UNIT - I

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.

“Computer network’’ to mean a collection of autonomous computers interconnected by a single technology. Two computers are said to be interconnected if they are able to exchange information.

The connection need not be via a copper wire; fiber optics, microwaves, infrared, and communication satellites can also be used.

Networks come in many sizes, shapes and forms, as we will see later. They are usually connected together to make larger networks, with the **Internet** being the most well-known example of a network of networks.

There is considerable confusion in the literature between a **computer network** and a **distributed system**. The key distinction is that in a distributed system, a collection of independent computers appears to its users as a single coherent system. Usually, it has a single model or paradigm that it presents to the users. Often a layer of software on top of the operating system, called **middleware**, is responsible for implementing this model. A well-known example of a distributed system is the **World Wide Web**. It runs on top of the Internet and presents a model in which everything looks like a document (Web page).

USES OF COMPUTER NETWORKS

1. Business Applications

- to distribute information throughout the company (**resource sharing).** sharing physical resources such as printers, and tape backup systems, is sharing information
- **client-server model**. It is widely used and forms the basis of much network usage.
- **communication medium** among employees.**email** (**electronic mail**), which employees generally use for a great deal of daily communication.
- Telephone calls between employees may be carried by the computer network instead of by the phone company. This technology is called **IP telephony** or **Voice over IP** (**VoIP**) when Internet technology is used.
- **Desktop sharing** lets remote workers see and interact with a graphical computer screen
- doing business electronically, especially with customers and suppliers. This new model is called **e-commerce** (**electronic commerce**) and it has grown rapidly in recent years.

2 Home Applications

- **peer-to-peer** communication
- person-to-person communication

- electronic commerce
- entertainment.(game playing,)

3 Mobile Users

- Text messaging or texting
- Smart phones,
- GPS (Global Positioning System)
- m-commerce
- NFC (Near Field Communication)

4 Social Issues

With the good comes the bad, as this new-found freedom brings with it many unsolved social, political, and ethical issues.

Social networks, message boards, content sharing sites, and a host of other applications allow people to share their views with like-minded individuals. As long as the subjects are restricted to technical topics or hobbies like gardening, not too many problems will arise.

The trouble comes with topics that people actually care about, like politics, religion, or sex. Views that are publicly posted may be deeply offensive to some people. Worse yet, they may not be politically correct. Furthermore, opinions need not be limited to text; high-resolution color photographs and video clips are easily shared over computer networks. Some people take a live-and-let-live view, but others feel that posting certain material (e.g., verbal attacks on particular countries or religions, pornography, etc.) is simply unacceptable and that such content must be censored. Different countries have different and conflicting laws in this area. Thus, the debate rages.

Computer networks make it very easy to communicate. They also make it easy for the people who run the network to snoop on the traffic. This sets up conflicts over issues such as **employee rights versus employer rights**. Many people read and write email at work. Many employers have claimed the right to read and possibly censor employee messages, including messages sent from a home computer outside working hours. Not all employees agree with this, especially the latter part.

Another conflict is centered around government versus citizen's rights.

A new twist with mobile devices is location privacy. As part of the process of providing service to your mobile device the network operators learn where you are at different times of day. This allows them to track your movements. They may know which nightclub you frequent and which medical center you visit.

Phishing ATTACK: ***Phishing*** is a type of social engineering ***attack*** often used to steal user data, including login credentials and credit card numbers. It occurs when an attacker, masquerading as a trusted entity, dupes a victim into opening an email, instant message, or text message.

BOTNET ATTACK: Botnets can be used to perform distributed denial-of-service attack (DDoS attack), steal data, send spam, and allows the attacker to access the device and its connection.

The effectiveness of a data communications system depends on four fundamental characteristics: delivery, accuracy, timeliness, and jitter.

I. **Delivery.** The system must deliver data to the correct destination. Data must be received by the intended device or user and only by that device or user.

2 **Accuracy.** The system must deliver the data accurately. Data that have been altered in transmission and left uncorrected are unusable.

3. **Timeliness**. The system must deliver data in a timely manner. Data delivered late are useless. In the case of video and audio, timely delivery means delivering data as they are produced, in the same order that they are produced, and without significant delay. This kind of delivery is called *real-time* transmission.

4. **Jitter**. Jitter refers to the variation in the packet arrival time. It is the uneven delay in the delivery of audio or video packets. For example, let us assume that video packets are sent every 30 ms. If some of the packets arrive with 30-ms delay and others with 40-ms delay, an uneven quality in the video is the result.

A data communications system has five components

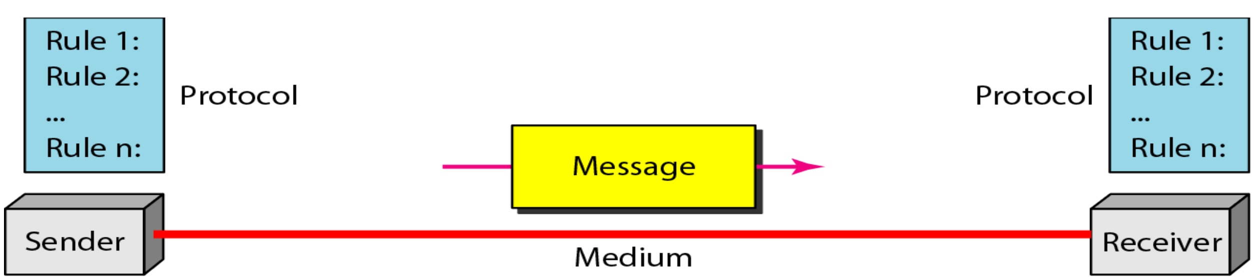
I. **Message**. The message is the information (data) to be communicated. Popular forms of information include text, numbers, pictures, audio, and video.

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

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

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

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

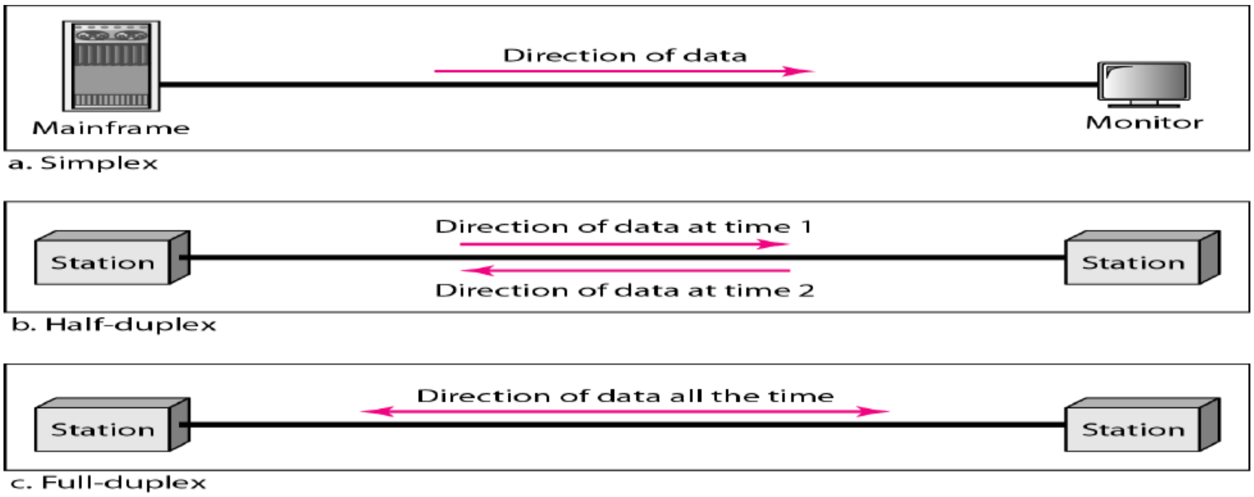


Data Representation

Text
Numbers
Images
Audio
Video

Data Flow

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



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 (Figure a). Keyboards and traditional monitors are examples of simplex devices.

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 (Figure b). Walkie-talkies and CB (citizens band) radios are both half-duplex systems.

Full-Duplex

In full-duplex, both stations can transmit and receive simultaneously (Figure c).
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.

Network Criteria

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

Performance

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

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

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

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

Physical Structures

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

Type of Connection

A network is two or more devices connected through links. A link is a communications pathway that transfers data from one device to another.

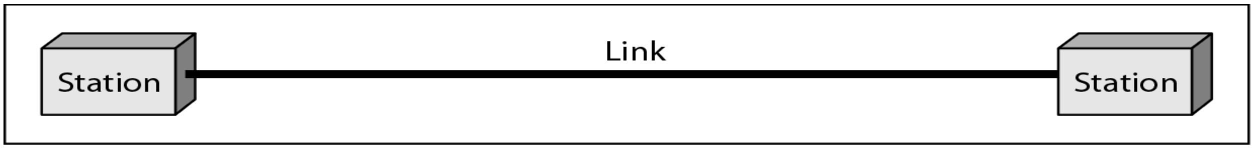
There are two possible types of connections: point-to-point and multipoint.

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

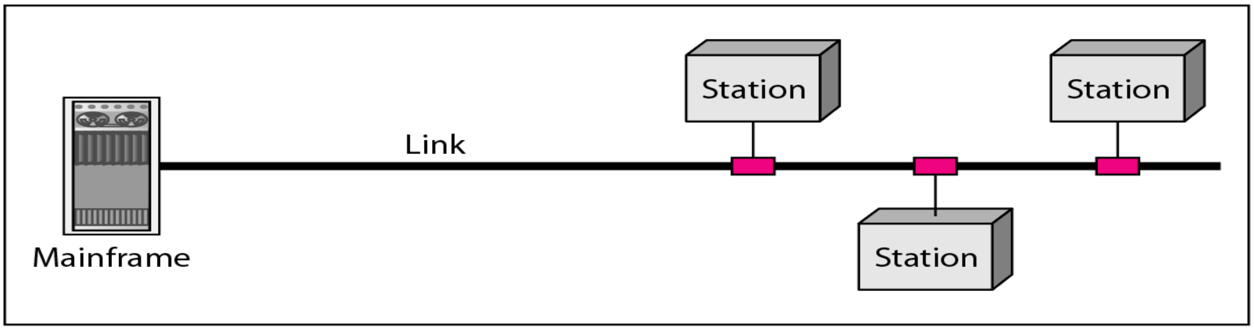
When you change television channels by infrared remote control, you are establishing a point-to-point connection between the remote control and the television's control system.

Multipoint A multipoint (also called multi-drop) connection is one in which more than two specific devices share a single link

In a multipoint environment, the capacity of the channel is shared, either spatially or temporally. If several devices can use the link simultaneously, it is a *spatially shared* connection. If users must take turns, it is a *timeshared* connection.



a. Point-to-point



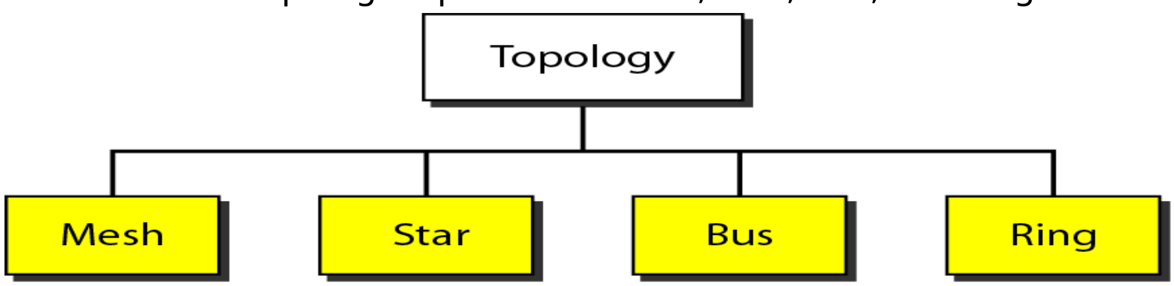
b. Multipoint

Physical Topology

The term *physical topology* refers to the way in which a network is laid out physically.

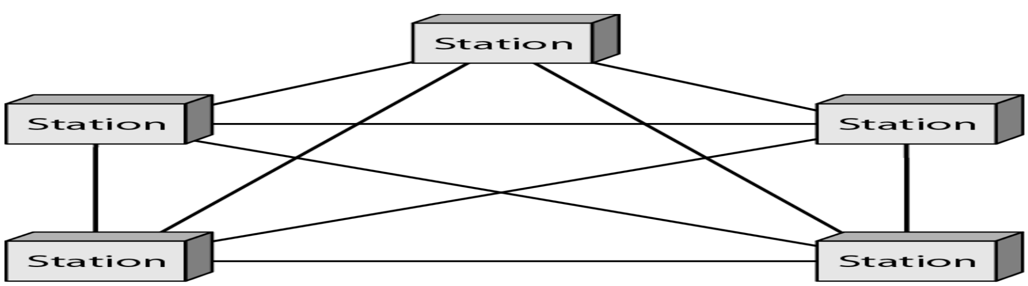
Two or more devices connect to a link; two or more links form a topology. The topology of a network is the geometric representation of the relationship of all the links and linking devices (usually called nodes) to one another.

There are four basic topologies possible: mesh, star, bus, and ring



MESH:

A mesh topology is the one where every node is connected to every other node in the network.



A mesh topology can be a **full mesh topology** or a **partially connected mesh topology**.

In a *full mesh topology*, every computer in the network has a connection to each of the other computers in that network. The number of connections in this network can be calculated using the following formula (***n*** is the number of computers in the network): **n(n-1)/2**

In a *partially connected mesh topology*, at least two of the computers in the network have connections to multiple other computers in that network. It is an inexpensive way to implement redundancy in a network. In the event that one of the primary computers or connections in the network fails, the rest of the network continues to operate normally.

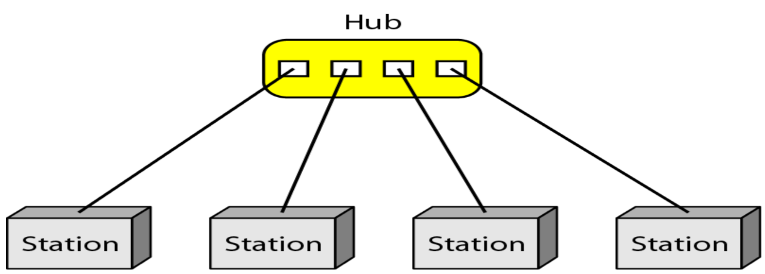
Advantages of a mesh topology

- Can handle high amounts of traffic, because multiple devices can transmit data simultaneously.
- A failure of one device does not cause a break in the network or transmission of data.
- Adding additional devices does not disrupt data transmission between other devices.

