UNIT I

DATA COMMUNCICATION

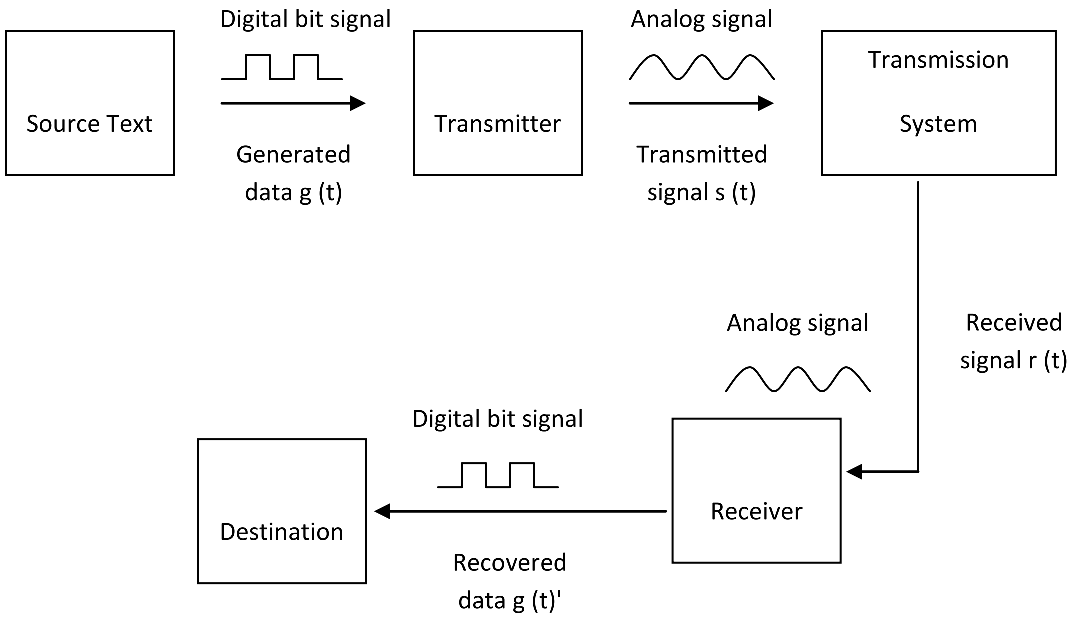
Data communication refers to the exchange of data between two devices via some form of transmission medium. Data communication is said to be local if communicating devices are in the same building or a similarly restricted geographical area. It is said to be remote if the devices are farther apart. There are three main characteristics of an efficient data communication system:

Reliable Delivery: Data sent from a source across the communication system must be delivered only to the intended destination.

Accuracy: Data should be delivered to the destination without any change or alteration.

Timely Delivery: Data should be delivered in time without any delay, otherwise it may loose its significance. For example, as in case of audio transmission delay in delivery may effect the communication process.

Simplified Data Communications Model



Components of data communication system

A communication system has following components:

1. Message: It is the information or data to be communicated. It can consist of text, number, pictures, sound or video or any combination of these.
2. Sender: It is the device/computer that generates and sends that message.
3. Receiver: It is the device or computer that receives the message. The location of receiver computer is generally different from the sender computer. The distance between sender and receiver depends upon the types of network used in between.
4. Medium: It is the channel or physical path through which the message is carried from sender to the receiver. The medium can be wired like twisted pair wire, coaxial cable, fibre-optic cable or wireless like laser, radio waves, microwaves.
5. Protocol: It is a set of rules that govern the communication between the devices. Both sender and receiver follow same protocols to communicate with each other.

CONCEPT OF BANDWIDTH AND CHANNEL CAPACITY

A communication channel is characterized by two main (related) attributes i.e. bandwidth and channel capacity.

Bandwidth

In computer networks, bandwidth is used as a synonym for data transfer rate, the amount of data that can be carried from one point to another in a given time period (usually a second). Network bandwidth is usually expressed in bits per second (bps); modern networks typically have speeds measured in the millions of bits per second (megabits per second, or Mbps) or billions of bits per second (gigabits per second, or Gbps). Note that bandwidth is not the only factor that affects network performance: There is also packet loss, latency and jitter, all of which degrade network throughput and make a link perform like one with lower bandwidth. A network path usually consists of a succession of links, each with its own bandwidth, so the end-to-end bandwidth is limited to the bandwidth of the lowest speed link (the bottleneck). Different applications require different bandwidths. An instant messaging conversation might take less than 1,000 bits per second (bps); a voice over IP (VoIP) conversation requires 56 kilobits per second (Kbps) to sound smooth and clear. Standard definition video (480p) works at 1 megabit per second (Mbps), but HD video (720p) wants around 4 Mbps, and HDX (1080p), more than 7 Mbps. Effective bandwidth -- the highest reliable transmission rate a path can provide -- is measured with a bandwidth test. This rate can be determined by repeatedly measuring the time required for a specific file to leave its point of origin and successfully download at its destination.

Bandwidth can also be defined as the range of frequencies -- the difference between the highest frequency signal component and the lowest-frequency signal component -- an electronic signal uses on a given transmission medium. Like the frequency of a signal, bandwidth is measured in hertz (cycles per second). This is the original meaning of bandwidth, although it is now used primarily in discussions about cellular networks and the spectrum of frequencies that operators license from various governments for use in mobile services.

Example: If a periodic signal is decomposed into five sine waves with frequencies of 100, 300, 500, 700, and 900 Hz, what is its bandwidth?

Sol: Let fh be the highest frequency, fl the lowest frequency, and B the bandwidth.

Then B =fh -fl = 900 -100 = 800 Hz

Channel Capacity

The maximum rate at which the data can be transmitted over a given communication path or a channel, under given conditions is known as Channel capacity.

The data rate of a channel usually depends on the three factors:

- Bandwidth
- Level of the signal
- Quality of the channel

The quality of a channel is basically concerned with the level of noise present on the channel.

<u>Quantifying Channel Capacity</u>

There are two different theoretical formulas to calculate data rate of a channel:

1. Nyquist theorem for noiseless channel
2. Shannon capacity formula for noisy channel

Channel Capacity for Noiseless Channel

For a noiseless channel, the Nyquist bit rate formula defines the theoretical maximum bit rate.

$$\textbf{BitRate = 2 x bandwidth x log2 L}$$

In this formula, bandwidth is the bandwidth of the channel, L is the number of signal levels used to represent data, and BitRate is the bit rate in bits per second. According to the formula, we might think that, given a specific bandwidth, we can have any bit rate we want by increasing the number of signa11eve1s. Although the idea is theoretically correct, practically there is a limit. When we increase the number of signal leve1s, we impose a burden on the receiver. If the number of levels in a signal is just 2, the receiver can easily distinguish between a 0 and a 1. If the level of a signal is 64, the receiver must be very sophisticated to distinguish between 64 different levels. In other words, increasing the levels of a signal reduces the reliability of the system.

<u>Example of Nyquist Law</u>

Assuming a simple telephone line is noise free within a bandwidth of 3000 Hz (range: 1000-4000), and only two(2) discrete signal levels can be distinguished on a simple telephone line, the maximum data rate or channel capacity is :

$$\text{Bit rate} = 2 * 3000 * \text{Log}_2 2$$

= 6000 bits/sec

For a fixed Bandwidth channel, its capacity can be improved by increasing V, i.e. introducing more discrete signal levels by using modulation techniques (amplitude, frequency) and data compression.

