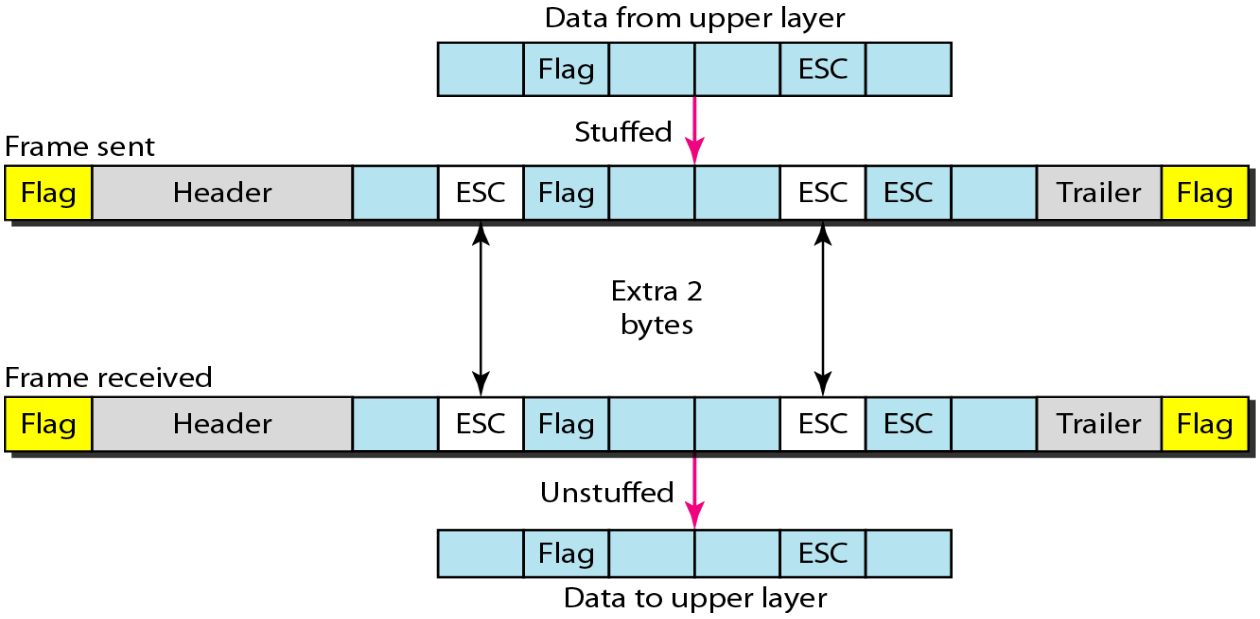


Figure : Byte stuffing and unstuffing

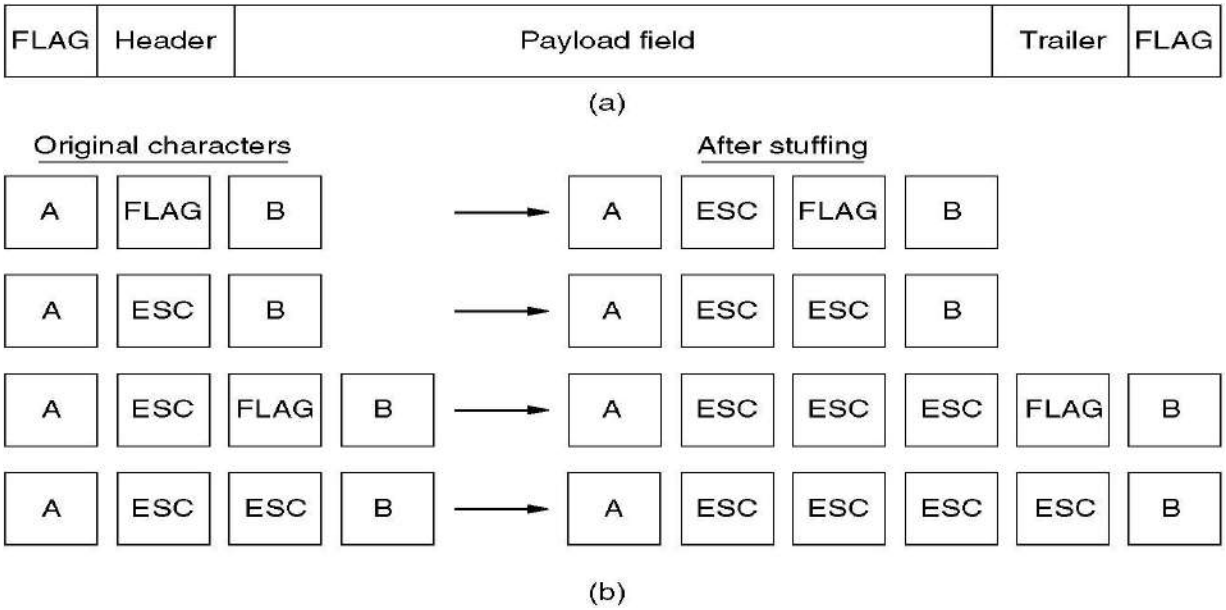


Fig: Framing with byte stuffing

Problem: fixed character size: assumes character size to be 8 bits: can’t handle heterogeneous environment.

Bit-Oriented framing approach

- Bit stuffing is the process of adding one extra 0 whenever five consecutive 1’s follow a 0 in the data, so that the receiver does not mistake the pattern 0111110 for a flag.
- Most protocols use a special 8-bit pattern flag 01111110 as the delimiter to define the beginning and the end of the frame, as shown in Figure below
- This flag can create the same type of problem. That is, if the flag pattern appears in the data, we need to somehow inform the receiver that this is not the end of the frame.

- We do this by stuffing 1 single bit (instead of I byte) to prevent the pattern from looking like a flag. The strategy is called bit stuffing.

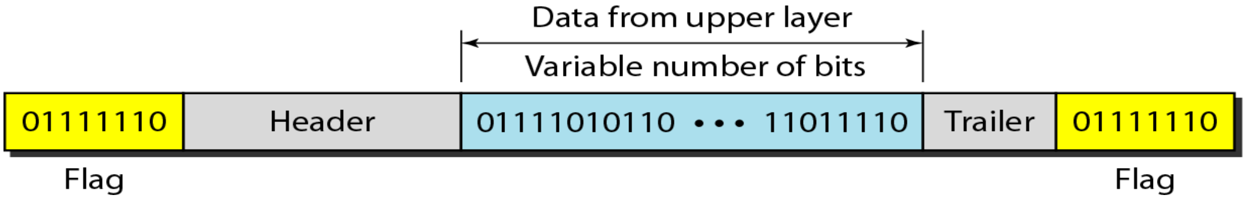


Figure (a)

Bit stuffing is the process of adding one extra 0 whenever five consecutive 1s follow a 0 in the data, so that the receiver does not mistake the pattern 0111110 for a flag.

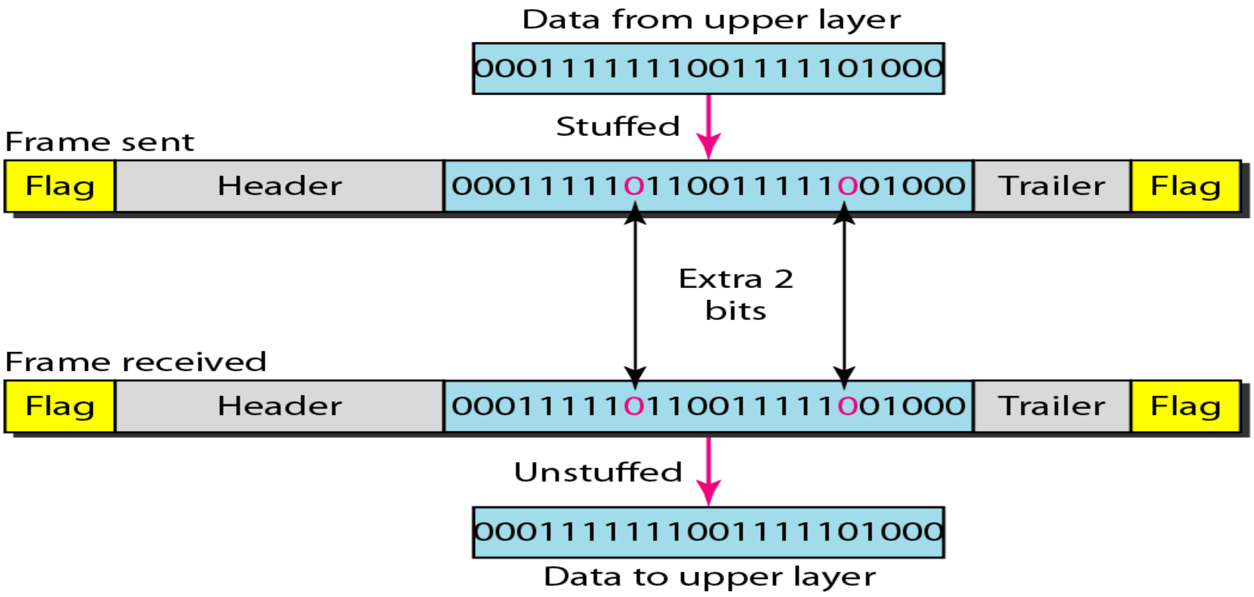


Figure (b)