Disadvantages of a mesh topology

- The cost to implement is higher than other network topologies, making it a less desirable option.
- Building and maintaining the topology is difficult and time consuming.
- The chance of redundant connections is high, which adds to the high costs and potential for reduced efficiency.

STAR:



A star network, **star topology** is one of the most common network setups. In this configuration, every node connects to a central network device, like a hub, switch, or computer. The central network device acts as a server and the peripheral devices act as clients. Depending on the type of network card used in each computer of the star topology, a coaxial cable or a RJ-45 network cable is used to connect computers together.

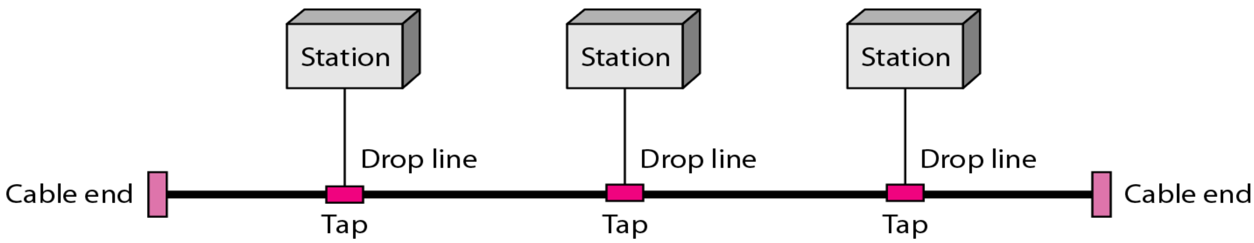
Advantages of star topology

- Centralized management of the network, through the use of the central computer, hub, or switch.
- Easy to add another computer to the network.
- If one computer on the network fails, the rest of the network continues to function normally.
- The star topology is used in local-area networks (LANs), High-speed LANs often use a star topology with a central hub.

Disadvantages of star topology

- Can have a higher cost to implement, especially when using a switch or router as the central network device.
- The central network device determines the performance and number of nodes the network can handle.
- If the central computer, hub, or switch fails, the entire network goes down and all computers are disconnected from the network

BUS:



a **line topology**, a **bus topology** is a network setup in which each computer and network device are connected to a single cable or backbone.

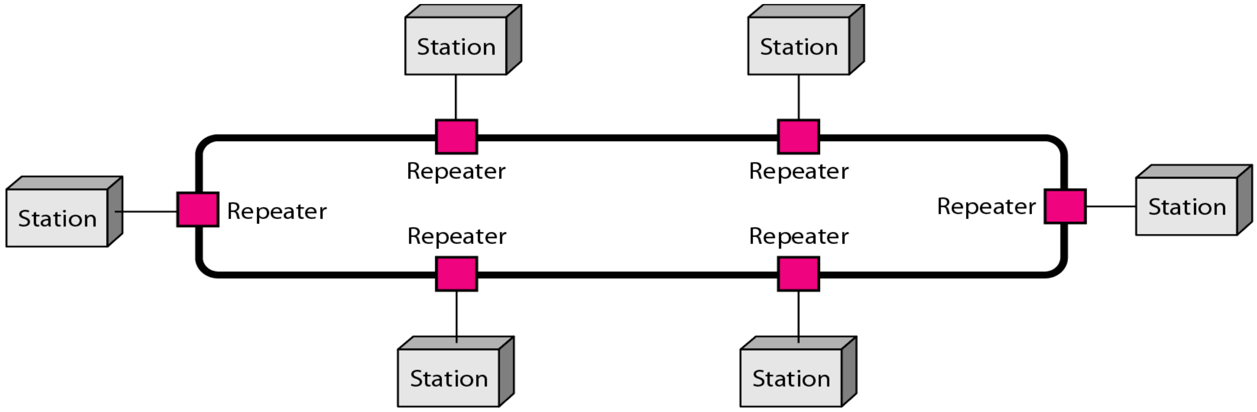
Advantages of bus topology

- It works well when you have a small network.
- It's the easiest network topology for connecting computers or peripherals in a linear fashion.
- It requires less cable length than a star topology.

Disadvantages of bus topology

- It can be difficult to identify the problems if the whole network goes down.
- It can be hard to troubleshoot individual device issues.
- Bus topology is not great for large networks.
- Terminators are required for both ends of the main cable.
- Additional devices slow the network down.
- If a main cable is damaged, the network fails or splits into two.

RING:



A **ring topology** is a network configuration in which device connections create a circular data path. In a ring network, packets of data travel from one device to the next until they reach their destination. Most ring topologies allow packets to travel only in one direction, called a **unidirectional** ring network. Others permit data to move in either direction, called **bidirectional**.
The major disadvantage of a ring topology is that if any individual connection in the ring is broken, the entire network is affected.
Ring topologies may be used in either local area networks (LANs) or wide area networks (WANs).

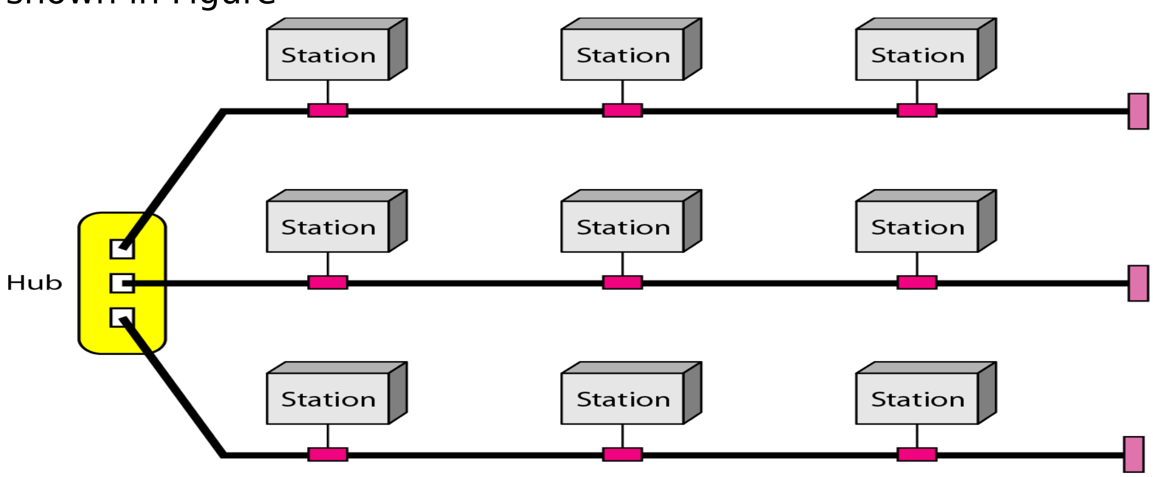
Advantages of ring topology

- All data flows in one direction, reducing the chance of packet collisions.
- A network server is not needed to control network connectivity between each workstation.
- Data can transfer between workstations at high speeds.
- Additional workstations can be added without impacting performance of the network.

Disadvantages of ring topology

- All data being transferred over the network must pass through each workstation on the network, which can make it slower than a star topology.
- The entire network will be impacted if one workstation shuts down.
- The hardware needed to connect each workstation to the network is more expensive than Ethernet cards and hubs/switches.

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



Types of Network based on size

The types of network are classified based upon the size, the area it covers and its physical architecture. The three primary network categories are LAN, WAN and MAN. Each network differs in their characteristics such as distance, transmission speed, cables and cost.
Basic types

LAN (Local Area Network)

Group of interconnected computers within a small area. (room, building, campus)

Two or more pc's can from a LAN to share files, folders, printers, applications and other devices.

Coaxial or CAT 5 cables are normally used for connections.

Due to short distances, errors and noise are minimum.

Data transfer rate is 10 to 100 mbps.

Example: A computer lab in a school.

MAN (Metropolitan Area Network)

Design to extend over a large area.

Connecting number of LAN's to form larger network, so that resources can be shared.

Networks can be up to 5 to 50 km.

Owned by organization or individual.

Data transfer rate is low compare to LAN.

Example: Organization with different branches located in the city.

WAN (Wide Area Network)

Are country and worldwide network.

Contains multiple LAN's and MAN's.

Distinguished in terms of geographical range.

Uses satellites and microwave relays.

Data transfer rate depends upon the ISP provider and varies over the location.

Best example is the internet.

Other types

WLAN (Wireless LAN)

A LAN that uses high frequency radio waves for communication.

Provides short range connectivity with high speed data transmission.

PAN (Personal Area Network)

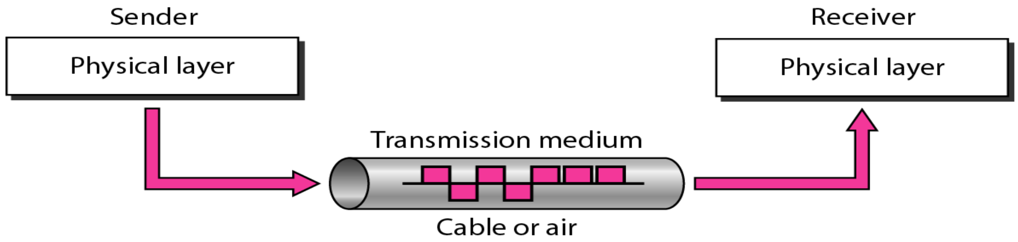
Network organized by the individual user for its personal use.

SAN (Storage Area Network)

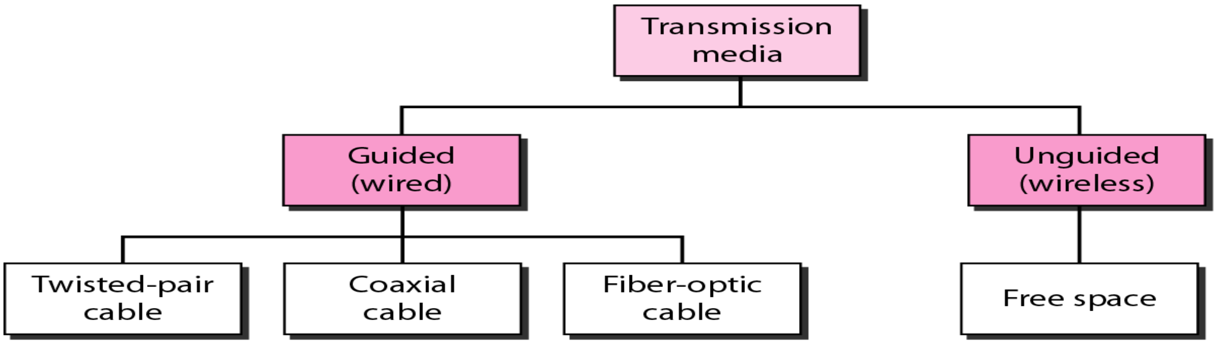
Connects servers to data storage devices via fiber-optic cables.

E.g.: Used for daily backup of organization or a mirror copy

A **transmission medium** can be broadly defined as anything that can carry information from a source to a destination.

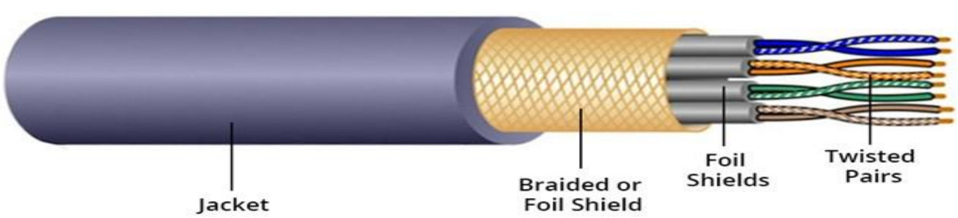


Classes of transmission media



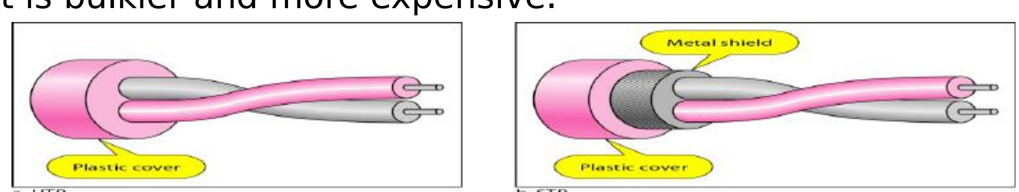
Guided Media: Guided media, which are those that provide a medium from one device to another, include twisted-pair cable, coaxial cable, and fiber-optic cable.

Twisted-Pair Cable: A twisted pair consists of two conductors (normally copper), each with its own plastic insulation, twisted together. One of the wires is used to carry signals to the receiver, and the other is used only as a ground reference.



Unshielded Versus Shielded Twisted-Pair Cable

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



The most common UTP connector is RJ45 (RJ stands for registered jack)

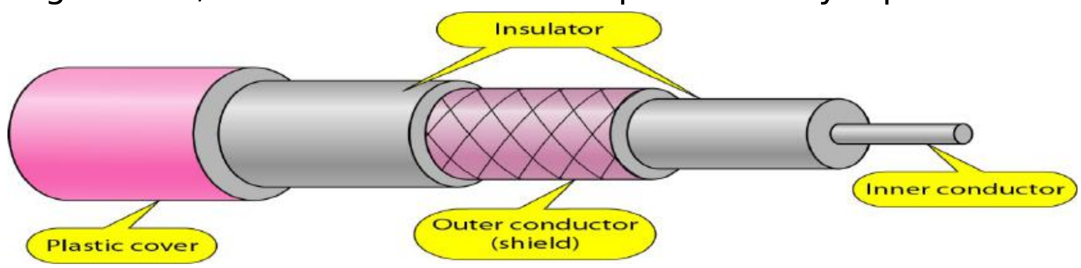
Applications

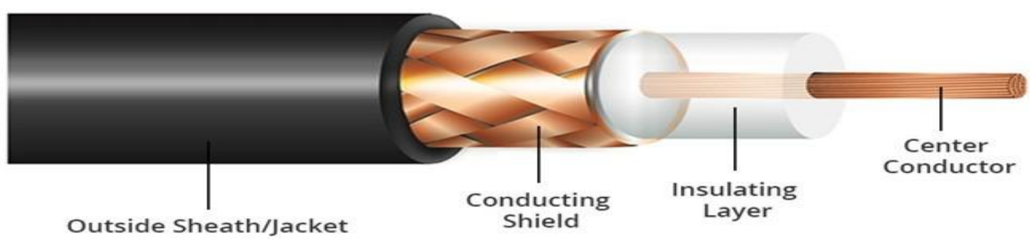
Twisted-pair cables are used in telephone lines to provide voice and data channels.