This requires better hardware at the sending and receiving end of the channel, If V = 8, C = 18 Kbps
Data compression techniques may be used to further increase the channel capacity.

Channel Capacity for Noisy Channel

In reality, we cannot have a noiseless channel; the channel is always noisy. In 1944, Claude Shannon introduced a formula, called the Shannon capacity, to determine the theoretical highest data rate for a noisy channel:

$$\text{Capacity} = \text{bandwidth} \times \log_2 (1 + \text{SNR})$$

In this formula, bandwidth is the bandwidth of the channel, SNR is the signal-to-noise ratio, and capacity is the capacity of the channel in bits per second. Note that in the Shannon formula there is no indication of the signal level, which means that no matter how many levels we have, we cannot achieve a data rate higher than the capacity of the channel. In other words, the formula defines a characteristic of the channel, not the method of transmission.

Signal to Noise Ratio

In analog and digital communications, signal-to-noise ratio, often written S/N or SNR, is a measure of signal strength relative to background noise. The ratio is usually measured in decibels (dB) using a signal-to-noise ratio formula. The amount of thermal noise present is measured by the ratio of the signal power to the noise power called signal-to-noise ratio.

$$\text{SNR}=\frac{\text{Average Signal Power}}{\text{Average Noise Power}}$$

Communications engineers always strive to maximize the S/N ratio. Traditionally, this has been done by using the narrowest possible receiving-system bandwidth consistent with the data speed desired. However, there are other methods. In some cases, spread spectrum techniques can improve system performance. The S/N ratio can be increased by providing the source with a higher level of signal output power if necessary.

Example

We can calculate the theoretical highest bit rate of a regular telephone line. A telephone line normally has a bandwidth of 3000 Hz (300 to 3300 Hz) assigned for data communications. The signal-to-noise ratio is usually 3162. For this channel the capacity is calculated as

$$C = B \log_2 (1+ \text{SNR}) = 3000 \log_2 (l + 3162) = 3000 \log_2 3163$$
$$= 3000 \times 11.62 = 34{,}860 \text{ bps}$$

This means that the highest bit rate for a telephone line is 34.860 kbps. If we want to send data faster than this, we can either increase the bandwidth of the line or improve the signal-to- noise ratio.

Capacity: Data rate & Baud Rate

Baud Rate: Baud rate is the number of signal elements transmitted on a channel per second. The signal element may be discrete voltage, phase or frequency value, which identifies a unique symbol transmitted on the channel. Shannon's Law establishes the maximum baud rate for a channel (i.e. symbols per second).

Bit rate: Data rate is the maximum rate at which information bits may be transmitted on a channel. This may exceed the baud rate if a signal element represents more than one bit.

Example: data rate vs. baud rate

Assume we need to download text documents at the rate of 100 pages per minute.
What is the required bit rate of the channel?

Sol:A page is an average of 24 lines with 80 characters in each line. If we assume that one character requires 8 bits, the bit rate is 100 x 24 x 80 x 8 =1,636,000 bps =1.636 Mbps

- If only two voltage levels are used, a signal element can represent a 1 or 0 i.e. only one bit. In this case, baud rate is the same as data rate.
- If 8 voltage levels are used, a signal element can represent 3 bits (v1=000, v2=001, v3=111). In this case, the data rate is three times the baud rate.

Bit Length
The bit length is the distance one bit occupies on the transmission medium.

Bit length =propagation speed x bit duration

Analog and Digital Transmission

Both analog and digital signals may be transmitted on suitable transmission media. The way these signals are treated is a function of the transmission system. Table 3.1 summarizes the methods of data transmission. Analog transmission is a means of transmitting analog signals without regard to their content; the signals may represent analog data (e.g., voice) or digital data (e.g., binary data that pass through a modem). In either case, the analog signal will become weaker (attenuate) after a certain distance. To achieve longer distances, the analog transmission system includes amplifiers that boost the energy in the signal. Unfortunately, the amplifier also boosts the noise components. With amplifiers cascaded to achieve long distances, the signal becomes more and more distorted.

(a) Data and Signals

	Analog Signal	Digital Signal
Analog Data	Two alternatives: (1) signal occupies the same spectrum as the analog data; (2) analog data are encoded to occupy a different portion of spectrum.	Analog data are encoded using a codec to produce a digital bit stream.
Digital Data	Digital data are encoded using a modem to produce analog signal.	Two alternatives: (1) signal consists of two voltage levels to represent the two binary values; (2) digital data are encoded to produce a digital signal with desir ed properties.

(b) Treatment of Signals

	Analog Transmission	Digital Transmission
Analog Signal	Is propagated through amplifiers; same treatment whether signal is used to represent analog data or digital data.	Assumes that the analog signal represents digital data. Signal is propagated through repeaters; at each repeater, digital data are recovered from inbound signal and used to generate a new analog outbound signal.
Digital Signal	Not used	Digital signal represents a stream of 1s and 0s, which may represent digital data or may be an encoding of analog data. Signal is propagated through repeaters; at each repeater, stream of 1s and 0s is recovered from inbound signal and used to generate a new digital outbound signal.

For analog data, such as voice, quite a bit of distortion can be tolerated and the data remain intelligible. However, for digital data, cascaded amplifiers will introduce errors.
Digital transmission, in contrast, assumes a binary content to the signal. A digital signal can be transmitted only a limited distance before attenuation, noise, and other impairments endanger the integrity of the data. To achieve greater distances, repeaters are used. A repeater receives the digital signal, recovers the pattern of 1s and 0s, and retransmits a new signal. Thus the attenuation is overcome. The same technique may be used with an analog signal if it is assumed that the signal carries digital data. At appropriately spaced points, the transmission system has repeaters rather than amplifiers. The repeater recovers the digital data from the analog signal and generates a new, clean analog signal. Thus noise is not cumulative. The question naturally arises as to which is the preferred method of transmission. The answer being supplied by the telecommunications industry and its customers is digital. Both long-haul telecommunications facilities and intra building services have moved to digital transmission and, where possible, digital signaling techniques

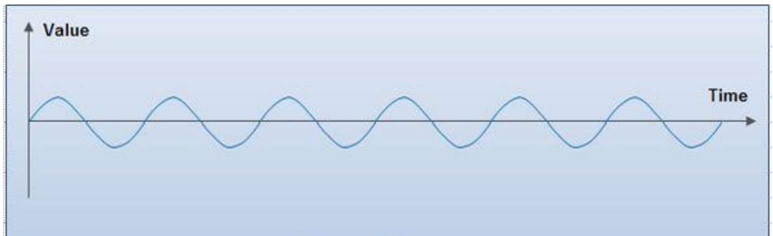
Periodic and Nonperiodic Signals

Both analog and digital signals can take one of two forms: periodic or nonperiodic (sometimes refer to as aperiodic, because the prefix a in Greek means "non"). A periodic signal completes a pattern within a measurable time frame, called a period, and repeats that pattern over subsequent identical periods. The completion of one full pattern is called a cycle. A nonperiodic signal changes without exhibiting a pattern or cycle that repeats over time. Both analog and digital signals can be periodic or nonperiodic. In data communications, we commonly use periodic analog signals (because they need less bandwidth,) and nonperiodic digital signals (because they can represent variation in data)

Analog Signal

An analog signal is one type of continuous time-varying signals, and these are classified into composite and simple signals. A simple type of analog signal is nothing but a sine wave, and that can't be decomposed, whereas a composite type analog signal can be decomposed into numerous sine waves. An analog signal can be defined by using amplitude, time period otherwise frequency & phase. A simple periodic analog signal, a sine wave, cannot be decomposed into simpler signals. A composite periodic analog signal is composed of multiple sine waves.