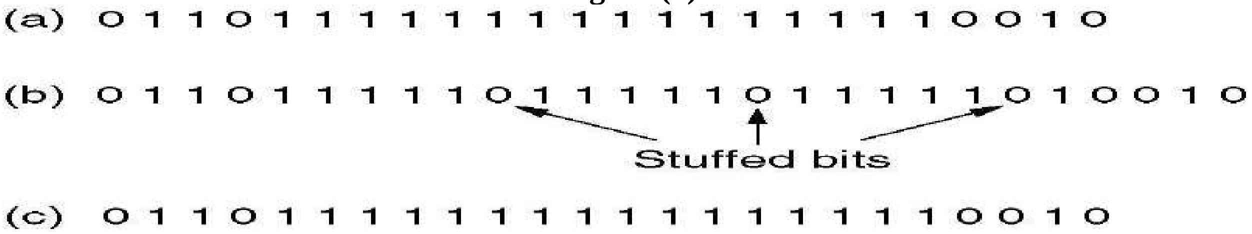

(a) 0 1 1 0 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 0 0 1 0
(b) 0 1 1 0 1 1 1 1 1 0 1 1 1 1 1 0 1 1 1 1 1 0 1 0 0 1 0
Stuffed bits
(c) 0 1 1 0 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 0 0 1 0

Figure (c)

(a) The original data.
(b) The data as they appear on the line.
(c) The data as they are stored in receiver's memory after destuffing.

4. Physical layer coding violation:

The last method of framing is only applicable to networks in which the encoding on the physical medium contains some redundancy

. For example, some LANs encode 1 bit of data by using 2 physical bits. Normally, a 1 bit is a high-low pair and a 0 bit is a low-high pair. The scheme means that every data bit has a transition in the middle, making it easy for the receiver to locate the bit boundaries. The combinations high-high and low-low are not used for data but are used for delimiting frames in some protocols.

As a final note on framing, many data link protocols use a combination of a character count with one of the other methods for extra safety. When a frame arrives, the count field is used to locate the end of the frame. Only if the appropriate delimiter is present at that position and the checksum is correct is the frame accepted as valid. Otherwise, the input stream is scanned for the next delimiter.

(2) ERROR CONTROL

- How do we make sure that all frames are eventually delivered to the network layer at the destination and in the proper order?
- Provide sender with some acknowledgement about what is happening with the receiver
- Sender could wait for acknowledgement

Disadvantages

- If a frame vanishes, the receiver will not send an acknowledgement thus, sender will wait forever
- Dealt with by timers and sequence numbers – important part of DLL
- Sender transmits a frame, starts a timer.
- Timer set to expire after interval long enough for frame to reach destination, be processed, and have acknowledgement sent to sender
- Is a danger of frame being transmitted several times, however dealt with by assigning sequence numbers to outgoing frames, so that receiver can distinguish retransmissions from originals.

(3) FLOW CONTROL

What do we do when a sender transmits frames faster than the receiver can accept them?

- **Feedback-based flow control** – receiver sends back information to the sender, giving it permission to send more data or at least telling the sender how the receiver is doing
- **Rate-based flow control** – the protocol has a built-in mechanism that limits the rate at which the sender may transmit data, using feedback from the receiver.

ERROR DETECTION AND CORRECTION METHODS

- Because of Attenuation, distortion, noise and interferences, errors during transmission are inevitable, leading to corruption transmitted bits.

- Longer the frame size and higher the probability of single bit error, lower is the probability receiving a frame without error.

ERROR

- When data is being transmitted from one machine to another, it may possible that data become corrupted on its way. Some of the bits may be altered, damaged or lost during transmission. Such a condition is known as **error**.

TYPES OF ERRORS

- **Single bit error**: Only one bit gets corrupted. Common in Parallel transmission.

- **Burst error:** More than one bit gets corrupted very common in serial transmission of data occurs when the duration of noise is longer than the duration of one bit.

Single bit error:

- The term single-bit error means that only one bit of given data unit (such as a byte, character, or data unit) is changed from 1 to 0 or from 0 to 1 as shown in Fig. 3.2.1.
- Single bit errors are least likely type of errors in serial data transmission.
- For example, if 16 wires are used to send all 16 bits of a word at the same time and one of the wires is noisy, one bit is corrupted in each word.

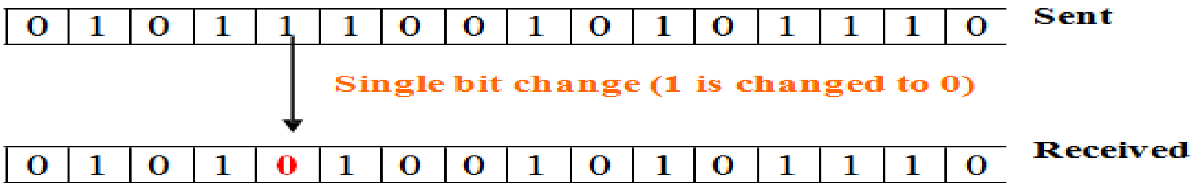


Figure 3.2.1 Single bit error

Burst error:

- More than one bit gets corrupted very common in serial transmission of data occurs when the duration of noise is longer than the duration of one bit.
- The noise affects data; it affects a set of bits.
- The number of bits affected depends on the data rate and duration of noise.

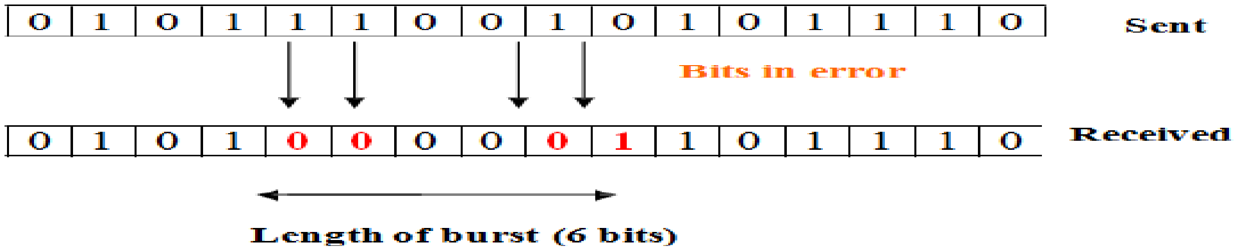


Figure 3.2.2 Burst Error

ERROR DETECTION TECHNIQUES

Basic approach used for error detection is the use of redundancy, where additional bits are added to facilitate detection and correction of errors. Popular techniques are

- Simple Parity check
- Two-dimensional Parity check
- Checksum
- Cyclic redundancy check

Redundancy is the method in which some extra bits are added to the data so as to check whether the data contain error or not.

m - data bits (i.e., message bits)
r - redundant bits (or check bits).
n - total number of bits
n= (m + r).

An n-bit unit containing data and check-bits is often referred to as an n-bit codeword.

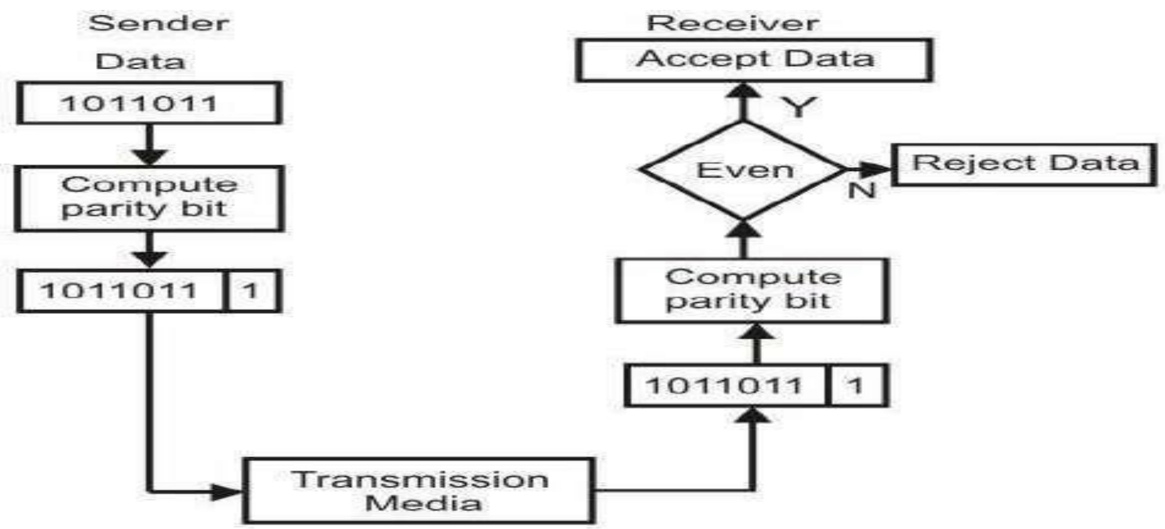
SIMPLE PARITY CHECK

The simplest and most popular error detection scheme. Appends a Parity bit to the end of the data.