Local-area networks, such as l0Base-T and l00Base-T, also use twisted-pair cables.

Coaxial Cable

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





The most common type of connector used today is the Bayone-Neill-Concelman (BNe), connector.

Applications

Coaxial cable was widely used in analog telephone networks,digital telephone networks

Cable TV networks also use coaxial cables.

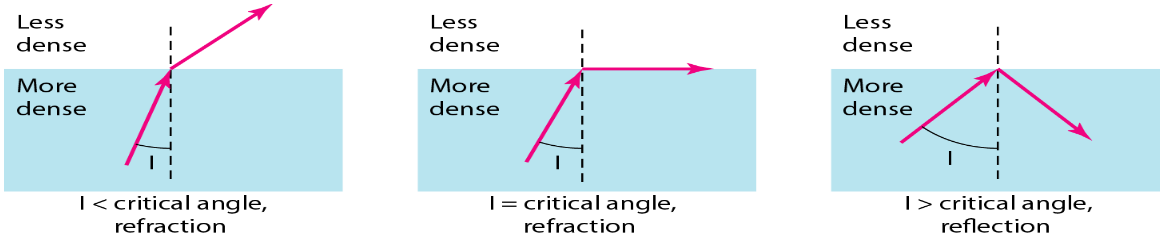
Another common application of coaxial cable is in traditional Ethernet LANs

Fiber-Optic Cable

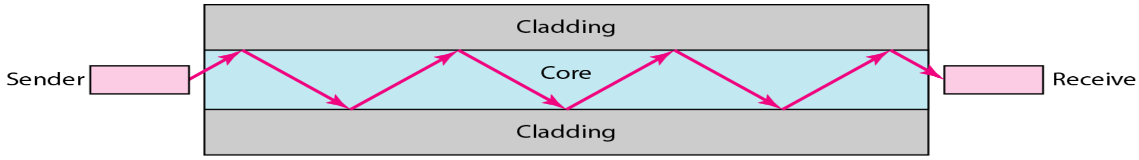
A fiber-optic cable is made of glass or plastic and transmits signals in the form of light. Light travels in a straight line as long as it is moving through a single uniform substance.

If a ray of light traveling through one substance suddenly enters another substance(of a different density), the ray changes direction.

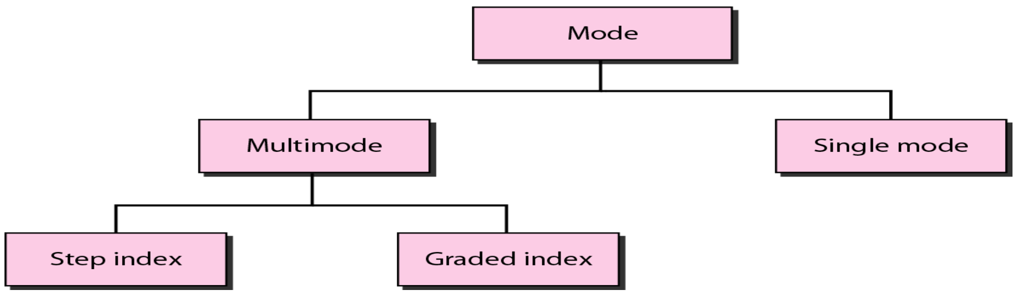
Bending of light ray



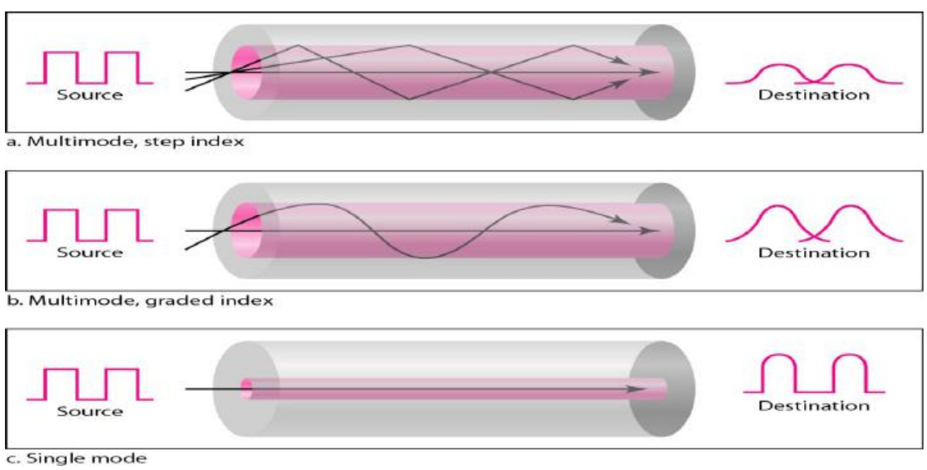
Optical fibers use reflection to guide light through a channel. A glass or plastic core is surrounded by a cladding of less dense glass or plastic.



Propagation Modes



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

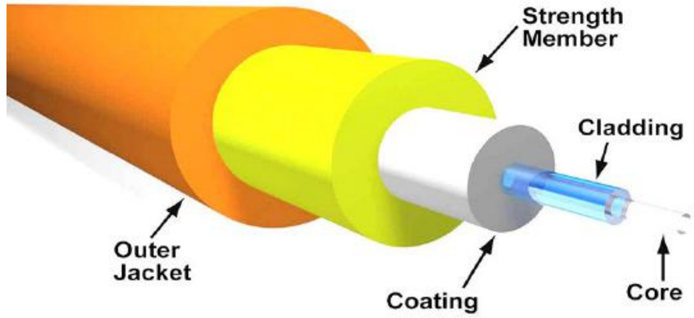


In **multimode step-index fiber**, the density of the core remains constant from the center to the edges. A beam of light moves through this constant density in a straight line until it reaches the interface of the core and the cladding. The term *step index* refers to the suddenness of this change, which contributes to the distortion of the signal as it passes through the fiber.

A second type of fiber, called **multimode graded-index fiber**, decreases this distortion of the signal through the cable. The word *index* here refers to the index of refraction.

Single-Mode: Single-mode uses step-index fiber and a highly focused source of light that limits beams to a small range of angles, all close to the horizontal.

Fiber Construction



The **subscriber channel** (SC) **connector,** The **straight-tip** (ST) **connector, MT-RJ(mechanical transfer registered jack)** is a connector

Applications

Fiber-optic cable is often found in backbone networks because its wide bandwidth is cost-effective..

Some cable TV companies use a combination of optical fiber and coaxial cable,thus creating a hybrid network.

Local-area networks such as 100Base-FX network (Fast Ethernet) and 1000Base-X also use fiber-optic cable

Advantages and Disadvantages of Optical Fiber

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

1 Higher bandwidth.

2 Less signal attenuation. Fiber-optic transmission distance is significantly greaterthan that of other guided media. A signal can run for 50 km without requiring regeneration. We need repeaters every 5 km for coaxial or twisted-pair cable.

3 Immunity to electromagnetic interference. Electromagnetic noise cannot affect fiber-optic cables.

4 Resistance to corrosive materials. Glass is more resistant to corrosive materials than copper.

5 Light weight. Fiber-optic cables are much lighter than copper cables.

6 Greater immunity to tapping. Fiber-optic cables are more immune to tapping than copper cables. Copper cables create antenna effects that can easily be tapped.

Disadvantages There are some disadvantages in the use of optical fiber.

1Installation and maintenance

2 Unidirectional light propagation. Propagation of light is unidirectional. If we need bidirectional communication, two fibers are needed.

3 Cost. The cable and the interfaces are relatively more expensive than those of other guided media. If the demand for bandwidth is not high, often the use of optical fiber cannot be justified.

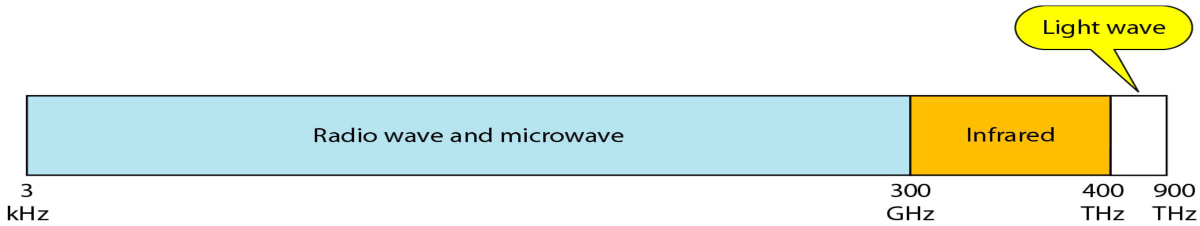
UNGUIDED MEDIA: WIRELESS

Unguided media transport electromagnetic waves without using a physical conductor. This type of communication is often referred to as wireless communication.

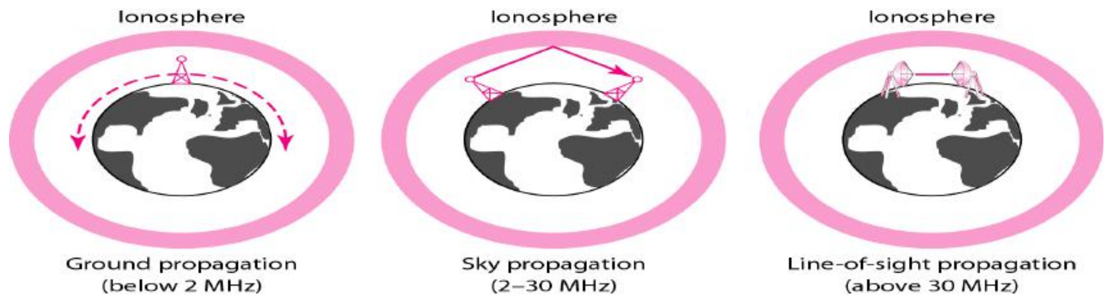
Radio Waves

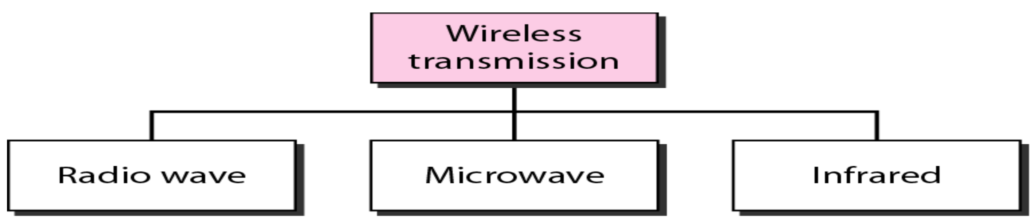
Microwaves

Infrared



Unguided signals can travel from the source to destination in several ways: ground propagation, sky propagation, and line-of-sight propagation, as shown in Figure



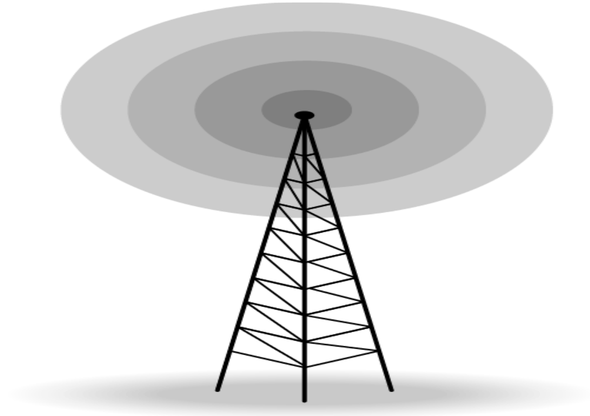


Radio Waves

Electromagnetic waves ranging in frequencies between 3 kHz and 1 GHz are normally called radio waves. Radio waves are omni directional. When an antenna transmits radio waves, they are propagated in all directions. This means that the sending and receiving antennas do not have to be aligned. A sending antenna sends waves that can be received by any receiving antenna. The omni directional property has a disadvantage, too. The radio waves transmitted by one antenna are susceptible to interference by another antenna that may send signals using the same frequency or band.

Omni directional Antenna

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



Applications

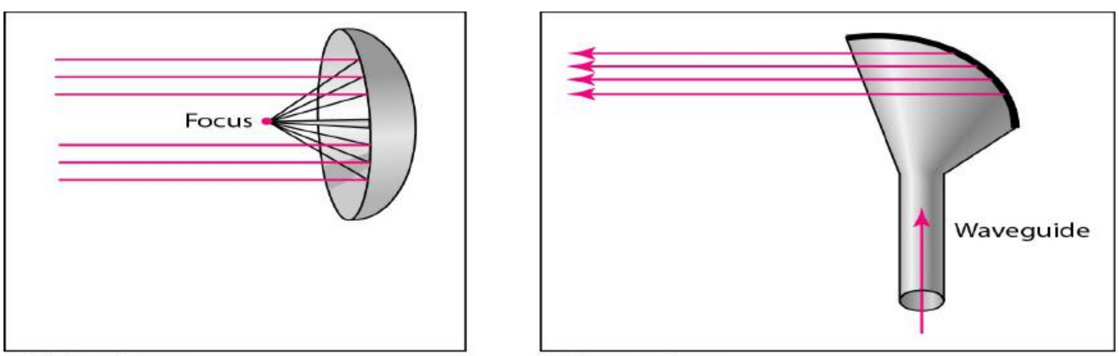
The Omni directional characteristics of radio waves make them useful for multicasting, in which there is one sender but many receivers. AM and FM radio, television, maritime radio, cordless phones, and paging are examples of multicasting.

Microwaves

Electromagnetic waves having frequencies between 1 and 300 GHz are called microwaves. Microwaves are unidirectional. The sending and receiving antennas need to be aligned. The unidirectional property has an obvious advantage. A pair of antennas can be aligned without interfering with another pair of aligned antennas

Unidirectional Antenna

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



a. Dish antenna b. Horn antenna

Applications:

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

Infrared

Infrared waves, with frequencies from 300 GHz to 400 THz (wavelengths from 1 mm to 770 nm), can be used for short-range communication. Infrared waves, having high frequencies, cannot penetrate walls. This advantageous characteristic prevents interference between one system and another; a short-range communication system in one room cannot be affected by another system in the next room.

When we use our infrared remote control, we do not interfere with the use of the remote by our neighbors. Infrared signals useless for long-range communication. In addition, we cannot use infrared waves outside a building because the sun's rays contain infrared waves that can interfere with the communication.