The sine wave is the most fundamental form of a periodic analog signal. When we visualize it as a simple oscillating curve, its change over the course of a cycle is smooth and consistent, a continuous, rolling flow. A sine wave can be represented by three parameters: the peak amplitude, the frequency, and the phase. These three parameters fully describe a sine wave.

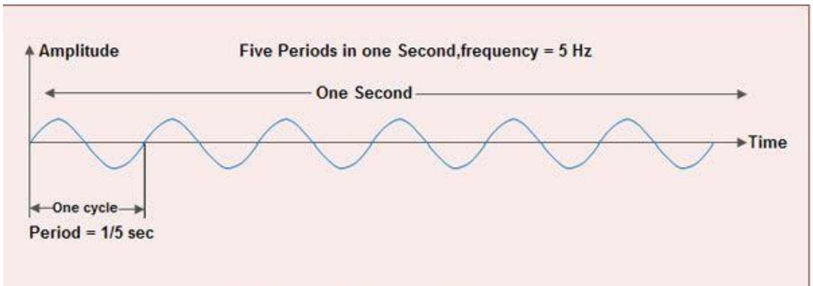


Sine Wave

Characteristics of Analog Signal

Amplitude

Amplitude of a signal refers to the height of the signal. It is equal to the vertical distance from a given point on the waveform to the horizontal axis. The maximum amplitude of a sine wave is equal to the highest value it reaches on the vertical axis as shown in figure. Amplitude is measured in volts, amperes or watts depending on the type of signal. A volt is used for voltage, ampere for current and watts for power.

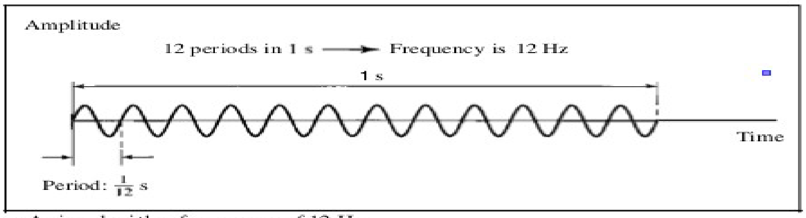


Period & Frequency

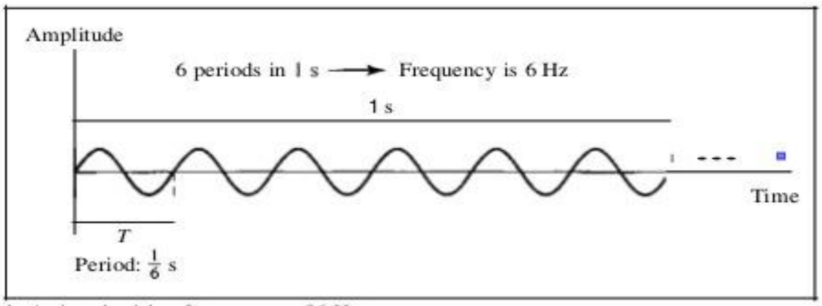
Period and Frequency

Period refers to the amount of time, in seconds, a signal needs to complete 1 cycle. Frequency refers to the number of periods in I s. Note that period and frequency are just one characteristic defined in two ways. Period is the inverse of frequency, and frequency is the inverse of period, as the following formulas show.

$$f = 1/T \text{ and } T = 1/f$$



a. A signal with a frequency of 12 Hz

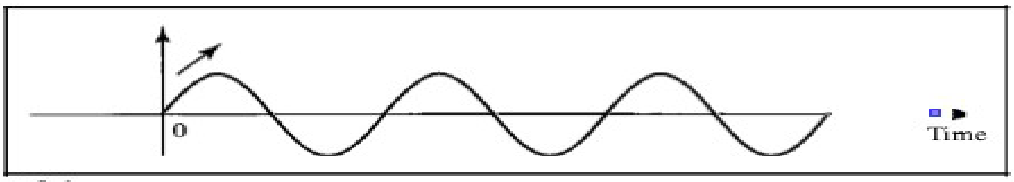


b. A signal with a frequency of6 Hz

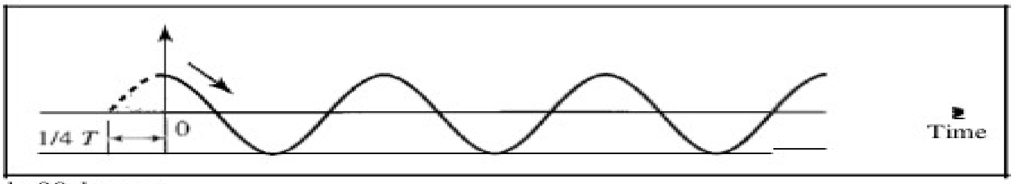
Period is formally expressed in seconds. Frequency is formally expressed in Hertz (Hz), which is cycle per second.

Phase

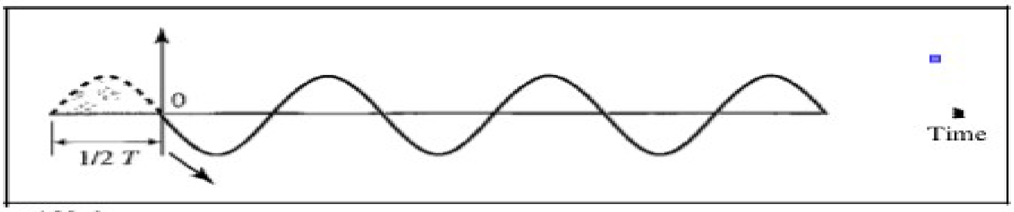
The term phase describes the position of the waveform relative to time O. If we think of the wave as something that can be shifted backward or forward along the time axis, phase describes the amount of that shift. It indicates the status of the first cycle.