Even Parity: **01000001** – Number of ones in the group of bits is even
Odd Parity: **11000001 -** Number of ones in the group of bits is odd

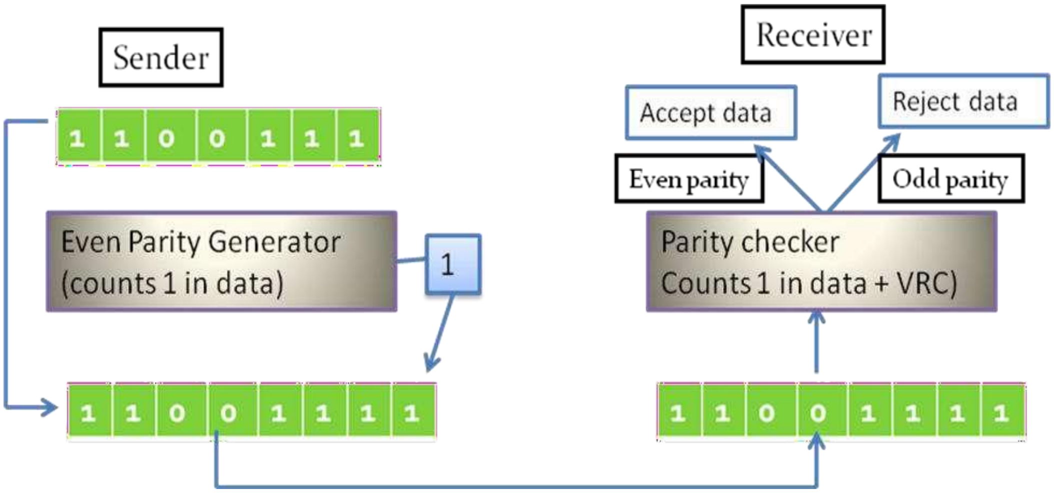
A parity of 1 is added to the block if it contains an odd number of 1's (ON bits) and 0 is added if it contains an even number of 1's. At the receiving end the parity bit is computed from the received data bits and compared with the received parity bit.

This scheme makes the total number of 1's even, that is why it is called *even parity checking*. Considering a 4-bit word, different combinations of the data words and the corresponding code words are given in Table 3.2.1.



Decimal value	Data Block	Parity bit	Code word
0	0000	0	00000
1	0001	1	00011
2	0010	1	00101
3	0011	0	00110
4	0100	1	01001
5	0101	0	01010
6	0110	0	01100
7	0111	1	01111
8	1000	1	10001
9	1001	0	10010
10	1010	0	10100
11	1011	1	10111
12	1100	0	11000
13	1101	1	11011
14	1110	1	11101
15	1111	0	11110

Example:

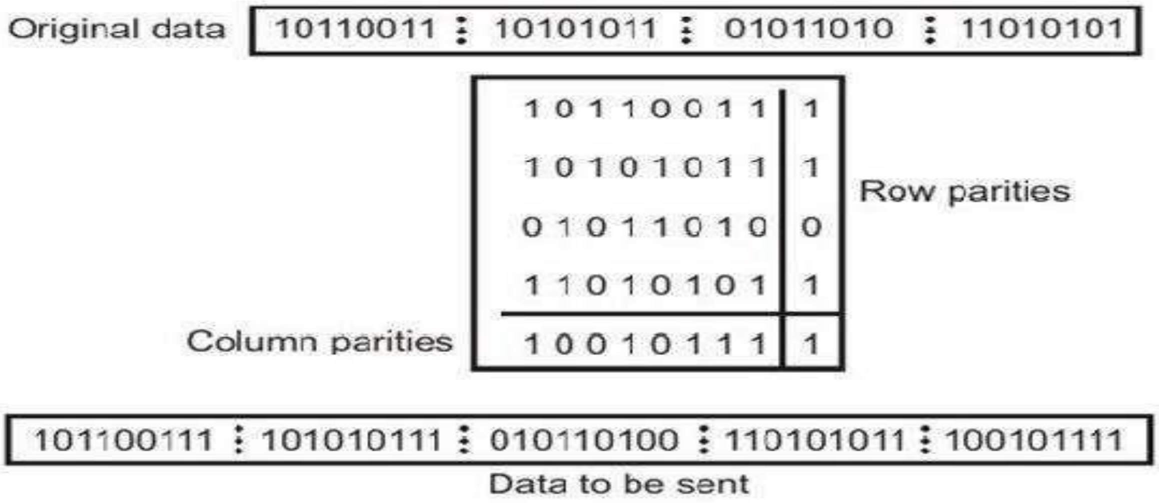


PERFORMANCE OF SIMPLE PARITY CHECK

- Simple parity check can **detect all single-bit error**
- It can also detect burst error, if the number of bits in **even or odd.**
- The technique is not foolproof against burst errors that **invert more than one bit**. If an even number of bits is inverted due to error, the **error is not detected.**

TWO-DIMENSION PARITY CHECKING

- Performance can be improved by using two dimensional parity check, which **organizes the block of bits in the form of table.**
- Parity check bits are **calculated from each row**, which is equivalent to a simple parity check.
- Parity check bits are also **calculated for all columns.**
- Both are sent along with the data.
- At the receiving end these are compared with the parity bits calculated on the received data.



Original data 10110011 ⋮ 10101011 ⋮ 01011010 ⋮ 11010101

	1 0 1 1 0 0 1 1	1
	1 0 1 0 1 0 1 1	1
	0 1 0 1 1 0 1 0	0
	1 1 0 1 0 1 0 1	1
Column parities	1 0 0 1 0 1 1 1	1

101100111 ⋮ 101010111 ⋮ 010110100 ⋮ 110101011 ⋮ 100101111

Data to be sent

Figure 3.2.4 Two-dimension Parity Checking

Performance:

- If two bits in one data unit are damaged and two bits in exactly same position in another data unit are also damaged, The 2-D Parity check **checker will not detect an error**.
- For example, if two data units: **11001100 and 10101100.**
- If first and second from last bits in each of them is changed, making the data units as **01001110 and 00101110**, the error cannot be detected by 2-D Parity check.

CHECKSUM

- In checksum error detection scheme, the **data is divided into k segments each of m bits.**
- In the sender's end the segments are added using **1's complement arithmetic to get the sum.**
- The sum is complemented to get the checksum. The **checksum** segment is sent **along with the data segments**

Example 1:

Sender **Reciever:**

```
10101001        subunit1
00111001        subunit 2
 ______
11100010        sum
 ______
00011101        Complement of sum
```

10101001	00111001	00011101
Data		checksum

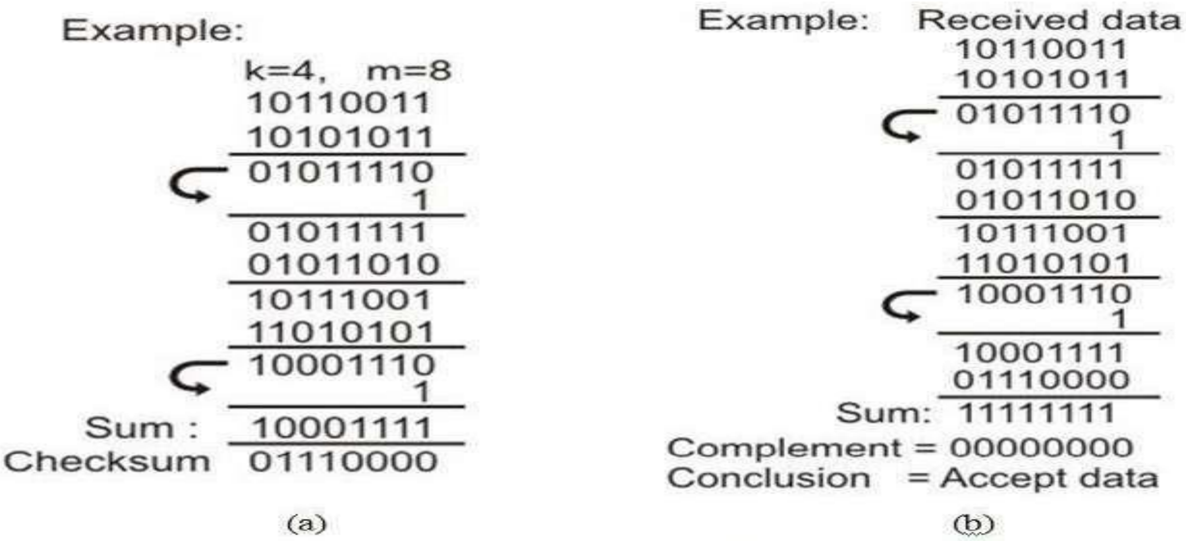
```
10101001      subunit1
00111001      subunit2
00011101      Checksum

11111111      sum

00000000      complement
```

Conclusion = Accept data.