Applications:

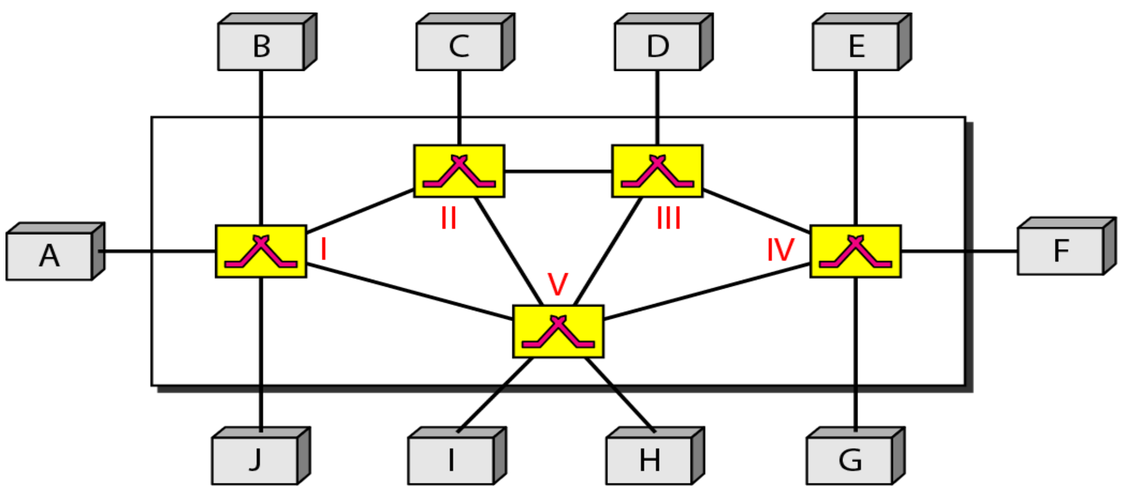
Infrared signals can be used for short-range communication in a closed area using line-of-sight propagation.

Switching

A network is a set of connected devices. Whenever we have multiple devices, we have the problem of how to connect them to make one-to-one communication possible. One solution is to make a point-to-point connection between each pair of devices (a mesh topology) or between a central device and every other device (a star topology). These methods, however, are impractical and wasteful when applied to very large networks.

The number and length of the links require too much infrastructure to be cost-efficient, and the majority of those links would be idle most of the time.

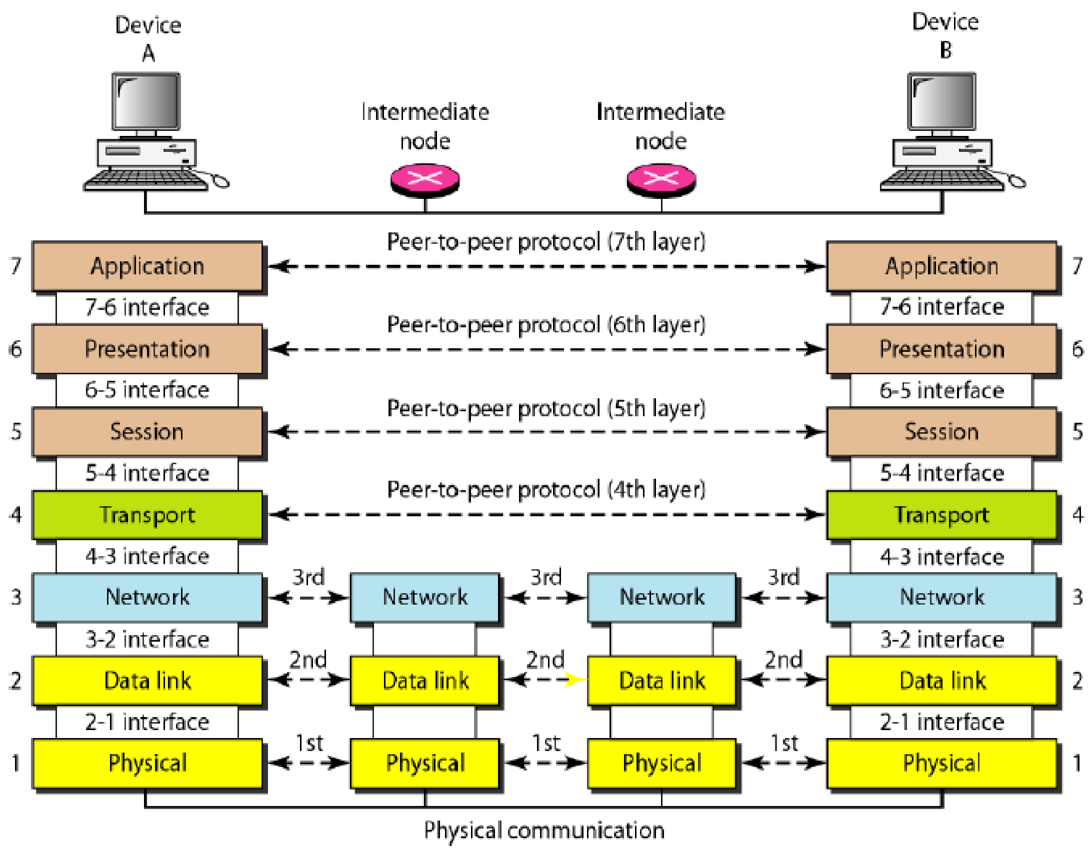
A better solution is switching. A switched network consists of a series of interlinked nodes, called switches. Switches are devices capable of creating temporary connections between two or more devices linked to the switch. In a switched network, some of these nodes are connected to the end systems (computers or telephones, for example). Others are used only for routing. Figure shows a switched network.



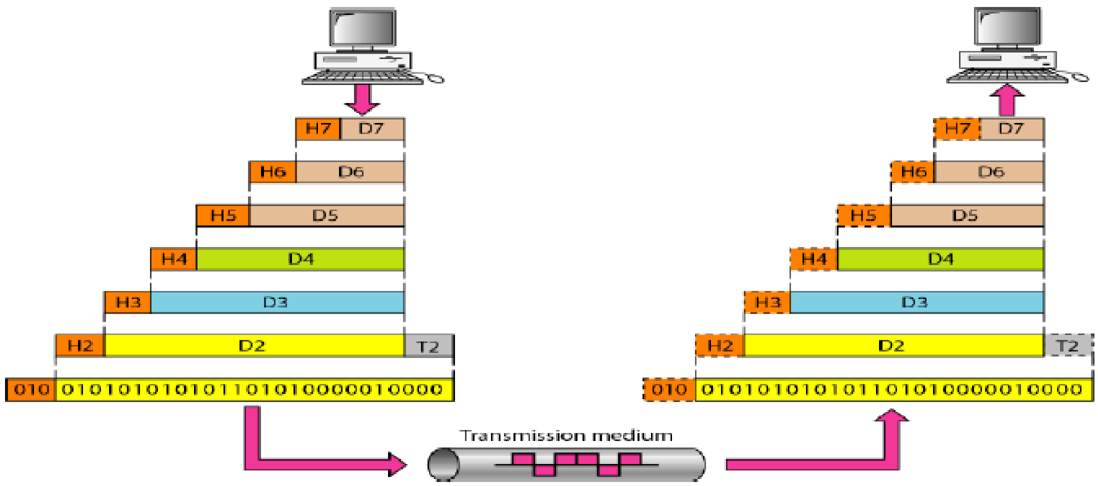
We can then divide today's networks into three broad categories: circuit-switched networks, packet-switched networks, and message-switched. Packet-switched networks can further be divided into two subcategories-virtual-circuit networks and datagram networks as shown in Figure.

OSI Model

- OSI stands for **Open System Interconnection** is a reference model that describes how information from a software application in one computer moves through a physical medium to the software application in another computer.
- OSI consists of seven layers, and each layer performs a particular network function.
- OSI model was developed by the International Organization for Standardization (ISO) in 1984, and it is now considered as an architectural model for the inter-computer communications.
- OSI model divides the whole task into seven smaller and manageable tasks. Each layer is assigned a particular task.



An exchange using the OSI model



Physical layer

- The main functionality of the physical layer is to transmit the individual bits from one node to another node.
- It is the lowest layer of the OSI model.
- It establishes, maintains and deactivates the physical connection.
- It specifies the mechanical, electrical and procedural network interface specifications.

Functions of a Physical layer:

- **Line Configuration:** It defines the way how two or more devices can be connected physically.
- **Data Transmission:** It defines the transmission mode whether it is simplex, half-duplex or full-duplex mode between the two devices on the network.
- **Topology:** It defines the way how network devices are arranged.
- **Signals:** It determines the type of the signal used for transmitting the information.

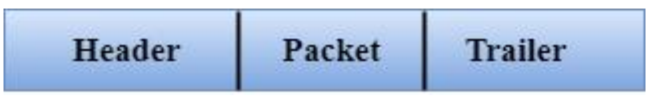
Data-Link Layer

- This layer is responsible for the error-free transfer of data frames.
- It defines the format of the data on the network.
- It provides a reliable and efficient communication between two or more devices.

- It is mainly responsible for the unique identification of each device that resides on a local network.
- It contains two sub-layers:
  - **Logical Link Control Layer**
    - It is responsible for transferring the packets to the Network layer of the receiver that is receiving.
    - It identifies the address of the network layer protocol from the header.
    - It also provides flow control.
  - **Media Access Control Layer**
    - A Media access control layer is a link between the Logical Link Control layer and the network's physical layer.
    - It is used for transferring the packets over the network.

Functions of the Data-link layer

- **Framing:** The data link layer translates the physical's raw bit stream into packets known as Frames. The Data link layer adds the header and trailer to the frame. The header which is added to the frame contains the hardware destination and source address.



- **Physical Addressing:** The Data link layer adds a header to the frame that contains a destination address. The frame is transmitted to the destination address mentioned in the header.
- **Flow Control:** Flow control is the main functionality of the Data-link layer. It is the technique through which the constant data rate is maintained on both the sides so that no data get corrupted. It ensures that the transmitting station such as a server with higher processing speed does not exceed the receiving station, with lower processing speed.
- **Error Control:** Error control is achieved by adding a calculated value CRC (Cyclic Redundancy Check) that is placed to the Data link layer's trailer which is added to the message frame before it is sent to the physical layer. If any error seems to occurr, then the receiver sends the acknowledgment for the retransmission of the corrupted frames.

- **Access Control:** When two or more devices are connected to the same communication channel, then the data link layer protocols are used to determine which device has control over the link at a given time.

Network Layer

- It is a layer 3 that manages device addressing, tracks the location of devices on the network.
- It determines the best path to move data from source to the destination based on the network conditions, the priority of service, and other factors.
- The Data link layer is responsible for routing and forwarding the packets.
- Routers are the layer 3 devices, they are specified in this layer and used to provide the routing services within an internetwork.
- The protocols used to route the network traffic are known as Network layer protocols. Examples of protocols are IP and Ipv6.

Functions of Network Layer:

- **Internetworking:** An internetworking is the main responsibility of the network layer. It provides a logical connection between different devices.
- **Addressing:** A Network layer adds the source and destination address to the header of the frame. Addressing is used to identify the device on the internet.
- **Routing:** Routing is the major component of the network layer, and it determines the best optimal path out of the multiple paths from source to the destination.
- **Packetizing:** A Network Layer receives the packets from the upper layer and converts them into packets. This process is known as Packetizing. It is achieved by internet protocol (IP).

Transport Layer

- The Transport layer is a Layer 4 ensures that messages are transmitted in the order in which they are sent and there is no duplication of data.
- The main responsibility of the transport layer is to transfer the data completely.
- It receives the data from the upper layer and converts them into smaller units known as segments.

- This layer can be termed as an end-to-end layer as it provides a point-to-point connection between source and destination to deliver the data reliably.

The two protocols used in this layer are:

- **Transmission Control Protocol**
    - It is a standard protocol that allows the systems to communicate over the internet.
    - It establishes and maintains a connection between hosts.
    - When data is sent over the TCP connection, then the TCP protocol divides the data into smaller units known as segments. Each segment travels over the internet using multiple routes, and they arrive in different orders at the destination. The transmission control protocol reorders the packets in the correct order at the receiving end.
- **User Datagram Protocol**
    - User Datagram Protocol is a transport layer protocol.
    - It is an unreliable transport protocol as in this case receiver does not send any acknowledgment when the packet is received, the sender does not wait for any acknowledgment. Therefore, this makes a protocol unreliable.

Functions of Transport Layer:

- **Service-point addressing:** Computers run several programs simultaneously due to this reason, the transmission of data from source to the destination not only from one computer to another computer but also from one process to another process. The transport layer adds the header that contains the address known as a service-point address or port address. The responsibility of the network layer is to transmit the data from one computer to another computer and the responsibility of the transport layer is to transmit the message to the correct process.
- **Segmentation and reassembly:** When the transport layer receives the message from the upper layer, it divides the message into multiple segments, and each segment is assigned with a sequence number that uniquely identifies each segment. When the message has arrived at the destination, then the transport layer reassembles the message based on their sequence numbers.

- **Connection control:** Transport layer provides two services Connection-oriented service and connectionless service. A connectionless service treats each segment as an individual packet, and they all travel in different routes to reach the destination. A connection-oriented service makes a connection with the transport layer at the destination machine before delivering the packets. In connection-oriented service, all the packets travel in the single route.
- **Flow control:** The transport layer also responsible for flow control but it is performed end-to-end rather than across a single link.
- **Error control:** The transport layer is also responsible for Error control. Error control is performed end-to-end rather than across the single link. The sender transport layer ensures that message reach at the destination without any error.

Session Layer

- It is a layer 3 in the OSI model.
- The Session layer is used to establish, maintain and synchronizes the interaction between communicating devices.

Functions of Session layer:

- **Dialog control:** Session layer acts as a dialog controller that creates a dialog between two processes or we can say that it allows the communication between two processes which can be either half-duplex or full-duplex.
- **Synchronization:** Session layer adds some checkpoints when transmitting the data in a sequence. If some error occurs in the middle of the transmission of data, then the transmission will take place again from the checkpoint. This process is known as Synchronization and recovery.

Presentation Layer

- A Presentation layer is mainly concerned with the syntax and semantics of the information exchanged between the two systems.
- It acts as a data translator for a network.
- This layer is a part of the operating system that converts the data from one presentation format to another format.

- The Presentation layer is also known as the syntax layer.