a. 0 degrees



b. 90 degrees



c. 180 degrees

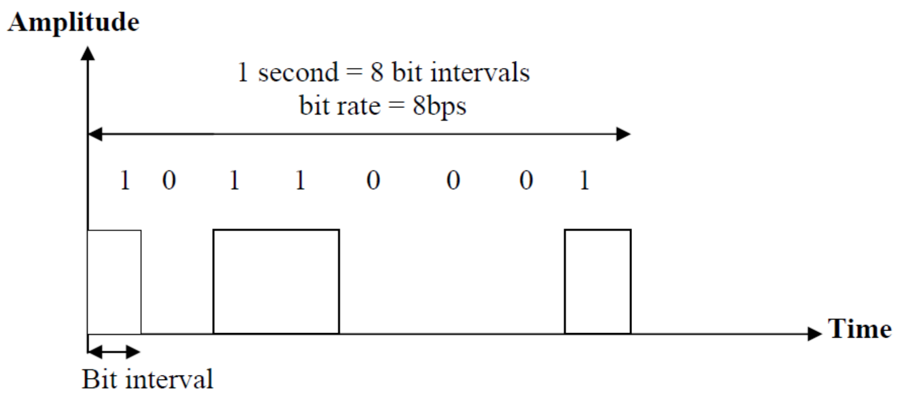
Phase is measured in degrees or radians [360° is 2n rad; 1° is 2n/360 rad, and 1 rad is 360/(2n)].
A phase shift of 360° corresponds to a shift of a complete period; a phase shift of 180° corresponds to a shift of one-half of a period; and a phase shift of 90° corresponds to a shift of one-quarter of a period (see Figure)

Wavelength

Wavelength is another characteristic of a signal travelling through a transmission medium. Wavelength binds the period or the frequency of a simple sine wave to the propagation speed of the medium while the frequency of a signal is independent of the medium, the wavelength depends on both the frequency and the medium. Wavelength is a property of any type of signal. In data communications, we often use wavelength to describe the transmission of light in an optical fiber. The wavelength is the distance a simple signal can travel in one period.

Digital Signal

A digital signal is a discrete-time signal for which not only the time but also the amplitude has discrete values; in other words, its samples take on only values from a discrete set (a countable set that can be mapped one-to-one to a subset of integers). Timing graphs of these signals look like square waves**.** Most digital signals are aperiodic and thus, period or frequency is not appropriate. Two new terms, bit interval (instead of period) and bit rate (instead of frequency) are used to describe digital signals.



The bit interval is the time required to send one single bit. The bit rate is the number of bit interval per second. This mean that the bit rate is the number of bits send in one second, usually expressed in bits per second (bps) as shown in Fig1. A digital signal can be considered as a signal with an infinite number of frequencies and transmission of digital requires a low-pass channel as shown in Fig2. On the other hand, transmission of analog signal requires band-pass channel shown in Fig. 3.

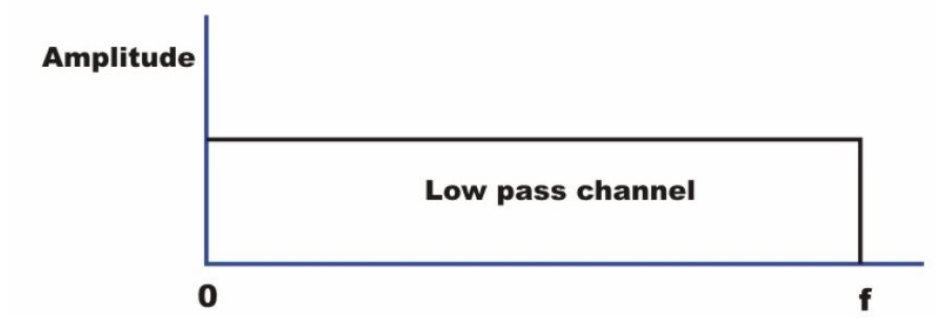


Fig1. Low pass channel required for Digital signal

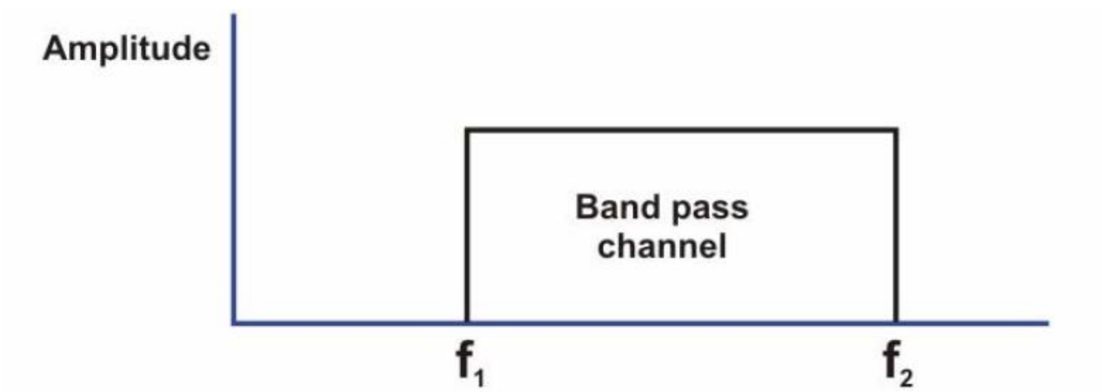
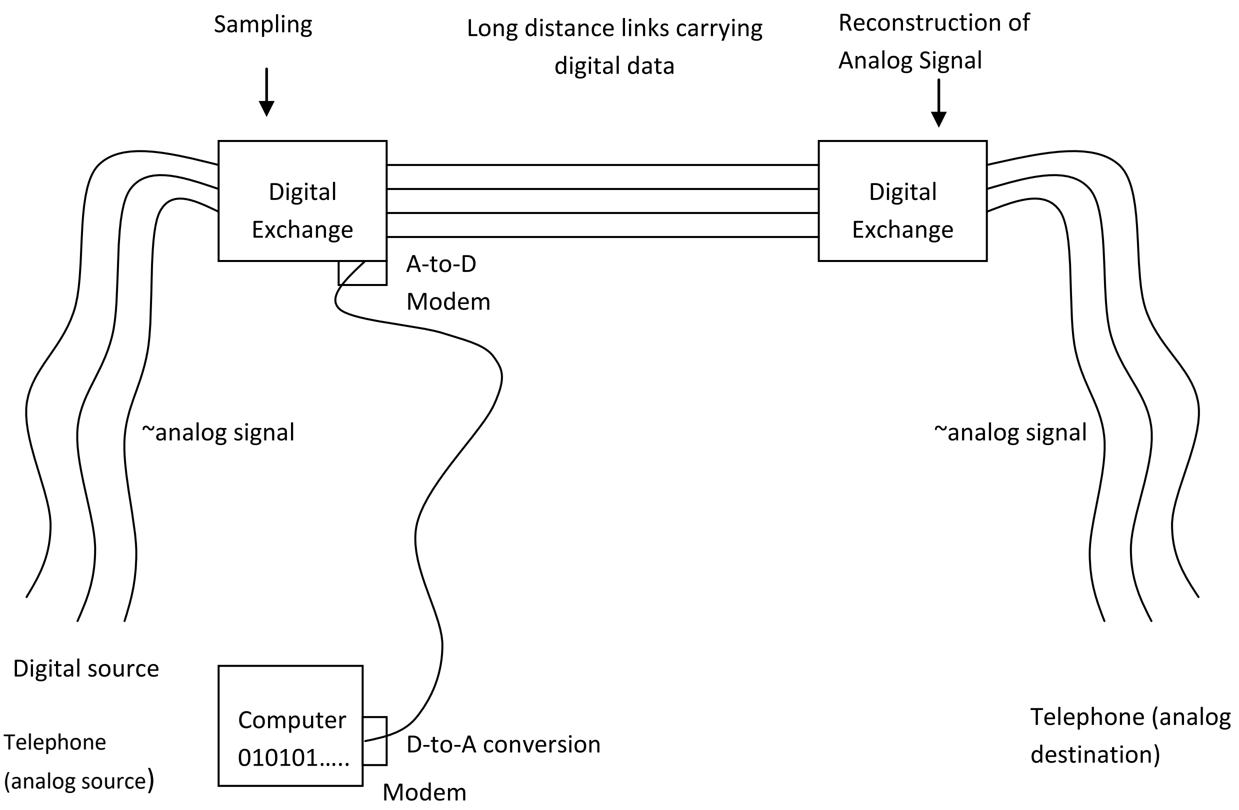
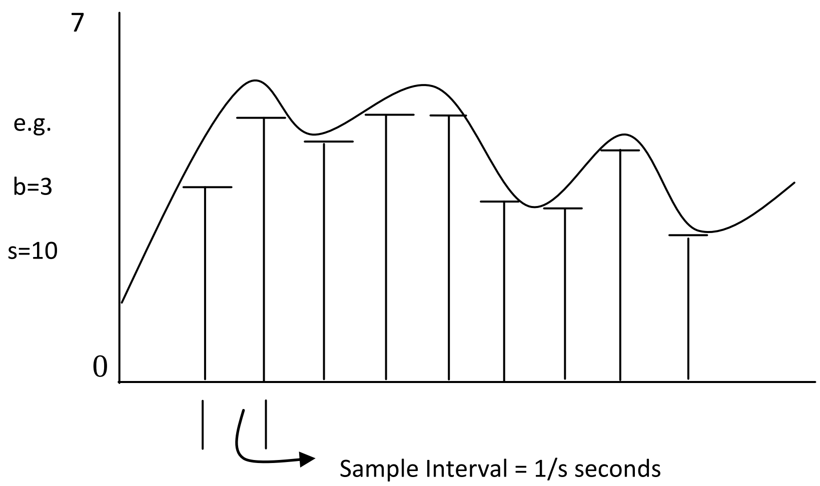