Example 2: K= 10110011, 10101011, 01011111, 11010101



```
Example:
              k=4,   m=8
              10110011
              10101011
             ----------
          ↰   01011110
                     1
             ----------
              01011111
              01011010
             ----------
              10111001
              11010101
             ----------
          ↰   10001110
                     1
             ----------
   Sum :      10001111
             ----------
 Checksum     01110000
                (a)
```

```
Example:   Received data
              10110011
              10101011
             ----------
          ↰   01011110
                     1
             ----------
              01011111
              01011010
             ----------
              10111001
              11010101
             ----------
          ↰   10001110
                     1
             ----------
              10001111
              01110000
             ----------
       Sum:   11111111
Complement = 00000000
Conclusion  = Accept data
                (b)
```

Figure 3.2.5 (a) Sender's end for the calculation of the checksum, (b) Receiving end for checking the checksum

<u>CYCLIC REDUNDANCY CHECK</u>

- One of the most powerful and commonly used error detecting codes.

Basic approach:

- Given a m-bit block of bit sequence, the sender generates an n-bit sequence known as **frame sequence check(FCS),** so that the resulting frame, consisting of m+n bits exactly divisible by **same predetermined number.**

- The receiver divides the incoming frame by that number and, if there is **no reminder, assumes there was no error.**

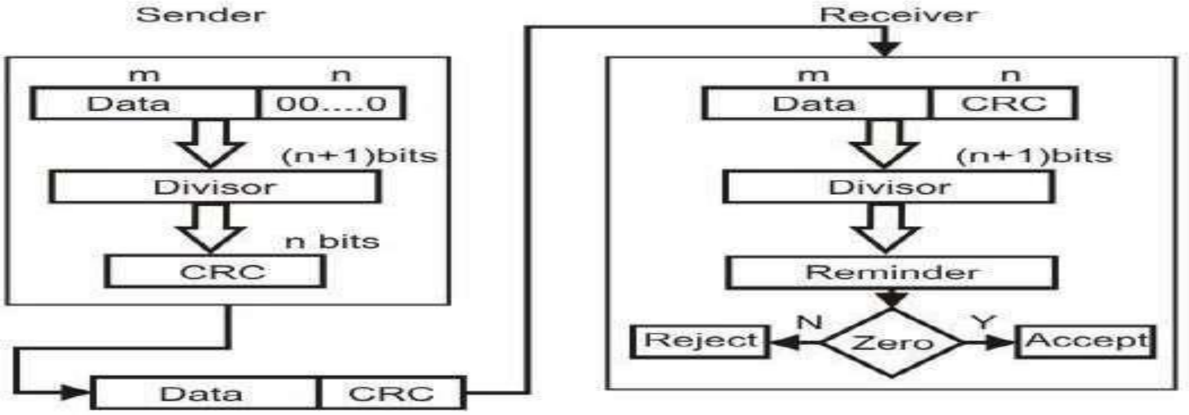


Fig. 3.2.7 by dividing a sample 4- bit number by the coefficient of the generator polynomial x^3+x+1, which is 1011, using the modulo-2 arithmetic.

Modulo-2 arithmetic is a binary addition process without any carry over, which is just the Exclusive-OR operation.

Consider the case where k=1101. Hence we have to divide 1101000 (i.e. *k* appended by 3 zeros) by 1011, which produces the remainder r=001, so that the bit frame *(k+r)* =1101001 is actually being transmitted through the communication channel.

At the receiving end, if the received number, i.e.,1101001 is divided by the same generator polynomial 1011 to get the remainder as 000, it can be assumed that the data is free of errors.

Sender: Sender transmit the data along with remainder (**CRC**)

```
      1111  Quotient                 Data: 1101

1011| 1101000  <-----  K             Divisor: 1011
      -------
      1011
        1100
        ----
        1011
          1110
          ----
          1011
            1010              Data to be sent:    1 1 0 1 0 0 1
            ----
            1011                                  Data | CRC
              001 Reminder (r)
```

Receiver:

```
         1111   Quotient              Data: 1101001
       ___________________
  1011 | 1101001                      Divisor: 1011
       |
       | 1011
        ----------
          1100
          1011
          --------
           1110
           1011
           --------
            1011
            1011
            --------
             000   Reminder
```

Note: Remainder is zero, no error. Receiver can accept the data.

Performance of CRC

- CRC can detect all single-bit errors.
- CRC can detect all double-bit errors(three1's)
- CRC can detect any odd number of errors of less than the degree of the polynomial.
- CRC detects most of the larger burst errors with a high probability.

ERROR CORRECTING CODES

Concept of error-correction can be easily understood by examining the simplest case of single-bit errors. As we have already seen that a single-bit error can be detected by addition of a parity bit with the data, which needed to be send.

A single additional bit can detect error, but it's not sufficient enough to correct that error too. For correcting an error one has to know the exact position of error, i.e. exactly which bit is in error (to locate the invalid bits).

For example, to correct a single-bit error in an ASCII character, the error correction must determine which one of the seven bits is in error. To this, we have to add some additional redundant bits.

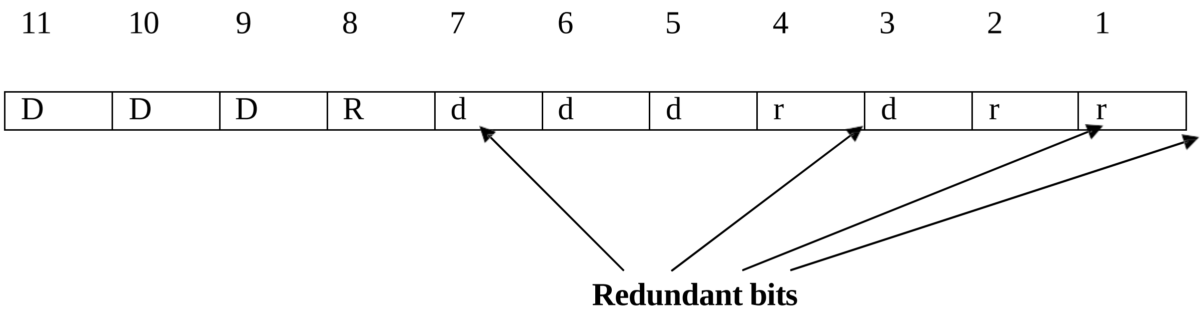
To calculate the numbers of redundant bits (r) required to correct d data bits, let us find out the relationship between the two. So we have (d+r) as the total number of bits, which are to be transmitted; then r must be able to indicate at least d+r+1 different values. Of these, one value means no error, and remaining d+r values indicate error location of error in each of d+r locations. So, d+r+1 states must be distinguishable by r bits, and r bits can indicates 2^r states. Hence, 2^r must be greater than d+r+1.

$$\mathbf{2^r >= d+r+1}$$

The value of r must be determined by putting in the value of d in the relation. For example, if d is 7, then the smallest value of r that satisfies the above relation is 4. So the total bits, which are to be transmitted is 11 bits (d+r = 7+4 =11).

Now let us examine how we can manipulate these bits to discover which bit is in error. A technique developed by R.W. Hamming provides a practical solution. The solution or coding scheme he developed is commonly known as **Hamming Code**.

Hamming code can be applied to data units of any length and uses the relationship between the data bits and redundant bits as discussed.

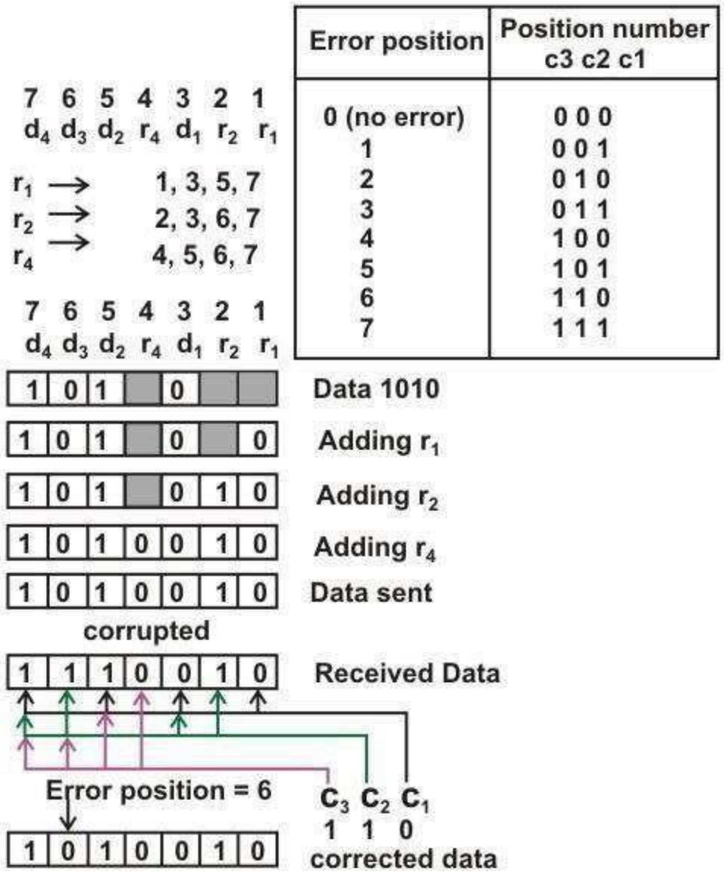


11	10	9	8	7	6	5	4	3	2	1
D	D	D	R	d	d	d	r	d	r	r

Figure 3.2.8 Positions of redundancy bits in hamming code

Basic approach for error detection by using Hamming code is as follows:

- To each group of m information bits k parity bits are added to form (m+k) bit code as shown in Fig. 3.2.8.
- Location of each of the (m+k) digits is assigned a decimal value.
- The k parity bits are placed in positions 1, 2, …, 2^{k-1} positions.–K parity checks are performed on selected digits of each codeword.
- At the receiving end the parity bits are recalculated. The decimal value of the k parity bits provides the bit-position in error, if any.