Functions of Presentation layer:

- **Translation:** The processes in two systems exchange the information in the form of character strings, numbers and so on. Different computers use different encoding methods, the presentation layer handles the interoperability between the different encoding methods. It converts the data from sender-dependent format into a common format and changes the common format into receiver-dependent format at the receiving end.
- **Encryption:** Encryption is needed to maintain privacy. Encryption is a process of converting the sender-transmitted information into another form and sends the resulting message over the network.
- **Compression:** Data compression is a process of compressing the data, i.e., it reduces the number of bits to be transmitted. Data compression is very important in multimedia such as text, audio, video.

Application Layer

- An application layer serves as a window for users and application processes to access network service.
- It handles issues such as network transparency, resource allocation, etc.
- An application layer is not an application, but it performs the application layer functions.
- This layer provides the network services to the end-users.

Functions of Application layer:

- **File transfer, access, and management (FTAM):** An application layer allows a user to access the files in a remote computer, to retrieve the files from a computer and to manage the files in a remote computer.
- **Mail services:** An application layer provides the facility for email forwarding and storage.
- Directory services: An application provides the distributed database sources and is used to provide that global information about various objects.

TCP/IP Model

The **OSI Model** we just looked at is just a reference/logical model. It was designed to describe the functions of the communication system by dividing the communication procedure into smaller and simpler components. But when we talk about the TCP/IP model, it was designed and developed by Department of Defense (DoD) in 1960s and is based on standard protocols. It stands for Transmission Control Protocol/Internet Protocol. The **TCP/IP model** is a concise version of the OSI model. It contains four layers, unlike seven layers in the OSI model. The layers are:

1. Process/Application Layer
2. Host-to-Host/Transport Layer
3. Internet Layer
4. Network Access/Link Layer

The diagrammatic comparison of the TCP/IP and OSI model is as follows :

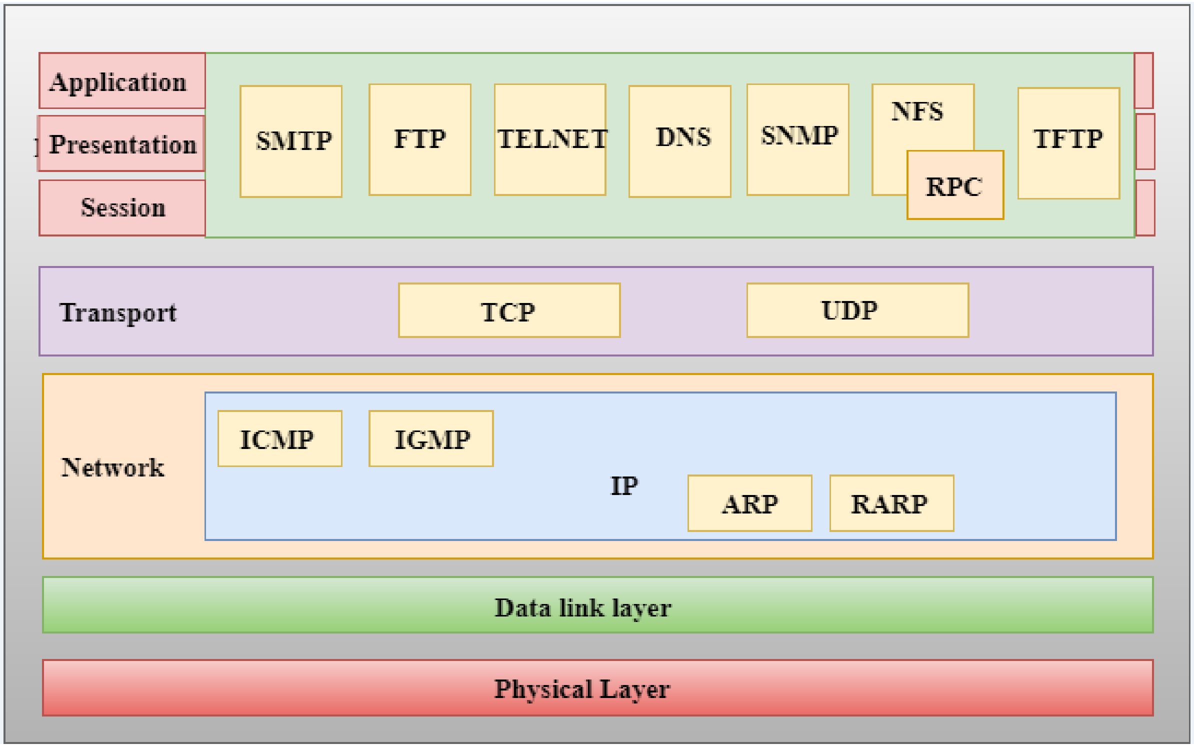
TCP/IP MODEL
Application Layer
Transport Layer
Internet Layer
Network Access Layer

OSI MODEL
Application Layer
Presentation Layer
Session Layer
Transport Layer
Network Layer
Data Link Layer
Physical Layer

1. Network Access Layer –

This layer corresponds to the combination of Data Link Layer and Physical Layer of the OSI model. It looks out for hardware addressing and the protocols present in this layer allows for the physical transmission of data.
We just talked about ARP being a protocol of Internet layer, but there is a conflict about declaring it as a protocol of Internet Layer or Network access layer. It is described as residing in layer 3, being encapsulated by layer 2 protocols.

Functions of TCP/IP layers:



2. Internet Layer –

This layer parallels the functions of OSI's Network layer. It defines the protocols which are responsible for logical transmission of data over the entire network. The main protocols residing at this layer are :

1. **IP –** stands for Internet Protocol and it is responsible for delivering packets from the source host to the destination host by looking at the IP addresses in the packet headers. IP has 2 versions: IPv4 and IPv6. IPv4 is the one that most of the websites are using currently. But IPv6 is growing as the number of IPv4 addresses are limited in number when compared to the number of users.
2. **ICMP –** stands for Internet Control Message Protocol. It is encapsulated within IP datagrams and is responsible for providing hosts with information about network problems.
3. **ARP –** stands for Address Resolution Protocol. Its job is to find the hardware address of a host from a known IP address. ARP has several types: Reverse ARP, Proxy ARP, Gratuitous ARP and Inverse ARP.

3. Host-to-Host Layer –

This layer is analogous to the transport layer of the OSI model. It is responsible for end-to-end communication and error-free delivery of data. It shields the upper-layer applications from the complexities of data. The two main protocols present in this layer are :

1. **Transmission Control Protocol (TCP) –** It is known to provide reliable and error-free communication between end systems. It performs sequencing and segmentation of data. It also has acknowledgment feature and controls the flow of the data through flow control mechanism. It is a very effective protocol but has a lot of overhead due to such features. Increased overhead leads to increased cost.
2. **User Datagram Protocol (UDP) –** On the other hand does not provide any such features. It is the go-to protocol if your application does not require reliable transport as it is very cost-effective. Unlike TCP, which is connection-oriented protocol, UDP is connectionless.

4. Application Layer –

This layer performs the functions of top three layers of the OSI model: Application, Presentation and Session Layer. It is responsible for node-to-node communication and controls user-interface specifications. Some of the protocols present in this layer are: HTTP, HTTPS, FTP, TFTP, Telnet, SSH, SMTP, SNMP, NTP, DNS, DHCP, NFS, X Window, LPD. Have a look at Protocols in Application Layer for some information about these protocols. Protocols other than those present in the linked article are :

1. **HTTP and HTTPS –** HTTP stands for Hypertext transfer protocol. It is used by the World Wide Web to manage communications between web browsers and servers. HTTPS stands for HTTP-Secure. It is a combination of HTTP with SSL(Secure Socket Layer). It is efficient in cases where the browser need to fill out forms, sign in, authenticate and carry out bank transactions.
2. **SSH –** SSH stands for Secure Shell. It is a terminal emulations software similar to Telnet. The reason SSH is more preferred is because of its ability to maintain the encrypted connection. It sets up a secure session over a TCP/IP connection.
3. **NTP –** NTP stands for Network Time Protocol. It is used to synchronize the clocks on our computer to one standard time source. It is very useful in situations like bank transactions. Assume the following situation without the presence of NTP. Suppose you carry out a transaction, where your computer reads the time at 2:30 PM while the server records it at 2:28 PM. The server can crash very badly if it’s out of sync.

Differences between OSI and TCP/IP Reference Model –

OSI	TCP/IP
OSI represents **Open System Interconnection**.	TCP/IP model represents the Transmission Control Protocol / Internet Protocol.
OSI is a generic, protocol independent standard. It is acting as an interaction gateway between the network and the final-user.	TCP/IP model depends on standard protocols about which the computer network has created. It is a connection protocol that assigns the network of hosts over the internet.
The OSI model was developed first, and then protocols were created to fit the network architecture's needs.	The protocols were created first and then built the TCP/IP model.
It provides quality services.	It does not provide quality services.
The OSI model represents defines administration, interfaces and conventions. It describes clearly which layer provides services.	It does not mention the services, interfaces, and protocols.
The protocols of the OSI model are better unseen and can be returned with another appropriate protocol quickly.	The TCP/IP model protocols are not hidden, and we cannot fit a new protocol stack in it.
It is difficult as distinguished to TCP/IP.	It is simpler than OSI.
It provides both connection and connectionless oriented transmission in the network layer; however, only connection-oriented transmission in the transport layer.	It provides connectionless transmission in the network layer and supports connecting and connectionless-oriented transmission in the transport layer.
It uses a horizontal approach.	It uses a vertical approach.
The smallest size of the OSI header is 5 bytes.	The smallest size of the TCP/IP header is 20 bytes.
Protocols are unknown in the OSI model and are returned while the technology modifies.	In TCP/IP, returning protocol is not difficult.

OSI MODEL	TCP/IP MODEL
Contains 7 Layers	Contains 4 Layers
Uses Strict Layering resulting in vertical layers.	Uses Loose Layering resulting in horizontal layers.
Supports both connectionless & connection-oriented communication in the Network layer, but only connection-oriented communication in Transport Layer	Supports only connectionless communication in the Network layer, but both connectionless & connection-oriented communication in Transport Layer
It distinguishes between Service, Interface and Protocol.	Does not clearly distinguish between Service, Interface and Protocol.
Protocols are better hidden and can be replaced relatively easily as technology changes (No transparency)	Protocols are not hidden and thus cannot be replaced easily. (Transparency) Replacing IP by a substantially different protocol would be virtually impossible
OSI reference model was devised before the corresponding protocols were designed.	The protocols came first and the model was a description of the existing protocols

THE INTERNET

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

A Brief History

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

In the mid-1960s, mainframe computers in research organizations were standalone devices. Computers from different manufacturers were unable to communicate with one another. The Advanced Research Projects Agency

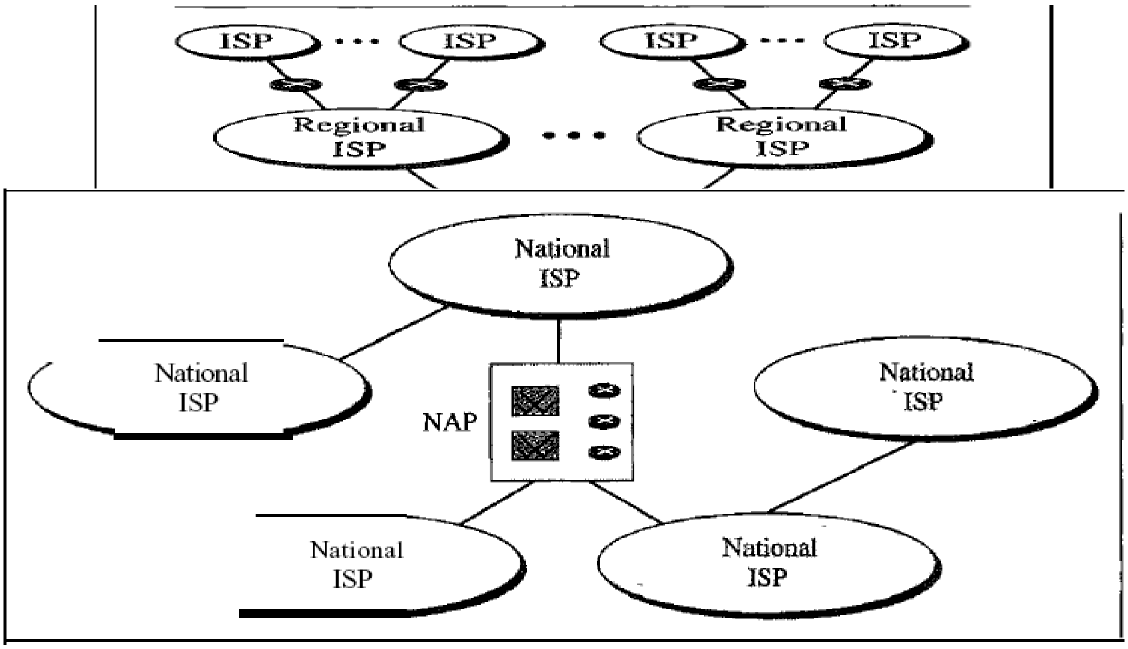
(ARPA) in the Department of Defense (DoD) was interested in finding a way to connect computers so that the researchers they funded could share their findings, thereby reducing costs and eliminating duplication of effort.

In 1967, at an Association for Computing Machinery (ACM) meeting, ARPA presented its ideas for ARPANET, a small network of connected computers. The idea was that each host computer (not necessarily from the same manufacturer) would be attached to a specialized computer, called an *inteiface message processor* (IMP). The IMPs, in tum, would be connected to one another. Each IMP had to be able to communicate with other IMPs as well as with its own attached host. By 1969, ARPANET was a reality. Four nodes, at the University of California at Los Angeles (UCLA), the University of California at Santa Barbara (UCSB), Stanford Research Institute (SRI), and the University of Utah, were connected via the IMPs to form a network. Software called the *Network Control Protocol* (NCP) provided communication between the hosts.

In 1972, Vint Cerf and Bob Kahn, both of whom were part of the core ARPANET group, collaborated on what they called the *Internetting Projec1.* Cerf and Kahn's landmark 1973 paper outlined the protocols to achieve end-to-end delivery of packets. This paper on Transmission Control Protocol (TCP) included concepts such as encapsulation, the datagram, and the functions of a gateway. Shortly thereafter, authorities made a decision to split TCP into two protocols: Transmission Control Protocol (TCP) and Internetworking Protocol (lP). IP would handle datagram routing while TCP would be responsible for higher-level functions such as segmentation, reassembly, and error detection. The internetworking protocol became known as TCPIIP.