Fig2. Low pass channel require for Analog signal

Digital Telephone System



Telephone signal which is analog up to exchange is sampled at the exchange by taking 5 samples per second. Each sample value is represented by b bits/sample. So s * b bits/second are generated.



Sampling

These bits are transmitted via a digital link (dedicated to a conversation) between the source and destination telephone exchanges. At the destination exchange the analog signal is reconstructed from the sample bits, and sent to the receiving telephone.

Analog-to-Digital Conversion

Voice and video signals are inherently analog in nature. These signals must be converted to digital form before transmission over a channel. This is accomplished in two steps:

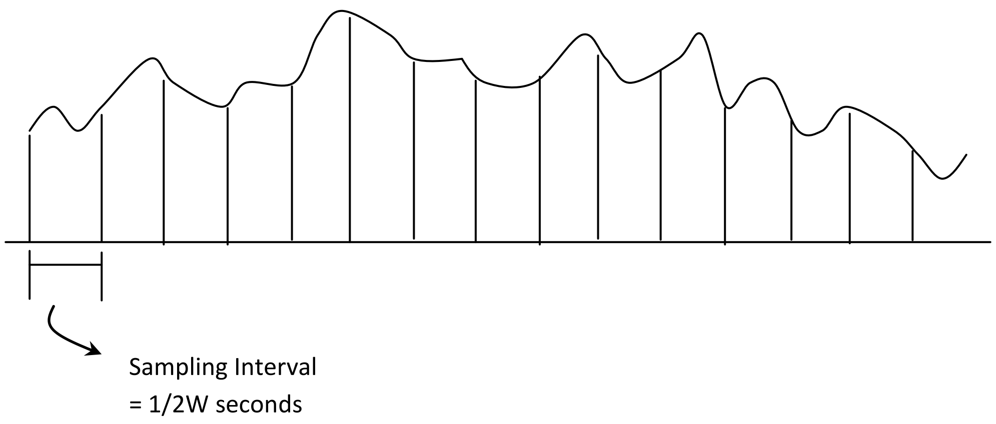
1. Sampling
2. Pulse Code Modulation (PCM)

Nyquist Criterion for Sampling

- The Nyquist criterion for sampling states that an analog signal which contains frequency components no higher than H can be completely reconstructed by obtaining only 2H samples per second.
- For example, an analog signal containing frequencies within 4000 Hz can be reconstructed by taking samples at a rate of 2 * 4000 samples per second.

<u>Sampling Techniques</u>

W = 4 KHz, T_s 1/8000 = 125μs



Pulse Code modulation (PCM)

The value of each sample (i.e. the amplitude of the signal at the sampling point) is represented by certain number of bits. This set of bits is then transmitted over a digital transmission line-a technique called PCM. For example, when analog voice signal is received at a digital exchange, it is sampled using 8 bits per sample. This requires that inter-exchange digital channel support 8000 samples/sec * 8 bits/sample, or, 64 Kbps. This channel capacity (also called as DSØ signal) is the most basic building block of digital telephone lines.

LOCAL AREA NETWORKS

Local area networks, generally called LANs, are privately-owned networks within a single building or campus of up to a few kilometers in size. They are widely used to connect personal computers and workstations in company offices and factories to share resources (e.g., printers) and exchange information. LANs are distinguished from other kinds of networks by three characteristics: (1) their size, (2) their transmission technology, and (3) their topology. LANs are restricted in size, which means that the worst-case transmission time is bounded and known in advance. Knowing this bound makes it possible to use certain kinds of designs that would not otherwise be possible. It also simplifies network management. Although a LAN can be used as an isolated network to connect computers in an organization for the sole purpose of sharing resources, most LANs today are also linked to a wide area network (WAN) or the Internet. The LAN market has seen several technologies such as Ethernet, Token Ring, Token Bus, FDDI, and ATM LAN. Some of these technologies survived for a while, but Ethernet is by far the dominant technology.

LANs may use a transmission technology consisting of a cable to which all the machines are attached, like the telephone company party lines once used in rural areas. Traditional LANs run at speeds of 10 Mbps to 100 Mbps, have low delay (microseconds or nanoseconds), and make very few errors. Newer LANs operate at up to 10 Gbps.

Some of the characteristics of LAN are:

- LAN is privately owned network that operates in a very small geographic area upto few kilometres.
- It is used to link devices in a single office, building or a campus
- A LAN is used to connect the computers and other network devices so that the devices can communicate with each other to share the resources.
- The resources to be shared can be a hardware device like a printer,software like an application program or data.
- The size of LAN is usually small. It is determined by the licesing restrictions on the number of users per copy of software, or by restrictions on the number of users licensed to access operating system.
- The various devices in LAN are connected to central devices called Hub or switch using a cable
- Hubs and switches are the communication devices used in the network. The cables from the computer to the hub allow the data transmission to pass from one computer to another.
- Now a days LAN’s are being installed using wireless technologies. Such a system makes use of access point or AP’s to transmit and receive data.
- LAN’s are also distinguished from MAN’s and WAN’s on the basis of transmission media and topology. In general, a given LAN will only use one type of transmission medium. The most common types of topologies used in LAN are bus, ring and star.