Error position	Position number c3 c2 c1
0 (no error)	0 0 0
1	0 0 1
2	0 1 0
3	0 1 1
4	1 0 0
5	1 0 1
6	1 1 0
7	1 1 1

Figure 3.2.9 Use of Hamming code for error correction for a 4-bit data

Figure 3.2.9 shows how hamming code is used for correction for 4-bit numbers (d4d3d2d1) with the help of three redundant bits (r3r2r1).

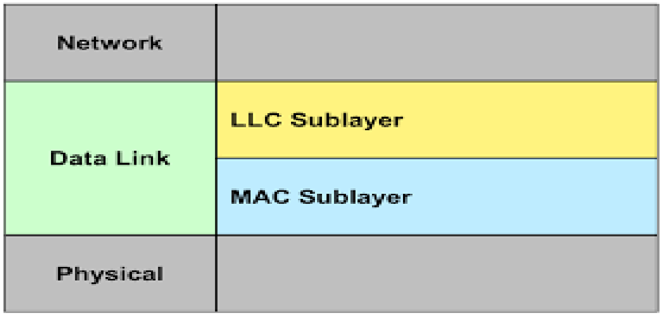
For the example data 1010, first r1 (0) is calculated considering the parity of the bit positions, 1, 3, 5 and 7. Then the parity bits r2 is calculated considering bit positions 2, 3, 6 and 7.

Finally, the parity bits r4 is calculated considering bit positions 4, 5, 6 and 7 as shown. If any corruption occurs in any of the transmitted code 1010010, the bit position in error can be found out by calculating r3r2r1 at the receiving end.

For example, if the received code word is 1110010, the recalculated value of r3r2r1 is 110, which indicates that bit position in error is 6, the decimal value of 110.

MEDIUM ACCESS CONTROL SUBLAYER

The MAC sublayer is the bottom part of the data link layer. The protocols used to determine who goes next on a multiaccess channel belong to a sublayer of the data link layer called the **MAC** (**Medium Access Control**) sublayer. The MAC sublayer is especially important in LANs, particularly wireless ones because wireless is naturally a broadcast channel. broadcast channels are sometimes referred to as **multi-access channels** or **random access channels**.



THE CHANNEL ALLOCATION PROBLEM

The channel allocation problem is how to allocate a single broadcast channel among competing users.

I. RANDOM ACCESS PROTOCOLS

- In a random access protocol, a transmitting node always transmits at the full rate of the channel, namely, R bps.
- When there is a collision, each node involved in the collision repeatedly retransmits its frame(that is ,packet) until the frame gets through without a collision. But when a node experiences a collision, it doesn't necessarily retransmitting the frame right away. Instead it waits a random delay before retransmitting the frame.
- Each node involved in a collision chooses independent random delays .Because the random delays are independently chosen, it is possible that one of the nodes will pick a delay that is sufficiently less than the delays of the other colliding nodes and will therefore be able to sneak its frame into the channel without a collision.

The most commonly used random access protocols

1. The ALOHA protocol ,
2. CSMA (carrier sense multiple access) protocol ,
3. CSMA/CD (carrier sense multiple access /collision detection) protocol and

4. Collision-Free Protocols

5. Limited-Contention Protocols

1. ALOHA

- In the 1970s, Norman Abramson and his colleagues at the University of Hawaii devised a new and elegant

  method to solve the channel allocation problem.
- Although Abramson's work, called the ALOHA system, used ground-based radio broadcasting, the basic idea is applicable to any system in which uncoordinated users are competing for the use of a single shared channel.
- The two versions of ALOHA here: **pure and slotted**.
- They differ with respect to whether time is divided into discrete slots into which all frames must fit.
- Pure ALOHA does not require global time synchronization; slotted ALOHA does.

PURE ALOHA

- If you have data to send, send the data
- If the message collides with another transmission, try resending "later"

Note that the first step implies that Pure ALOHA does not check whether the channel is busy before transmitting. The critical aspect is the "later" concept: the quality of the backoff scheme chosen significantly influences the efficiency of the protocol, the ultimate channel capacity, and the predictability of its behavior.

A sketch of frame generation in an ALOHA system is given in Fig. 4-1. We have made the frames all the same length because the throughput of ALOHA systems is maximized by having a uniform frame size rather than by allowing variable length frames.

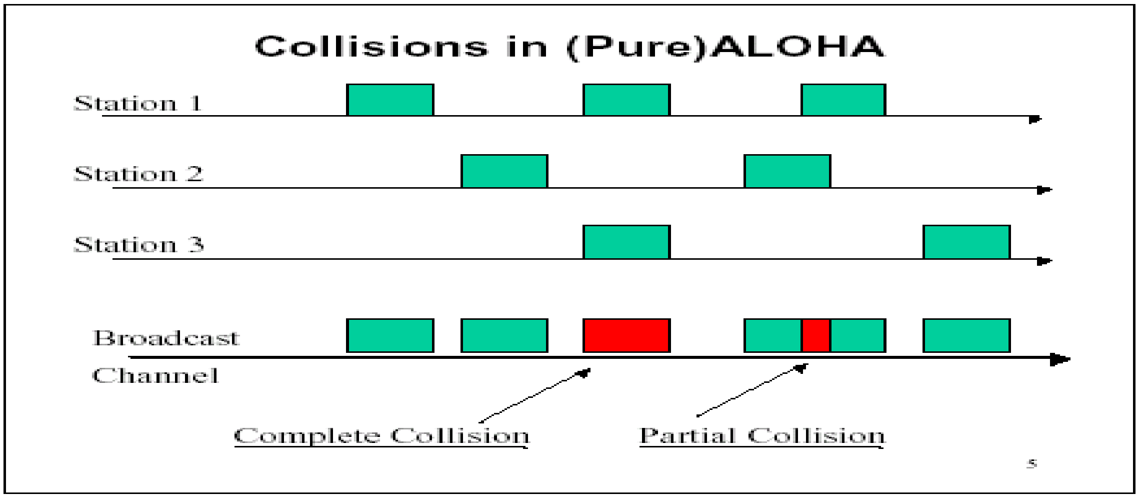
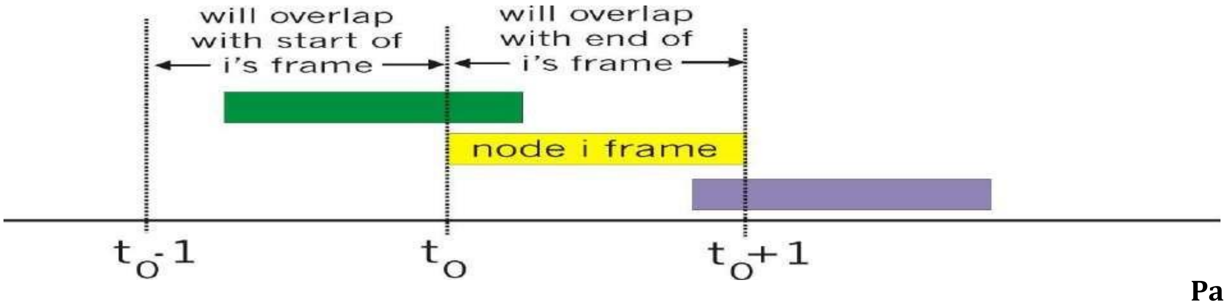


Figure 4-1. In pure ALOHA, frames are transmitted at completely arbitrary times.

To assess Pure ALOHA, we need to predict its throughput, the rate of (successful) transmission of frames. First, let's make a few simplifying assumptions:

- All frames have the same length.
- Stations cannot generate a frame while transmitting or trying to transmit.
- The population of stations attempts to transmit (both new frames and old frames that collided) according to a Poisson distribution.