The Internet Today

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



b. Interconnection of national ISPs

International Internet Service Providers:

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

National Internet Service Providers:

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

Regional Internet Service Providers:

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

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

Data Link Layer in OSI Model

The Data Link Layer is the 2^{nd} layer from the bottom to the top of the OSI model. Its job is to provide node-to-node delivery of data. The primary role of the data link layer is to check whether the data transmitted from one point to another node point on the physical layer is error-free or not. If any error occurs during data transmission, the data link layer will discard that data and resend the data. This layer is responsible for reliable and efficient communication between devices.

<u>Functions of the Data-Link Layer</u>

The working of the data-link layer in the OSI model requires the need functioning of the following attributes:

1. Framing

- The data packets received from the network layer are encapsulated in frames by the data-link layer for bit-to-bit sharing over the channel.
- It is also responsible for restructuring the framed data in the network model, and each data frame is different from the others.

2. Addressing

- The task of adding a physical address to the frame in the header format is known as addressing.
- It acts as an identification service for transmitting the frames to multiple network models over the channel.

3. Flow Control

- During data transmission, the sender or receiver's data flow may differ, causing network traffic in the channel.
- The Data-link layer in such situations acts as a flow control for the sender side to prevent data overflow at the receiver side.

4. Access Control

- In this network model, when multiple devices share the same communication channel, this leads to data collision in the network channel.
- To prevent such data collision, the data-link layer performs checks on the devices with the same network channel to avoid data loss.

5. Error Control

- During data transmission, due to noise or signal loss, errors might occur in the data being transmitted.
- To minimize such data error rate, the data-link layer performs error detection and correction techniques on the transmitted data.

Sub-Layers of the Data-Link Layer

The data-link layer depending on the communication with the other OSI model is divided into two types of sub-layers:

1. Logical Link Control (LLC)

This is the upper sub-layer of the data-link layer.

- The LLC sub-layer is responsible for handling and maintaining the communication between the other layers of the OSI Model.
- It is also responsible for handling error messages and reliability checks for the data.

2. Media Access Control (MAC)

This is the lower sub-layer of the data-link layer.

- The MAC sub-layer is responsible for framing the data received from the upper layers.
- It also is responsible for data encapsulation and media access control for the data received.

Design Issues in Data Link Layer

1. Services provided to the network layer –
The data link layer act as a service interface to the network layer. The principle service is transferring data from network layer on sending machine to the network layer on destination machine. This transfer also takes place via DLL (Data link-layer).
It provides three types of services:

1. Unacknowlwdged and connectionless services.
2. Acknowledged and connectionless services.
3. Acknowledged and connection-oriented services

Unacknowledged and connectionless services.

- Here the sender machine sends the independent frames without any acknowledgement from the sender.
- There is no logical connection established.

Acknowledged and connectionless services.

- There is no logical connection between sender and receiver established.
- Each frame is acknowledged by the receiver.
- If the frame didn't reach the receiver in a specific time interval it has to be sent again.
- It is very useful in wireless systems.

Acknowledged and connection-oriented services

- A logical connection is established between sender and receiver before data is trimester.
- Each frame is numbered so the receiver can ensure all frames have arrived and exactly once.

2. Frame synchronization –

The source machine sends data in the form of blocks called frames to the destination machine. The starting and ending of each frame should be identified so that the frame can be recognized by the destination machine.

Framing and Framing Protocols

The data link layer, on the other hand, needs to pack bits into frames, so that each frame is distinguishable from another. Framing in the data link layer separates a message from one source to a destination, or from other messages to other destinations, by adding a sender address and a destination address. The destination address defines where the packet is to go; the sender address helps the recipient acknowledge the receipt.

Fixed size framing:

In fixed-size framing, there is no need for defining the boundaries of the frames; the size itself can be used as a delimiter. An example of this type of framing is the ATM wide-area network, which uses frames of fixed size called cells.

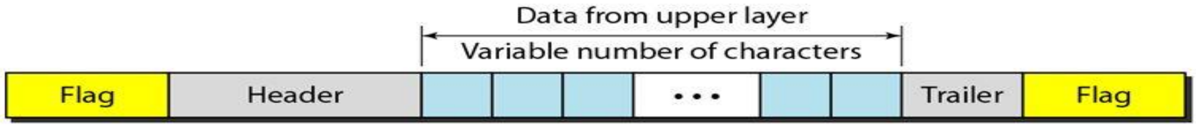
Variable-Size Framing:

In variable-size framing, we need to define the end of the frame and the beginning of the next. There are two approaches which are used for this purpose: *A character-oriented approach* and *A bit-oriented approach*.

Character-Oriented Protocols:

In a character-oriented protocol, data to be carried are 8-bit characters from a coding system such as ASCII (see Appendix A). The header, which normally carries the source and destination addresses and other control information, and the trailer, which carries error detection or error correction redundant bits, are also multiples of 8 bits.

To separate one frame from the next, an 8- bit (I-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. The following figure shows the format of a frame in a character-oriented protocol.

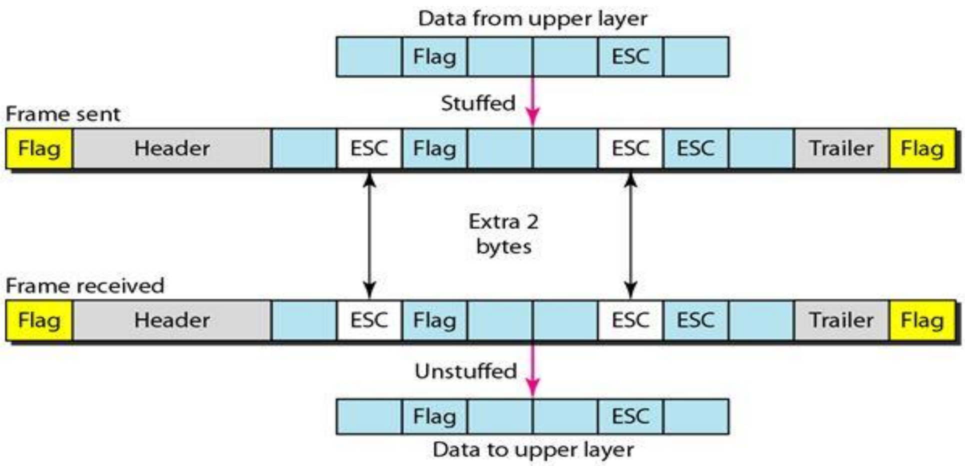


Character-oriented framing was popular when only text was exchanged by the data link layers. The flag could be selected to be any character not used for text communication.

However, if we send other types of information such as graphs, audio, and video. Any pattern used for the flag could also be part of the information. If this happens, the receiver, when it encounters this pattern in the middle of the data, thinks it has reached the end of the frame.

To fix this problem, a byte-stuffing strategy was added to character-oriented framing. In byte stuffing (or character stuffing), a special byte is added to the data section of the frame when there is a character with the same pattern as the flag. The data section is stuffed with an extra byte. This byte is usually called the escape character (ESC), which has a predefined bit pattern. Whenever the receiver encounters the ESC character, it removes it from the data section and treats the next character as data, not a delimiting flag.

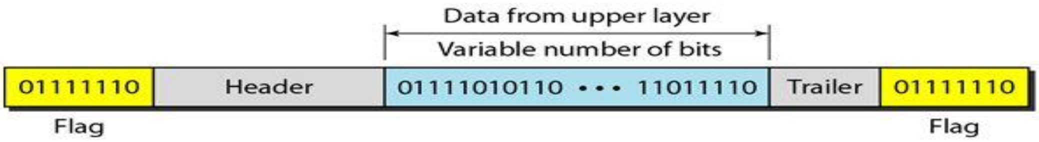
Byte stuffing by the escape character allows the presence of the flag in the data section of the frame, but it creates another problem. What happens if the text contains one or more escape characters followed by a flag? The receiver removes the escape character, but keeps the flag, which is incorrectly interpreted as the end of the frame. To solve this problem, the escape characters that are part of the text must also be marked by another escape character. In other words, if the escape character is part of the text, an extra one is added to show that the second one is part of the text. The following figure shows the situation.



Character-oriented protocols present another problem in data communications. The universal coding systems in use today, such as Unicode, have 16-bit and 32-bit characters that conflict with 8-bit characters. We can say that in general, the tendency is moving toward the bit-oriented protocols.

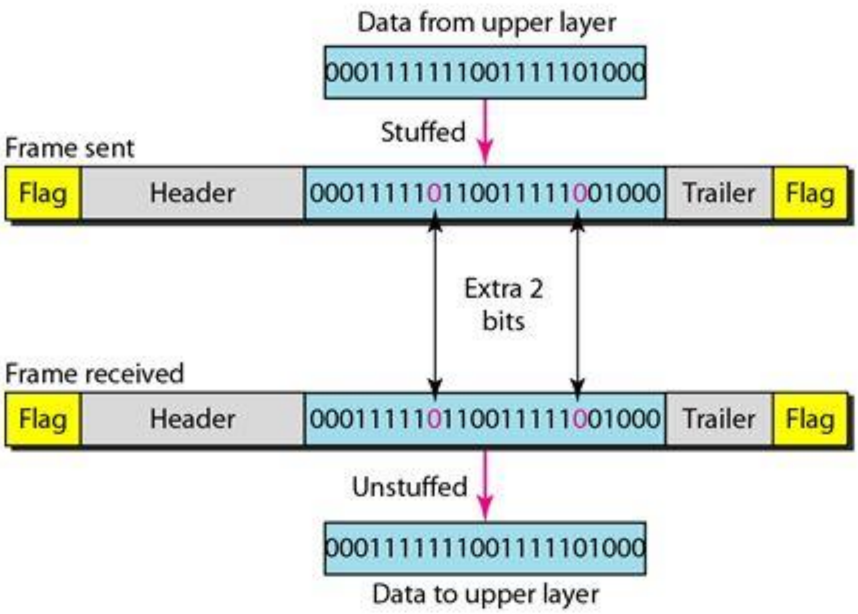
Bit-Oriented Protocols:

In a bit-oriented protocol, the data section of a frame is a sequence of bits to be interpreted by the upper layer as text, graphic, audio, video, and so on. However, in addition to headers (and possible trailers), we need a delimiter to separate one frame from the other. 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 the following figure.



This flag can create the same type of problem we saw in the byte-oriented protocols. 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 1 byte) to prevent the pattern from looking like a flag. The strategy is called bit stuffing. In bit stuffing, if a 0 and five consecutive 1 bits are encountered, an extra 0 is added. This extra stuffed bit is eventually removed from the data by the receiver. Note that the extra bit is added after one 0 followed by five 1s regardless of the value of the next bit. This guarantees that the flag field sequence does not inadvertently appear in the frame.

The following figure shows bit stuffing at the sender and bit removal at the receiver. Note that even if we have a 0 after five 1s, we still stuff a 0. The 0 will be removed by the receiver.



This means that if the flag like pattern 01111110 appears in the data, it will change to 011111010 (stuffed) and is not mistaken as a flag by the receiver. The real flag 01111110 is not stuffed by the sender and is recognized by the receiver.

Flow Control

Flow control coordinates the amount of data that can be sent before receiving an acknowledgment and is one of the most important duties of the data link layer. In most protocols, flow control is a set of procedures that tells the sender how much data it can transmit before it must wait for an acknowledgment from the receiver.

Any receiving device has a limited speed at which it can process incoming data and a limited amount of memory in which to store incoming data. The receiving device must be able to inform the sending device before those limits are reached and to request that the transmitting device send fewer frames or stop temporarily. Incoming data must be checked and processed before they can be used. The rate of such processing is often slower than the rate of transmission. For this reason, each receiving device has a block of memory, called a buffer, reserved for storing incoming data until they are processed. If the buffer begins to fill up, the receiver must be able to tell the sender to halt transmission until it is once again able to receive.

Error Control:

Error control is both error detection and error correction. It allows the receiver to inform the sender of any frames lost or damaged in transmission and coordinates the retransmission of those frames by the sender. In the data link layer, the term error control refers primarily to methods of error detection and retransmission. Error control in the data link layer is often implemented simply: Any time an error is detected in an exchange, specified frames are retransmitted. This process is called automatic repeat request (ARQ).

Types of Errors:

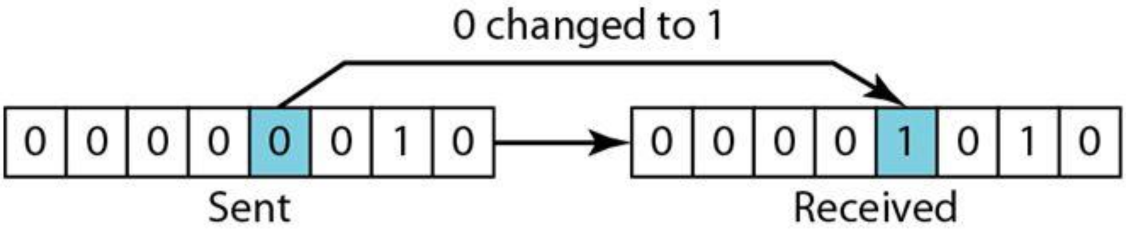
Networks must be able to transfer data from one device to another with acceptable accuracy. Any time data are transmitted from one node to the next, they can become corrupted in passage. Many factors can alter one or more bits of a message. Some applications require a mechanism for detecting and correcting errors.

Whenever bits flow from one point to another, they are subject to unpredictable changes because of interference. This interference can change the shape of the signal. In a single-bit error, a 0 is changed to a 1 or a 1 to a 0. In a burst error, multiple bits are changed.

Single-Bit Error

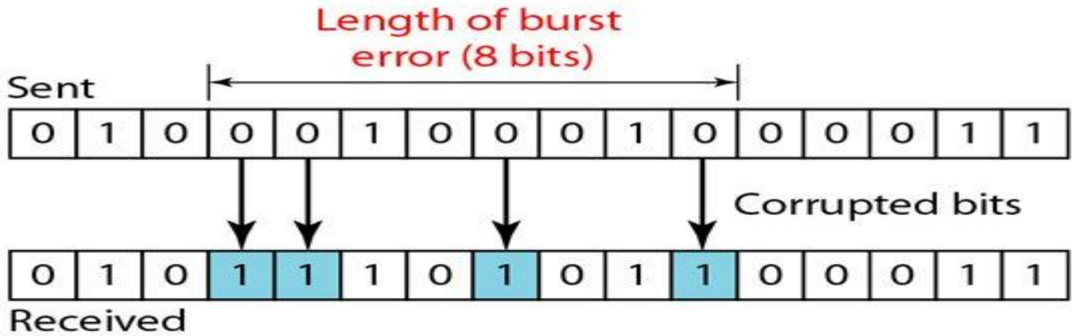
The term single-bit error means that only 1 bit of a given data unit (such as a byte, character, or packet) is changed from 1 to 0 or from 0 to 1.