The increasing use of distributed processing applications and personal computers has led to a need for a flexible strategy for local networking. Support of premises-wide data communications requires a networking service that is capable of spanning the distances involved and that interconnects equipment in a single (perhaps large) building or a cluster of buildings. Although it is possible to develop a single LAN to interconnect all the data processing equipment of a premises, this is probably not a practical alternative in most cases.

There are several drawbacks to a single-LAN strategy:

• Reliability: With a single LAN, a service interruption, even of short duration, could result in a major disruption for users.
• Capacity: A single LAN could be saturated as the number of devices attached to the network grows over time.
• Cost: A single LAN technology is not optimized for the diverse requirements of interconnection and communication. The presence of large numbers of low cost microcomputers dictates that network support for these devices be provided at low cost. LANs that support very low-cost attachment will not be suitable for meeting the overall requirement.A more attractive alternative is to employ lower-cost, lower-capacity LANs within buildings or departments and to interconnect these networks with a higher-capacity LAN.This latter network is referred to as a backbone LAN. If confined to a single building or cluster of buildings, a high-capacity LAN can perform the backbone function.

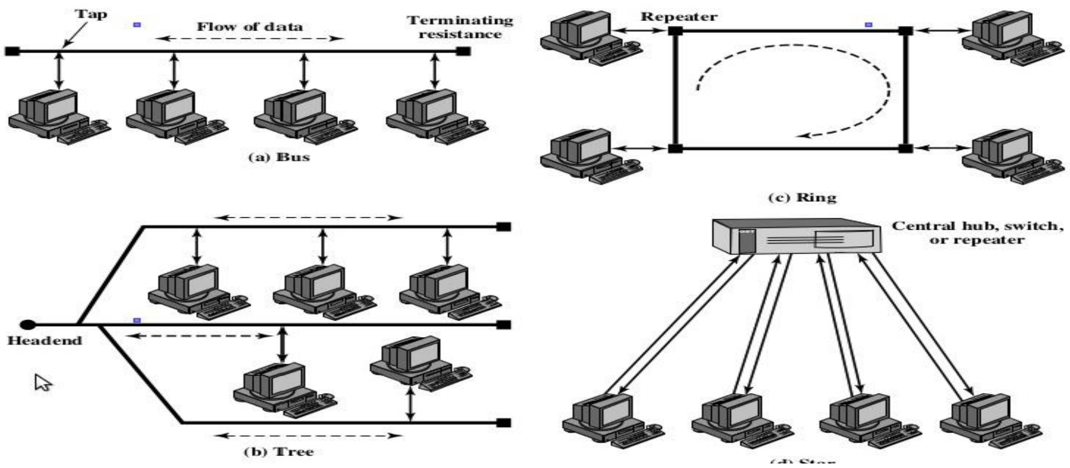
The key elements of a LAN are:

1. Topology
2. Transmission medium
3. Wiring layout
4. Medium access control

Together, these elements determine not only the cost and capacity of the LAN, but also the type of data that may be transmitted, the speed and efficiency of communications, and even the kinds of applications that can be supported.

LAN TOPOLOGIES

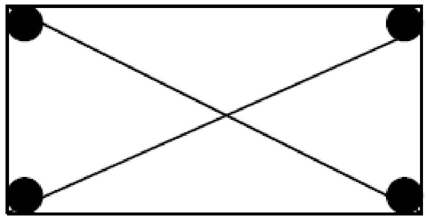
Topology refers to the way in which the network of computers is connected. Each topology is suited to specific tasks and has its own advantages and disadvantages. The choice of topology is dependent upon type and number of equipment being used, planned applications and rate of data transfer required, response time, and cost. Topology can also be defined as the geometrically interconnection pattern by which the stations (nodes/computers) are connected using suitable transmission media (which can be point to-point and broadcast). Various commonly used topologies are discussed as below:



Mesh Topology

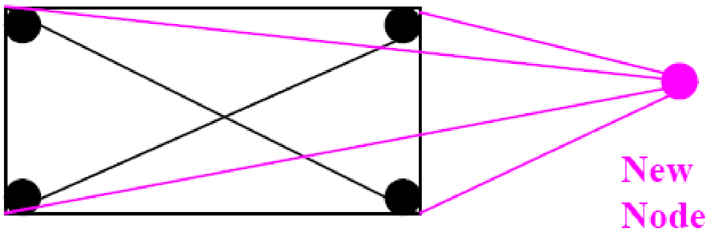
In this topology each node or station is connected to every other station as shown in Fig(a). The key characteristics of this topology are as follows:

o Fully connected
o Robust – Highly reliable
o Not flexible
o Poor expandability



Fig(a) Mesh Topology

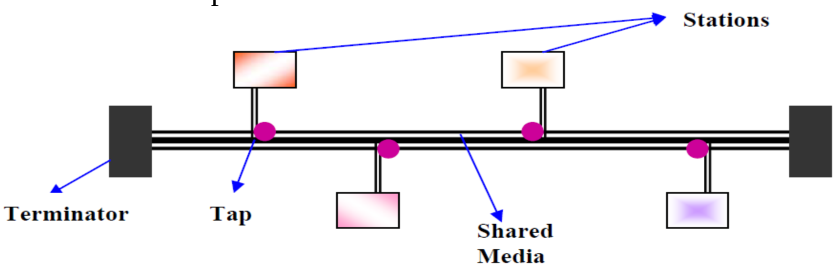
Two nodes are connected by dedicated point-point links between them. So the total number of links to connect n nodes = n(n-1)/2; which is proportional to n2 . Media used for the connection (links) can be twisted pair, co-axial cable or optical fiber. With this topology there is no need to provide any additional information, that is from where the packet is coming, along with the packet because two nodes have a point-point dedicated link between them. And each node knows which link is connected to which node on the other end. Mesh Topology is not flexible and has a poor expandability as to add a new node n links have to be laid because that new node has to be connected to each of the existing nodes via dedicated link as shown in Fig. (b). For the same reason the cost of cabling will be very high for a larger area. And due to these reasons this topology is rarely used in practice.



Fig(b) Adding a new node in Mesh Topology

Bus Topology

In Bus Topology, all stations attach through appropriate hardware interfacing known as a tap, directly to a linear transmission medium, or bus as shown in Fig.(c). Full-duplex operation between the station and the tap allows data to be transmitted onto the bus and received from the bus. A transmission from any station propagates the length of the medium in both directions and can be received by all other stations. At each end of the bus there is a terminator, which absorbs any signal, preventing reflection of signal from the endpoints. If the terminator is not present, the endpoint acts like a mirror and reflects the signal back causing interference and other problems.