Let "*T*" refer to the time needed to transmit one frame on the channel, and let's define "frame-time" as a unit of time equal to *T*. Let "*G*" refer to the mean used in the Poisson distribution over transmission-attempt amounts: that is, on average, there are *G* transmission-attempts per frame-time.



will overlap with start of i's frame
will overlap with end of i's frame
node i frame
t_o-1
t_o
t_o+1

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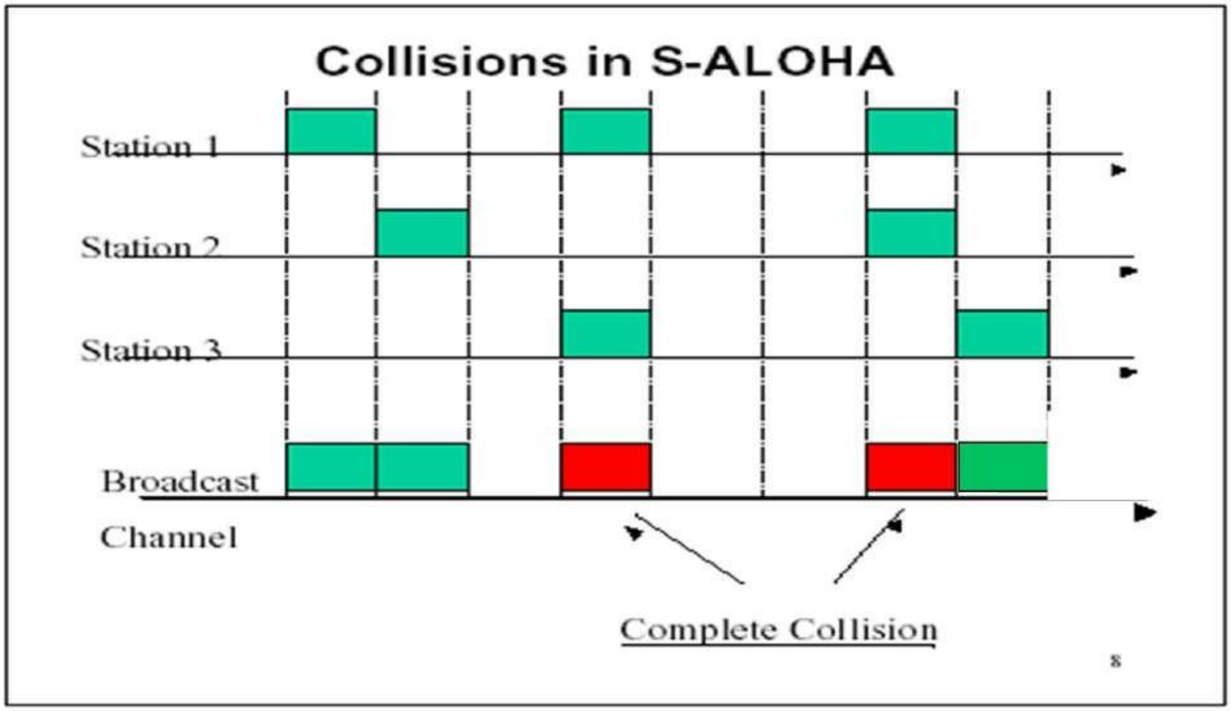
SLOTTED ALOHA

- An improvement to the original ALOHA protocol was "Slotted ALOHA", which introduced discrete timeslots and increased the maximum throughput.
- A station can send only at the beginning of a timeslot, and thus collisions are reduced. In this case, we only need to worry about the transmission-attempts within 1 frame-time and not 2 consecutive frame-times, since collisions can only occur during each timeslot. Thus, the probability of there being zero transmission-attempts in a single timeslot is:



- Slotted ALOHA is used in low-data-rate tactical satellite communications networks by military forces, in subscriber-based satellite communications networks, mobile telephony call setup, and in the contactless RFID technologies.

Pros

- single active node can continuously transmit at full rate of channel
- highly decentralized: only slots in nodes need to be in sync
- simple

Cons

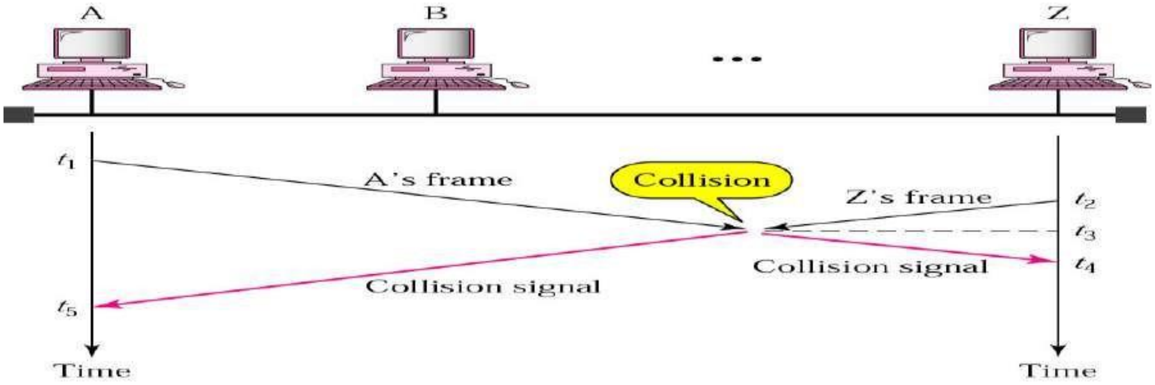
- collisions, wasting slots
- idle slots
- nodes may be able to detect collision in less than time to transmit packet
- clock synchronization

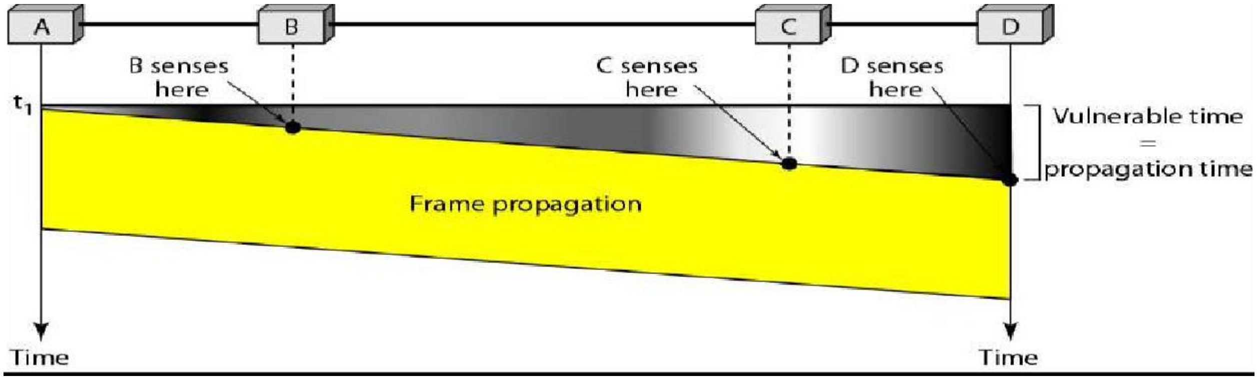
CARRIER SENSE MULTIPLE ACCESS

Carrier Sense Multiple Access (**CSMA**) is a probabilistic Media Access Control (MAC) protocol in which a node verifies the absence of other traffic before transmitting on a shared transmission medium, such as an electrical bus, or a band of the electromagnetic spectrum.

"**Carrier Sense**" describes the fact that a transmitter uses feedback from a receiver that detects a carrier wave before trying to send. That is, it tries to detect the presence of an encoded signal from another station before attempting to transmit. If a carrier is sensed, the station waits for the transmission in progress to finish before initiating its own transmission.

"**Multiple Access**" describes the fact that multiple stations send and receive on the medium. Transmissions by one node are generally received by all other stations using the medium.





ADVANTAGES

- Fairly simple to implement
- Functional scheme that works

DISADVANTAGES

- Cannot recover from a collision (inefficient waste of medium time)

Types of CSMA Protocols :

Different CSMA protocols that determine:

- What a station should do when the medium is idle?
- What a station should do when the medium is busy?

1. Non-Persistent CSMA
2. 1-Persistent CSMA
3. p-Persistent CSMA

1) Nonpersistent CSMA :

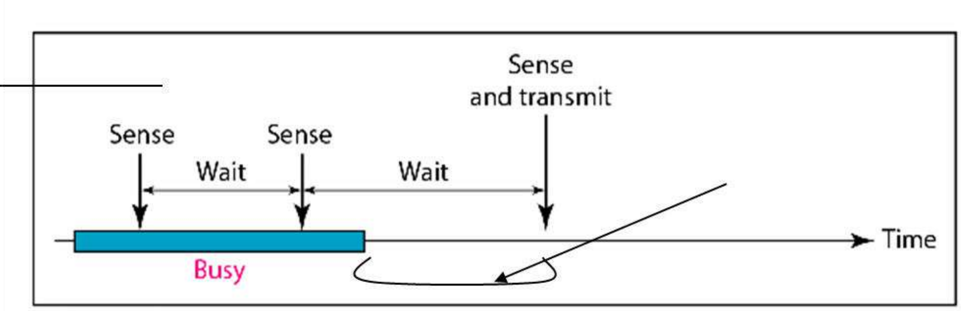
- A station with frames to be sent, should sense the medium

1. If medium is idle, transmit; otherwise, go to 2
2. If medium is busy, (backoff) wait a *random* amount of time and repeat 1

- Non-persistent Stations are deferential (respect others)
- Performance:

1. Random delays reduce probability of collisions because two stations with data to be transmitted will wait for different amount of times.
2. Bandwidth is wasted if waiting time (backoff) is large because medium will remain idle following end of transmission even if one or more stations have frames to send

Random Waiting times



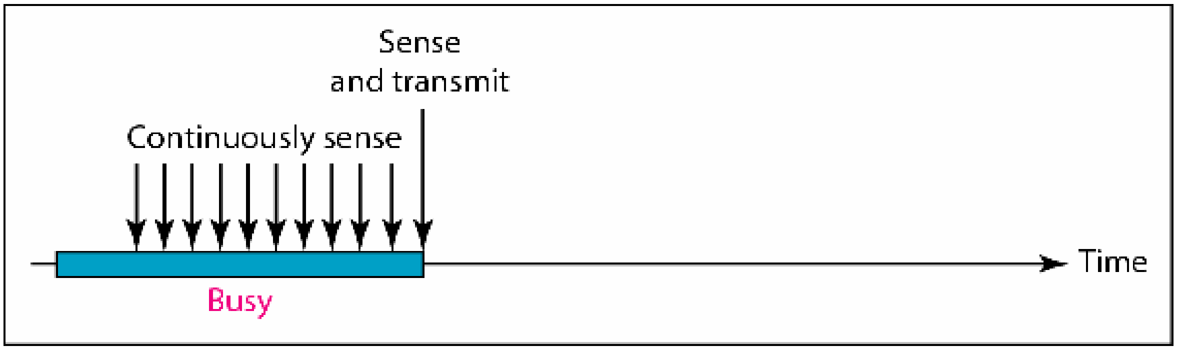
b. Nonpersistent

2) 1-persistent CSMA: To avoid idle channel time, 1-persistent protocol used

- Station wishing to transmit listens to the medium:

1. If medium idle, **transmit** immediately;
2. If medium busy, **continuously listen** until medium becomes idle; then transmit immediately with probability 1.