The following figure shows the effect of a single-bit error on a data unit. To understand the impact of the change, imagine that each group of 8 bits is an ASCII character with a 0 bit added to the left. In the figure 00000010 (ASCII STX) was sent, meaning start of text, but 00001010 (ASCII LF) was received, meaning line feed.



Burst Error:

The term burst error means that 2 or more bits in the data unit have changed from 1 to 0 or from 0 to 1.
The following figure shows the effect of a burst error on a data unit. In this case, 0100010001000011 was sent, but 0101110101100011 was received. Note that a burst error does not necessarily mean that the errors occur in consecutive bits. The length of the burst is measured from the first corrupted bit to the last corrupted bit. Some bits in between may not have been corrupted.



Error Detection and Correction

The central concept in detecting or correcting errors is redundancy. To be able to detect or correct errors, we need to send some extra bits with our data. These redundant bits are added by the sender and removed by the receiver. Their presence allows the receiver to detect or correct corrupted bits.

Detection versus Correction

The correction of errors is more difficult than the detection. In error detection, we are looking only to see if any error has occurred. The answer is a simple yes or no.

In error correction, we need to know the exact number of bits that are corrupted and more importantly, their location in the message. The number of the errors and the size of the message are important factors. If we need to correct one single error in an 8-bit data unit, we need to consider eight possible error locations; if we need to correct two errors in a data unit of the same size, we need to consider 28 possibilities.

Coding

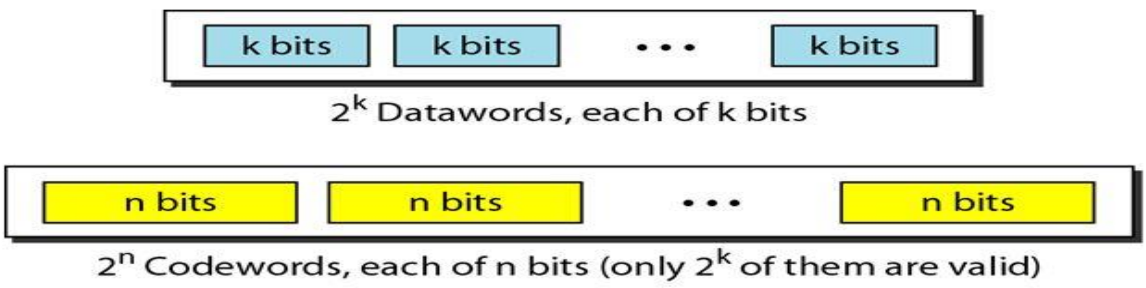
Redundancy is achieved through various coding schemes. The sender adds redundant bits through a process that creates a relationship between the redundant bits and the actual data bits. The receiver checks the relationships between the two sets of bits to detect or correct the errors. The ratio of redundant bits to the data bits and the robustness of the process are important factors in any coding scheme. The following figure shows the general idea of coding.

Block Coding Techniques in Error Detection and Correction

In block coding, we divide our message into blocks, each of k bits, called data words. We add r redundant bits to each block to make the length n = k + r. The resulting n-bit blocks are called code words.

For example, we have a set of data words, each of size k, and a set of code words, each of size of n. With k bits, we can create a combination of 2k data words, with n bits; we can create a combination of 2n code words. Since n > k, the number of possible code words is larger than the number of possible data words.

The block coding process is one-to-one; the same data word is always encoded as the same code word. This means that we have 2n-2k code words that are not used. We call these code words invalid or illegal. The following figure shows the situation.



Error Detection

If the following two conditions are met, the receiver can detect a change in the original code word by using Block coding technique.

1. The receiver has (or can find) a list of valid code words.
2. The original code word has changed to an invalid one.

The sender creates code words out of data words by using a generator that applies the rules and procedures of encoding (discussed later). Each code word sent to the receiver may change during transmission. If the received code word is the same as one of the valid code words, the word is accepted; the corresponding data word is extracted for use.

If the received code word is not valid, it is discarded. However, if the code word is corrupted during transmission but the received word still matches a valid code word, the error remains undetected. This type of coding can detect only single errors. Two or more errors may remain undetected.

For example consider the following table of data words and Code words:

Datawords	Codewords
00	000
01	011
10	101
11	110

Assume the sender encodes the data word 01 as 011 and sends it to the receiver. Consider the following cases:

1. The receiver receives O11. It is a valid code word. The receiver extracts the data word 01 from it.
2. The code word is corrupted during transmission, and 111 is received (the leftmost bit is corrupted). This is not a valid code word and is discarded.
3. The code word is corrupted during transmission, and 000 is received (the right two bits are corrupted). This is a valid code word. The receiver incorrectly extracts the data word 00. Two corrupted bits have made the error undetectable.

Error Correction:

Error correction is much more difficult than error detection. In error detection, the receiver needs to know only that the received code word is invalid, in error correction the receiver needs to find (or guess) the original code word sent. So, we need more redundant bits for error correction than for error detection.
Consider the following table of Data words and Code words.

Dataword	Codeword
00	00000
01	01011
10	10101
11	11110

Assume the data word is 01. The sender consults the table (or uses an algorithm) to create the code word 01011. The code word is corrupted during transmission, and 01001 is received (error in the second bit from the right). First, the receiver finds that the received code word is not in the table. This means an error has occurred. (Detection must come before correction.) The receiver, assuming that there is only 1 bit corrupted, uses the following strategy to guess the correct data word.

1. Comparing the received code word with the first code word in the table (01001 versus 00000), the receiver decides that the first code word is not the one that was sent because there are two different bits.
2. By the same reasoning, the original code word cannot be the third or fourth one in the table.
3. The original code word must be the second one in the table because this is the only one that differs from the received code word by 1 bit. The receiver replaces 01001 with 01011 and consults the table to find the data word 01.

Simple Parity Check Code

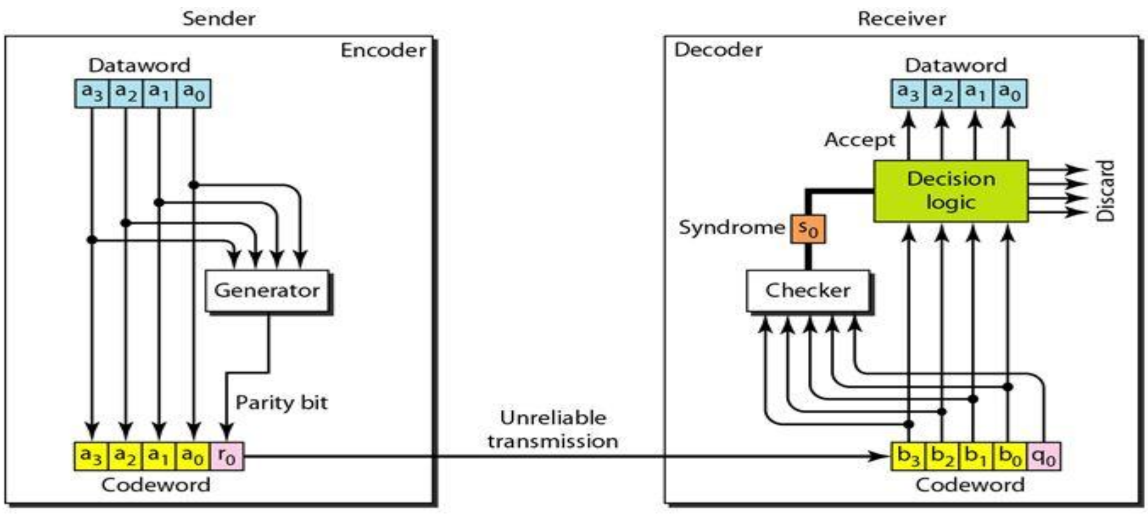
Linear Block Codes:

A linear block code is a code in which the exclusive OR (addition modulo-2) of two valid code words creates another valid code word.

The scheme in the above table is a linear block code because the result of XORing any code word with any other code word is a valid code word. For example, the XORing of the second and third code words creates the fourth one.

Simple Parity-Check Code:

The simple parity-check code is the most familiar error-detecting code. In this code, a k-bit data word is changed to an n-bit code word where n = k + 1. The extra bit, called the parity bit, is selected to make the total number of 1s in the code word even. Although some implementations specify an odd number of 1s. The minimum Hamming distance for this category is dmin =2, which means that the code is a single-bit error-detecting code and it cannot correct any error.
The following figure shows possible structure of an encoder (at the sender) and a decoder (at the receiver).



The encoder uses a generator that takes a copy of a 4-bit data word (a0, a1, a2 and a3) and generates a parity bit r0. The data word bits and the parity bit create the 5-bit code word. The parity bit that is added makes the number of 1s in the code word even.

This is normally done by adding the 4 bits of the data word (modulo-2).

r0=a3+a2+a1+a0 (modulo-2)

The result is the parity bit. In other words, If the number of 1s is even, the result is 0; if the number of 1s is odd, the result is 1.In both cases, the total number of 1s in the code word is even.
The sender sends the code word which may be corrupted during transmission. The receiver receives a 5-bit word. The checker at the receiver does the same thing as the generator in the sender with one exception: The addition is done over all 5 bits.

s0=b3+b2+b1+b0+ q0 (modulo-2)

The result, which is called the syndrome, is just 1 bit. The syndrome is 0 when the number of 1s in the received code word is even; otherwise, it is 1.

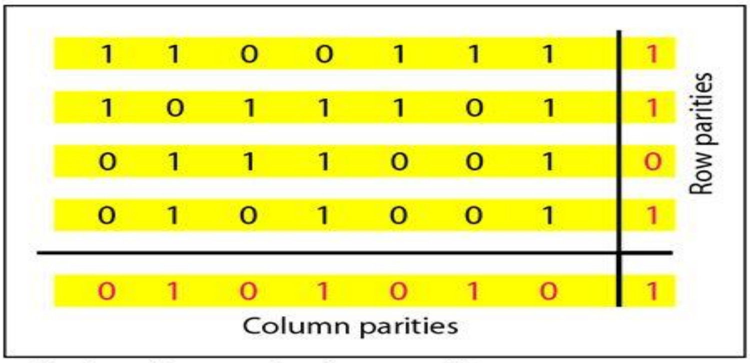
The syndrome is passed to the decision logic analyzer. If the syndrome is 0, there is no error in the received code word, the data portion of the received code word is accepted as the data word, if the syndrome is 1, the data portion of the received code word is discarded. The data word is not created.

For example, the sender sends the data word 1011. The code word created from this data word is 10111, which is sent to the receiver. We examine five cases:

1**. No error occurs**; the received code word is 10111. The syndrome is 0. The data word 1011 is created.
2. **One single-bit error changes a1**. The received code word is 10011. The syndrome is 1. No data word is created.
3. **One single-bit error changes r0**. The received code word is 10110. The syndrome is 1. No data word is created. Note that although none of the data word bits are corrupted, no data word is created because the code is not sophisticated enough to show the position of the corrupted bit.
4. **An error changes r0 and a second error changes a3**. The received code word is 00110. The syndrome is O. The data word 0011 is created at the receiver. Note that here the data word is wrongly created due to the syndrome value. The simple parity-check decoder cannot detect an even number of errors. The errors cancel each other out and give the syndrome a value of O.
5. **Three bits-a3, a2, and a1-are changed by errors**. The received code word is 01011. The syndrome is 1. The data word is not created. This shows that the simple parity check, guaranteed to detect one single error, can also find any odd number of errors.

A better approach is the two-dimensional parity check. In this method, the data word is organized in a table (rows and columns). In the following figure, the data to be sent, five 7-bit bytes, are put in separate rows. For each row and each column, 1 parity-check bit is calculated. The whole table is then sent to the receiver, which finds the syndrome for each row and each column.

As shown in the figure, the two-dimensional parity check can detect up to three errors that occur anywhere in the table (arrows point to the locations of the created nonzero syndromes). However, errors affecting 4 bits may not be detected.



a. Design of row and column parities

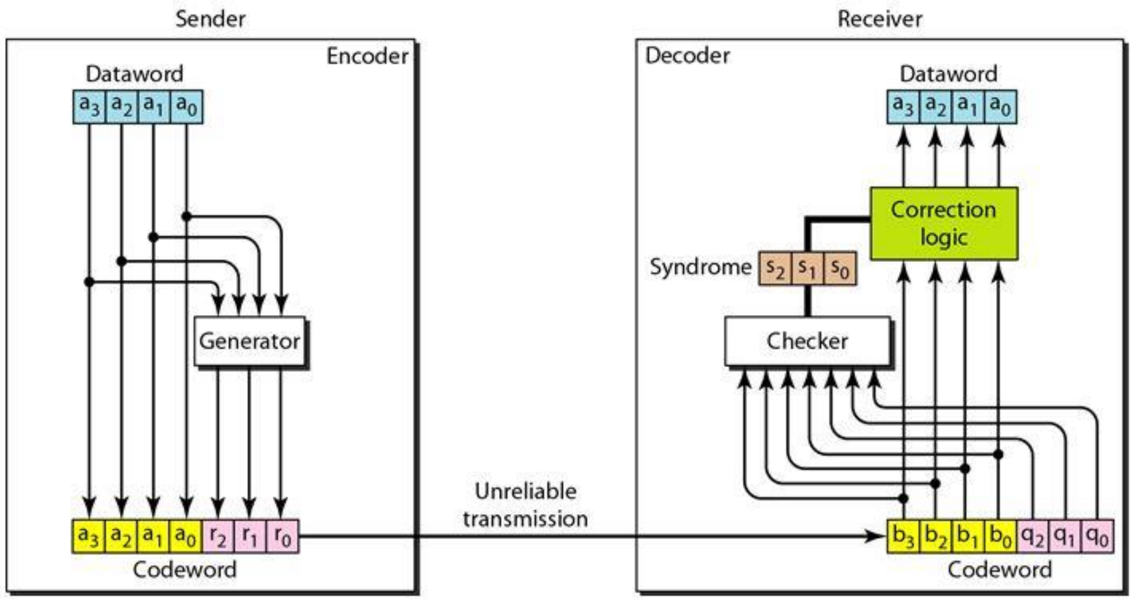
Hamming Codes:

The Hamming codes were originally designed with $d_{min} = 3$, which means that they can detect up to two errors or correct one single error. But there are some Hamming codes that can correct more than one error.