Fig(c)Bus Topology

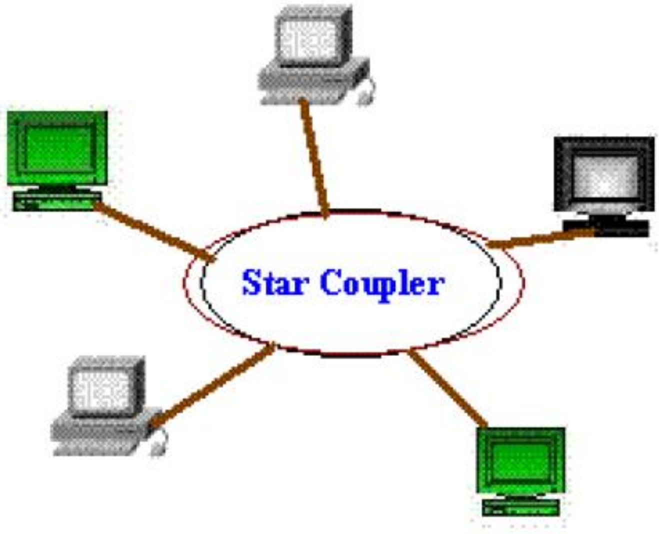
Key Characteristics of this topology are:

o Flexible
o Expandable
o Moderate Reliability
o Moderate performance

A shared link is used between different stations. Hence it is very cost effective. One can easily add any new node or delete any node without affecting other nodes; this makes this topology easily expandable. Because of the shared medium, it is necessary to provide some extra information about the desired destination, i.e. to explicitly specify the destination in the packet, as compared to mesh topology. This is because the same medium is shared among many nodes. As each station has a unique address in the network, a station copies a packet only when the destination address of the packet matches with the self-address. This is how data communications take place among the stations on the bus. As there are dedicated links in the mess topology, there is a possibility of transferring data in parallel. But in bus topology, only one station is allowed to send data at a time and all other stations listen to it, as it works in a broadcast mode. Hence, only one station can transfer the data at any given time. Suitable medium access control technique should be used so as to provide some way to decide “who” will go next to send data? Usually a distributed medium access control technique, as discussed in the next lesson, is used for this purpose. As the distance through which signal traverses increases, the attenuation increases. If the sender sends data (signal) with a small strength signal, the farthest station will not be able to receive the signal properly. While on the other hand if the transmitter sends the signal with a larger strength (more power) then the farthest station will get the signal properly but the station near to it may face over-drive. Hence, delay and signal unbalancing will force a maximum length of shared medium, which can be used in bus topology.

STAR Topology

In the star topology, each station is directly connected to a common central node as shown in Fig. (d) Typically, each station attaches to a central node, referred to as the star coupler, via two point-to-point links, one for transmission and one for reception.



Fig(d) Star Toplogy

Key features of Star topology are:
o High Speed
o Very Flexible
o High Reliability
o High Maintainability

In general, there are two alternatives for the operation of the central node:

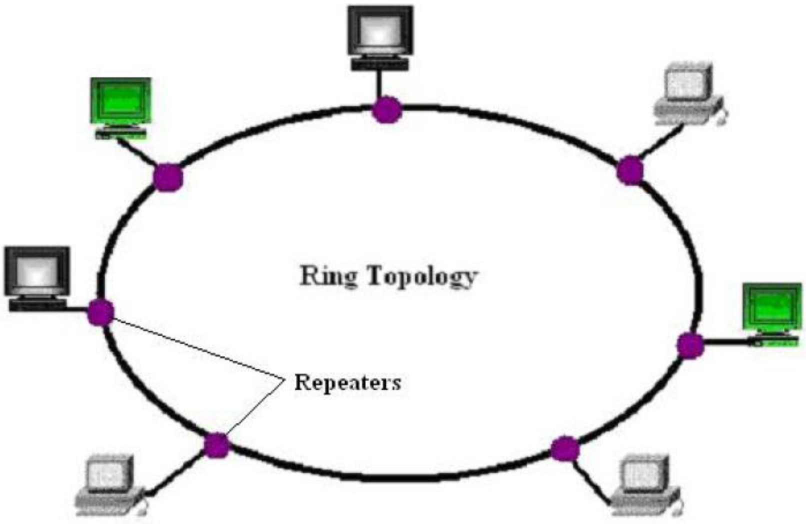
o One approach is for the central node to operate in a broadcast fashion. A transmission of a frame from one station to the node is retransmitted on all of the outgoing links. In this case, although the arrangement is physically a star, it is logically a bus; a transmission from any station is received by all other stations, and only one station at a time may successfully transmit. In this case the central node acts as a repeater.

o Another approach is for the central node to act as a frame-switching device. An incoming frame is buffered in the node and then retransmitted on an outgoing link to the destination station. In this approach, the central node acts as a switch and performs the switching or routing function. This mode of operation can be compared with the working of a telephone exchange, where the caller party is connected to a single called party and each pair of subscriber who needs to talk have a different connection.

Very High speeds of data transfer can be achieved by using star topology, particularly when the star coupler is used in the switch mode. This topology is the easiest to maintain, among the other topologies. As the number of links is proportional to n, this topology is very flexible and is the most preferred topology.

Ring topology

In the ring topology, the network consists of a set of repeaters joined by point-to-point links in a closed loop as shown in Fig.(e). The repeater is a comparatively simple device, capable of receiving data on one link and transmitting them, bit by bit, on the other link as fast as they are received, with no buffering at the repeater. The links are unidirectional; that is data are transmitted in one direction only and all are oriented in the same way. Thus, data circulate around the ring in one direction (clockwise or counter clockwise).



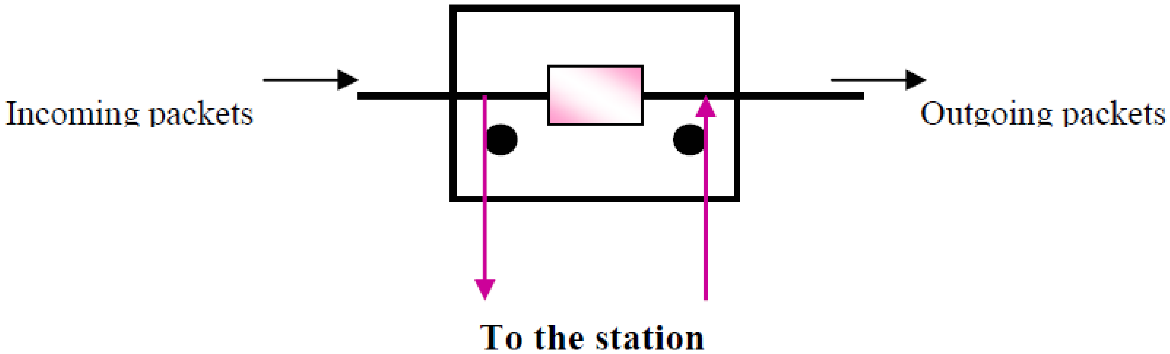
Fig(e) Ring Topology

Each station attaches to the network at a repeater and can transmit data onto the network through that repeater. As with the bus and tree, data are transmitted in frames. As a frame circulates past all the other stations, the destination station recognizes its address and copies the frame into a local buffer as it goes by. The frame continues to circulate until it returns to the source station, where it is removed. Because multiple stations share the ring, medium access control is needed to determine at what time each station may insert frames.

How the source knows whether it has to transmit a new packet and whether the previous packet has been received properly by the destination or not. For this, the destination change a particular bit (bits) in the packet and when the receiver sees that packet with the changed bit, it comes to know that the receiver has received the packet.