- Performance
  - 1-persistent stations are **selfish**
  - If two or more stations becomes ready at the same time, **collision guaranteed**

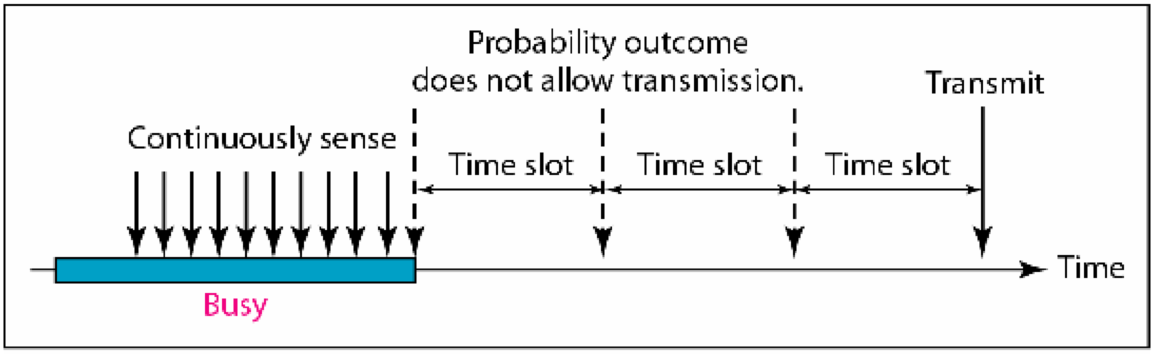


a. 1-persistent

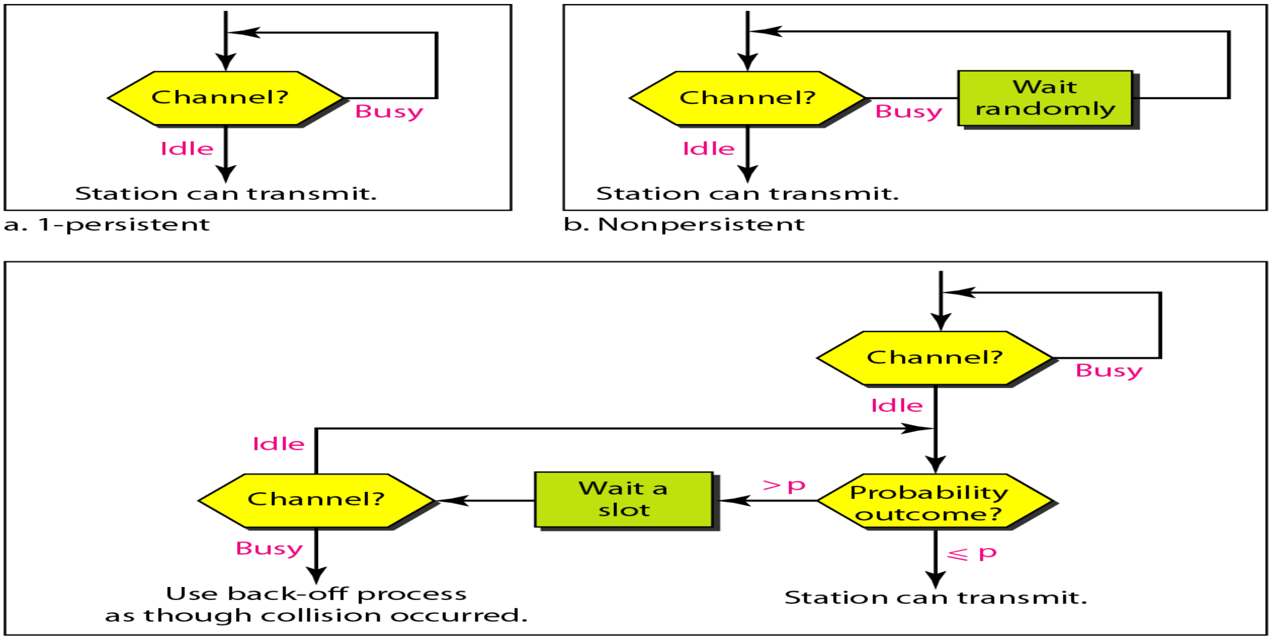
3).P-persistent CSMA: Time is divided to slots where each Time unit (slot) typically equals **maximum propagation delay**

- Station wishing to transmit listens to the medium:

1. If medium idle,
   - transmit with probability (**p**), OR
   - wait **one time unit (slot)** with probability (**1 – p**), then repeat 1.
2. If medium busy, **continuously listen until idle** and repeat step **1**
3. Performance
   - Reduces the possibility of collisions like **nonpersistent**
   - Reduces channel idle time like **1-persistent**



c. p-persistent



a. 1-persistent

b. Nonpersistent

c. p-persistent

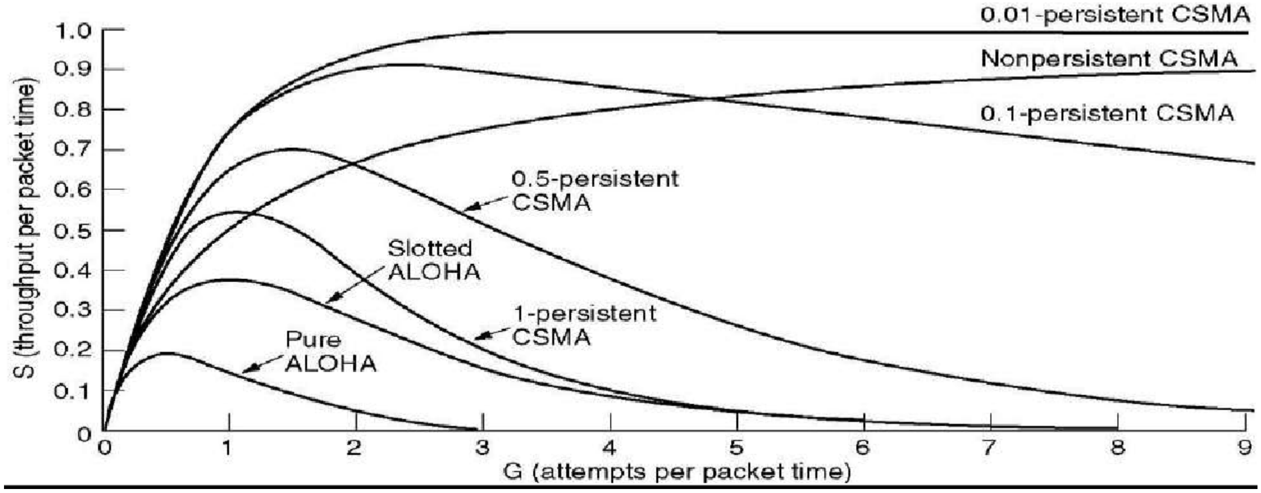
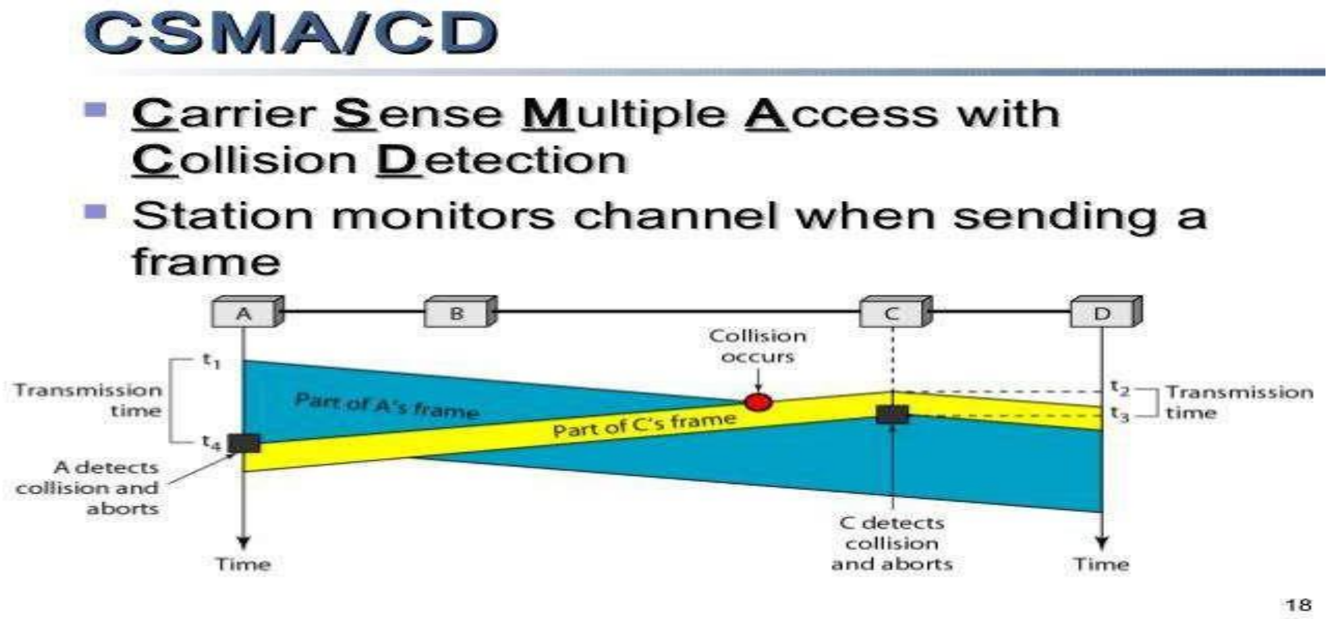