For example, if m=3, then n= 7 and k= 4. This is a Hamming code C(7, 4) with $d_{min} = 3$. The following Table shows the data words and code words for this code.

Datawords	*Codewords*	*Datawords*	*Codewords*
0000	0000000	1000	1000110
0001	0001101	1001	1001011
0010	0010111	1010	1010001
0011	0011010	1011	1011100
0100	0100011	1100	1100101
0101	0101110	1101	1101000
0110	0110100	1110	1110010
0111	0111001	1111	1111111

The following figure shows the structure of the encoder and decoder for this example.



A copy of a 4-bit data word is fed into the generator that creates three parity checks r0, r1 and r2 as shown below:

r0=a2+a1+a0 (modulo-2)
r1=a3+a2+a1(modulo-2)
r2=a1+a0+a3(modulo-2)

Each of the parity-check bits handles 3 out of the 4 bits of the data word. The total number of 1s in each 4-bit combination (3 data word bits and 1 parity bit) must be even. We are not saying that these three equations are unique; any three equations that involve 3 of the 4 bits in the data word and create independent equations (a combination of two cannot create the third) are valid.

The checker in the decoder creates a 3-bit syndrome (s2s1s0) in which each bit is the parity check for 4 out of the 7 bits in the received code word.

S0=b2+b1+b0+ q0   (modulo-2)
S1 =b3 +b2 + q1 (modulo-2)
S2=b1+b0+q3 (modulo-2)

The equations used by the checker are the same as those used by the generator with the parity-check bits added to the right-hand side of the equation. The 3-bit syndrome creates eight different bit patterns (000 to 111) that can represent eight different conditions.

These conditions define a lack of error or an error in 1 of the 7 bits of the received code word, as shown in the following Table.

Syndrome	000	001	010	011	100	101	110	111
Error	None	q_0	q_1	b_2	q_2	b_0	b_3	b_1

Note that the generator is not concerned with the four cases shaded in the above Table because there is either no error or an error in the parity bit. In the other four cases, 1 of the bits must be flipped (changed from 0 to 1 or 1 to 0) to find the correct data word.

The syndrome values in the above Table are based on the syndrome bit calculations. For example, if qo is in error, So is the only bit affected; the syndrome, therefore, is 001. If b2 is in error, So and S1 are the bits affected; the syndrome, therefore is 011. Similarly, if b1 is in error, all 3 syndrome bits are affected and the syndrome is 111.
There are two points we need to emphasize here. First, if two errors occur during transmission, the created data word might not be the right one. Second, if we want to use the above code for error detection, we need a different design.

For Example

Let us trace the path of three data words from the sender to the destination:

1. The data word 0100 becomes the code word 0100011. The code word 0100011 is received. The syndrome is 000, the final data word is 0100.
2. The data word 0111 becomes the code word 0111001. The syndrome is 011. After flipping b2 (changing the 1 to 0), the final data word is 0111.
3. The data word 1101 becomes the code word 1101000. The syndrome is 101. After flipping b0, we get 0000, the wrong data word. This shows that our code cannot correct two errors.

Cyclic codes in Error Detection and Correction

Cyclic codes are special linear block codes with one extra property. In a cyclic code, if a code word is cyclically shifted (rotated), the result is another code word. For example, if 1011000 is a code word and we cyclically left-shift, then 0110001 is also a code word.

In this case, if we call the bits in the first word a0 to a6 and the bits in the second word b0 to b6, we can shift the bits by using the following:

b1=a0 b2=a1 b3=a2 b4=a3 b5=a4 b6=a5   b0=a6

Cyclic Redundancy Check:

One of the categories of cyclic codes called the cyclic redundancy check (CRC) that is used in networks such as LANs and WANs.

The following table shows an example of a CRC code which shows both the linear and cyclic properties of this code.

Dataword	Codeword	Dataword	Codeword
0000	0000000	1000	1000101
0001	0001011	1001	1001110
0010	0010110	1010	1010011
0011	0011101	1011	1011000
0100	0100111	1100	1100010
0101	0101100	1101	1101001
0110	0110001	1110	1110100
0111	0111010	1111	1111111

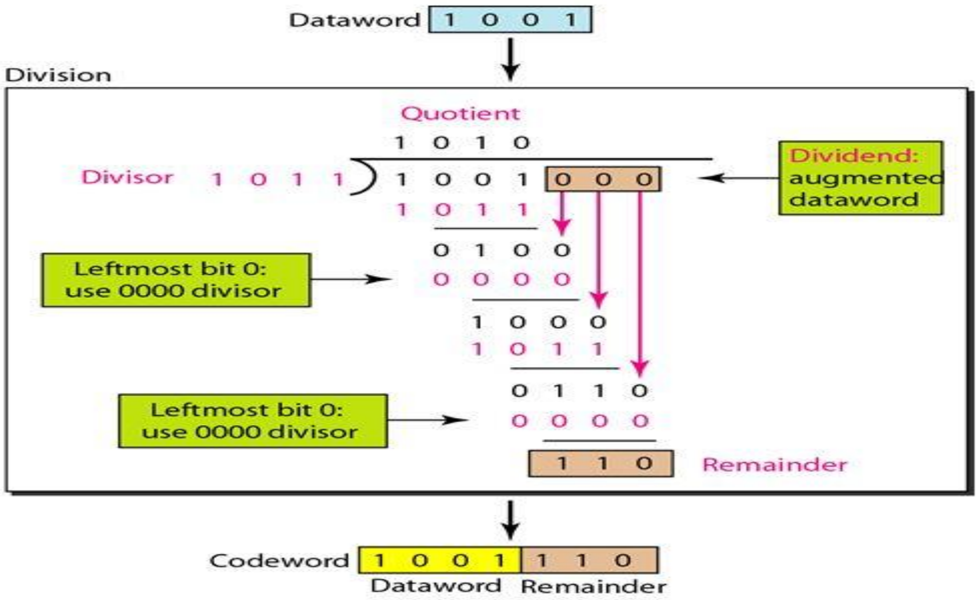
In the encoder, the data word has k bits (4 here); the code word has n bits (7 here). The size of the data word is augmented by adding n - k (3 here) 0s to the right-hand side of the word. The n-bit result is fed into the generator.

The generator uses a divisor of size n - k + I (4 here), predefined and agreed upon. The generator divides the augmented Data word by the divisor (modulo-2 division). The quotient of the division is discarded; the remainder (r2r1r0) is appended to the data word to create the code word.The decoder receives the possibly corrupted code word. A copy of all n bits is fed to the checker which is a replica of the generator. The remainder produced by the checker is a syndrome of n - k (3 here) bits, which is fed to the decision logic analyzer.

The analyzer has a simple function. If the syndrome bits are all 0s, the 4 leftmost bits of the code word are accepted as the data word (interpreted as no error); otherwise, the 4 bits are discarded (error).

Encoder:

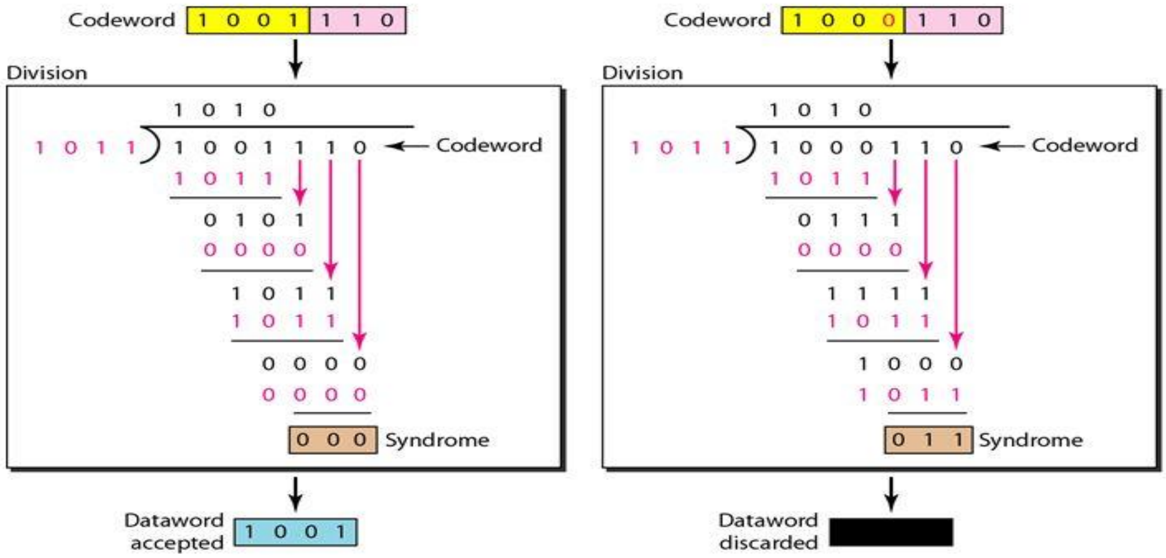
For example, consider the following figure where the encoder takes the data word and augments it with n - k number of 0s. It then divides the augmented data word by the divisor, as shown in Figure.



The process of modulo-2 binary division is the same as the familiar division process for decimal numbers. In each step, a copy of the divisor is XORed with the 4 bits of the dividend. The result of the XOR operation (remainder) is 3 bits (in this case), which is used for the next step after 1 extra bit is pulled down to make it 4 bits long. There is one important point we need to remember in this type of division. If the leftmost bit of the dividend (or the part used in each step) is 0, the step cannot use the regular divisor; we need to use an all-0s divisor.

When there are no bits left to pull down, we have a result. The 3-bit remainder forms the check bits (r2, r1 and r0). They are appended to the data word to create the code word.

Decoder:
The code word can change during transmission. The decoder does the same division process as the encoder. The remainder of the division is the syndrome. If the syndrome is all 0s, there is no error; the data word is separated from the received code word and accepted. Otherwise, everything is discarded. Consider the following figure which shows two cases: The left hand figure shows the value of syndrome when no error has occurred; the syndrome is 000. The right-hand part of the figure shows the case in which there is one single error. The syndrome is not all 0s (it is 011).



Advantages of Cyclic Codes:

The cyclic codes have a very good performance in detecting single-bit errors, double errors, an odd number of errors, and burst errors. They can easily be implemented in hardware and software. They are especially fast when implemented in hardware. This has made cyclic codes a good candidate for many networks.

Checksum

The checksum is used in the Internet by several protocols although not at the data link layer. Like linear and cyclic codes, the checksum is based on the concept of redundancy. Several protocols still use the checksum for error detection.

Consider the following example:

Our data is a list of five 4-bit numbers that we want to send to a destination. In addition to sending these numbers, we send the sum of the numbers. For example, if the set of numbers is (7, 11, 12, 0, 6), we send (7, 11, 12, 0, 6, 36), where 36 is the sum of the original numbers. The receiver adds the five numbers and compares the result with the sum. If the two are the same, the receiver assumes no error, accepts the five numbers, and discards the sum. Otherwise, there is an error somewhere and the data are not accepted.

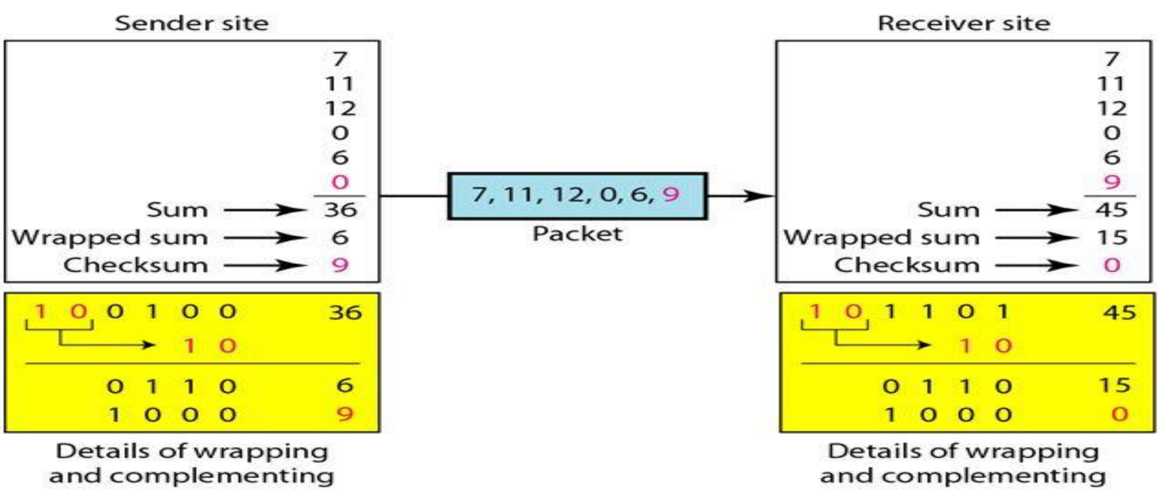
To make the job of the receiver easy if we send the negative (complement) of the sum, called the checksum. In this case, we send (7, 11, 12, 0, 6, -36). The receiver can add all the numbers received (including the checksum). If the result is 0, it assumes no error; otherwise, there is an error.

One's Complement:

The previous example has one major drawback. All of our data can be written as a 4-bit word (they are less than 15) except for the checksum. One solution is to use one's complement arithmetic. In this arithmetic, we can represent unsigned numbers between 0 and 2n - 1 using only n bits. If the number has more than n bits, the extra leftmost bits need to be added to the n rightmost bits (wrapping).

In one's complement arithmetic, a negative number can be represented by inverting all bits (changing a 0 to a 1 and a 1 to a 0). This is the same as subtracting the number from 2n - 1.

The following Figure shows the process at the sender and at the receiver.



The sender initializes the checksum to 0 and adds all data items and the checksum (the checksum is considered as one data item and is shown in color). The result is 36. However, 36 cannot be expressed in 4 bits. The extra two bits are wrapped and added with the sum to create the wrapped sum value 6.

In the figure, we have shown the details in binary. The sum is then complemented, resulting in the checksum value 9 (15 - 6 = 9). The sender now sends six data items to the receiver including the checksum 9. The receiver follows the same procedure as the sender. It adds all data items (including the checksum); the result is 45. The sum is wrapped and becomes 15. The wrapped sum is complemented and becomes 0. Since the value of the checksum is 0, this means that the data is not corrupted. The receiver drops the checksum and keeps the other data items. If the checksum is not zero, the entire packet is dropped.