Fig: Comparison of the channel utilization versus load for various random access protocols.

CSMA/CD (Collision Detection)

Carrier sense multiple access with collision detection (**CSMA/CD**) is a Media Access Control method in which.

- a carrier sensing scheme is used.
- a transmitting data station that detects another signal while transmitting a frame, stops transmitting that frame, transmits a **jam signal**, and then waits for a random time interval before trying to resend the frame.
- CSMA/CD is a modification of pure carrier sense multiple access (CSMA). CSMA/CD is used to improve CSMA performance by terminating transmission as soon as a collision is detected, thus shortening the time required before a retry can be attempted.



CSMA/CD

- Carrier Sense Multiple Access with Collision Detection
- Station monitors channel when sending a frame

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ALGORITHM : The following procedure is used to initiate a transmission. The procedure is complete when the frame is transmitted successfully or a collision is detected during transmission

When a station wants to send some information, it uses the following algorithm.

Main procedure

1. Is a frame ready for transmission? If not, wait for a frame.
2. Is medium idle? If not, wait until it becomes ready
3. Start transmitting and monitor for collision during transmission.
4. Did a collision occur? If so, go to collision detected procedure.
5. Reset retransmission counters and end frame transmission.

Collision detected procedure:

The following procedure is used to resolve a detected collision. The procedure is complete when retransmission is initiated or the retransmission is aborted due to numerous collisions.

1. Continue transmission until minimum packet time is reached to ensure that all receivers detect the collision.
2. Increment retransmission counter.
3. Was the maximum number of transmission attempts reached? If so, abort transmission.
4. Calculate and wait random backoff period based on number of collisions.
5. Re-enter main procedure at stage 1.

Methods for collision detection are media dependent, but on an electrical bus such as 10BASE-5 or 10BASE-2, collisions can be detected by comparing transmitted data with received data or by recognizing a higher than normal signal amplitude on the bus.

JAM SIGNAL

The **jam signal** is a signal that carries a 32-bit binary pattern sent by a data station to inform the other stations that they must not transmit.

ADVANTAGES

More efficient than basic CSMA

DISADVANTAGES

Requires ability to detect collisions

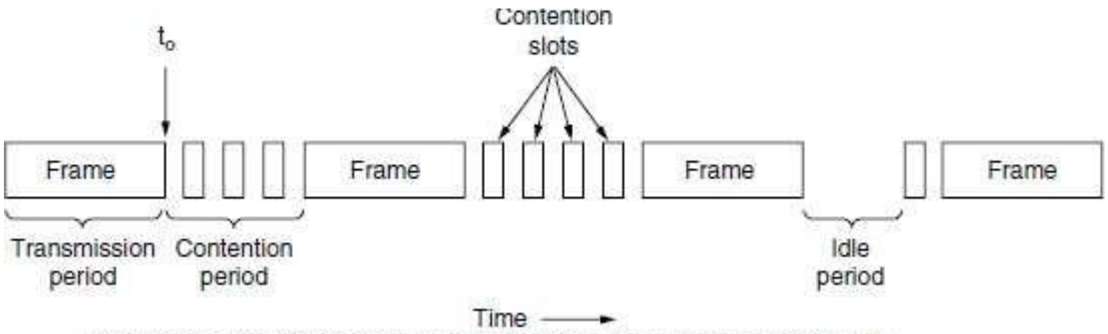

Figure 4-5. CSMA/CD can be in contention, transmission, or idle state.

Ethernet

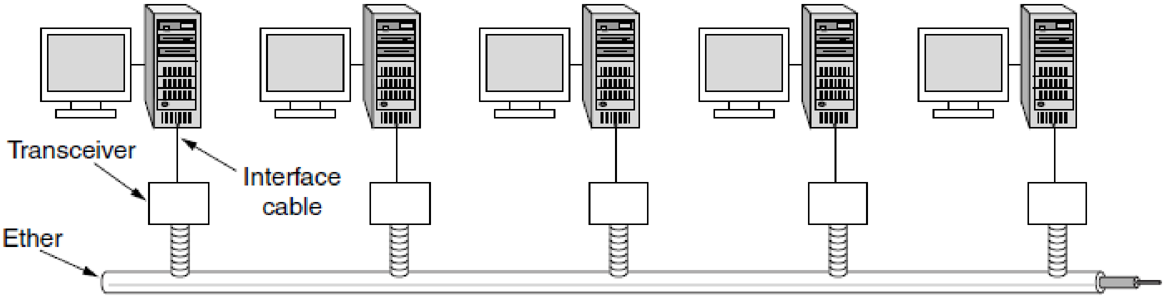

Fig : Architecture of classic Ethernet

Ethernet :

Ethernet is a family of computer networking technologies commonly used in local area networks (LAN), metropolitan area networks (MAN) and wide area networks (WAN). It was commercially introduced in 1980 and first standardized in 1983 as IEEE 802.3. Ethernet has since retained a good deal of backward compatibility and has been refined to support higher bit rates, a greater number of nodes, and longer link distances. Over time, Ethernet has largely replaced competing wired LAN technologies such as Token Ring, FDDI(**Fiber Distributed Data Interface**) (**FDDI**). and ARCNET. **(Attached Resource Computer NETwork** (**ARCNET** or **ARCnet**))

The original 10BASE5 Ethernet uses coaxial cable as a shared medium, while the newer Ethernet variants use twisted pair and fiber optic links in conjunction with switches. Over the course of its history, Ethernet data transfer rates have been increased from the original 2.94 megabits per second (Mbit/s) to the latest 400 gigabits per second (Gbit/s). The Ethernet standards comprise several wiring and signaling variants of the OSI physical layer in use with Ethernet.

Switched Ethernet : switched Ethernet. An Ethernet LAN that uses switches to connect individual hosts or segments. In the case of individual hosts, the switch replaces the repeater and effectively gives the device full 10 Mbps bandwidth (or 100 Mbps for Fast Ethernet) to the rest of the network

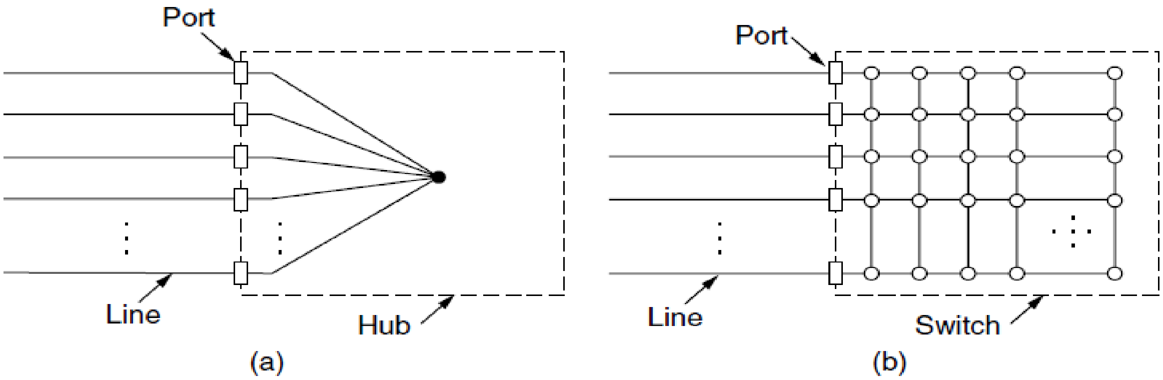

Fig : (a) Hub. (b) Switch.