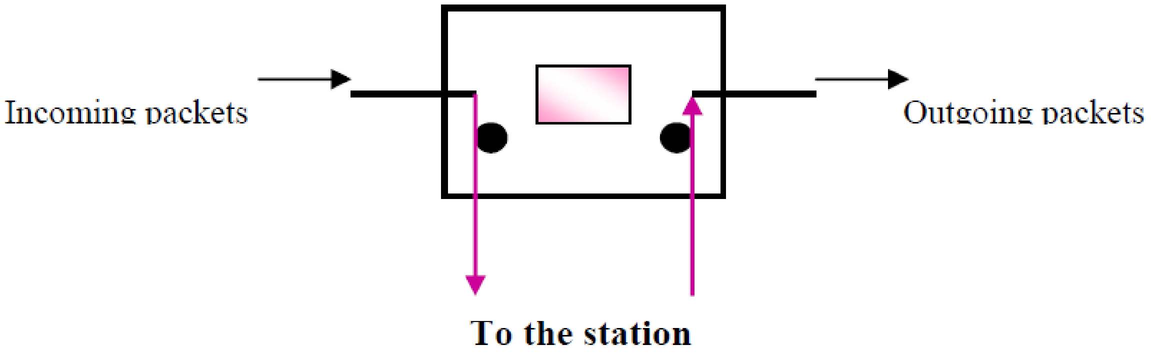
This topology is not very reliable, because when a link fails the entire ring connection is broken. But reliability can be improved by using wiring concentrator, which helps in bypassing a faulty node and somewhat is similar to star topology. Repeater works in the following three modes:

• Listen mode: In this mode, the station listens to the communication going over the shared medium as shown in Fig.(f).



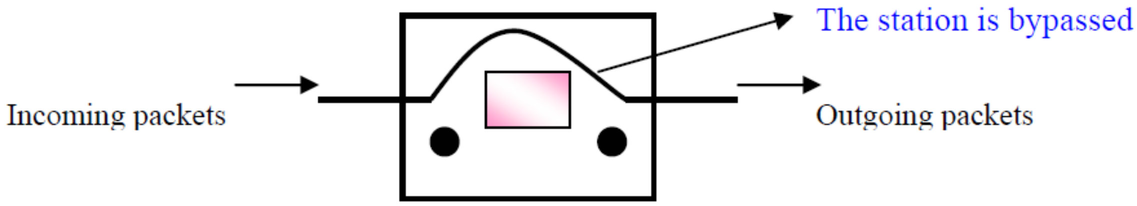
Fig(f) Repeater in Listen mode

• Transmit mode: In this mode the station transmit the data over the network as shown in Fig.(g)



Fig(g) Repeater in Transmit mode

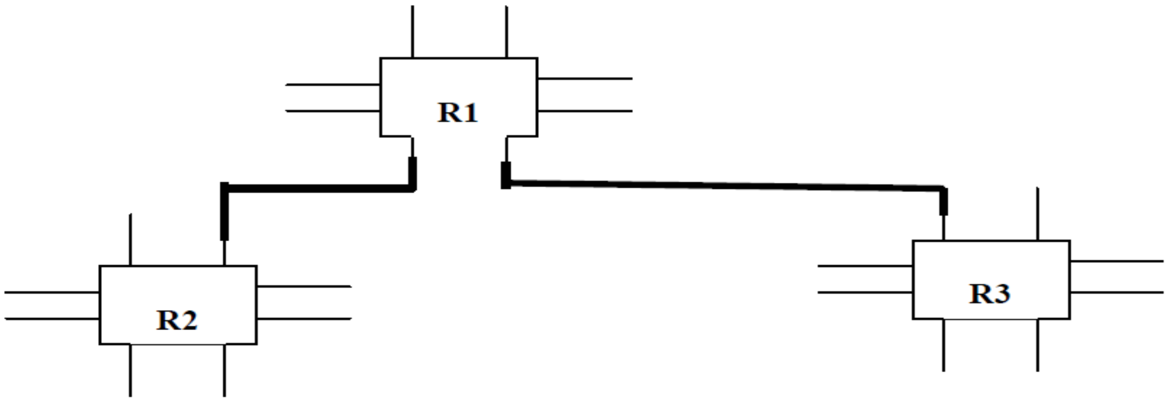
•By-Pass mode: When the node is faulty then it can be bypassed using the repeater in bypass mode, i.e. the station doesn’t care about what data is transmitted through the network, as shown in Fig.(h). In this mode there is no delay introduced because of this repeater.



Fig(h) Repeater in By-Pass mode

Tree Topology

This topology can be considered as an extension to bus topology. It is commonly used in cascading equipments. For example, you have a repeater box with 8-port, as far as you have eight stations, this can be used in a normal fashion. But if you need to add more stations then you can connect two or more repeaters in a hierarchical format (tree format) and can add more stations. In the Fig.(i), R1 refers to repeater one and so on and each repeater is considered to have 8-ports.

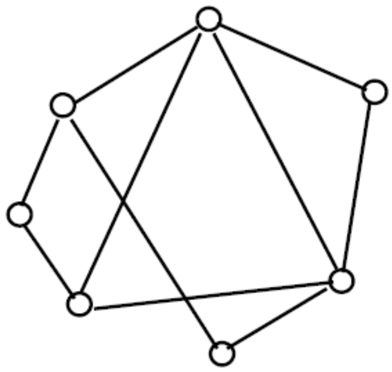


Fig(i) Tree Topology

This tree topology is very good in an organization as incremental expansion can be done in this way. Main features of this topology are scalability and flexibility. This is because, when the need arises for more stations that can be accomplished easily without affecting the already established network.

Unconstrained Topology

All the topologies discussed so far are symmetric and constrained by well-defined interconnection pattern. However, sometimes no definite pattern is followed and nodes are interconnected in an arbitrary manner using point-to-point links as shown in Fig 5.(j). Unconstrained topology allows a lot of configuration flexibility but suffers from the complex routing problem. Complex routing involves unwanted overhead and delay.



Fig(j) Unconstrained Topology

Choice of Topology

The choice of topology depends on a variety of factors, including reliability, expandability, and performance. This choice is part of the over- all task of designing a LAN and thus cannot be made in isolation, independent of the choice of transmission medium, wiring layout, and access control technique. A few general remarks can be made at this point.

There are four alternative media that can be used for a bus LAN:

• **Twisted pair**: In the early days of LAN development, voice-grade twisted pair was used to provide an inexpensive, easily installed bus LAN. A number of systems operating at 1 Mbps were implemented. Scaling twisted pair up to higher data rates in a shared-medium bus configuration is not practical, so this approach was dropped long ago.

• **Baseband coaxial cable**: A baseband coaxial cable is one that makes use of digital signaling.The original Ethernet scheme makes use of baseband coaxial cable.

• **Broadband coaxial cable**: Broadband coaxial cable is the type of cable used in cable television systems. Analog signaling is used at radio and television frequencies. This type of system is more expensive and more difficult to install and maintain than baseband coaxial cable.This approach never achieved popularity and such LANs are no longer made.

• **Optical fiber**: There has been considerable research relating to this alternative over the years, but the expense of the optical fiber taps and the availability of better alternatives have resulted in the demise of this option as well

Choice of Transmission Medium

The choice of transmission medium is determined by a number of factors. It is, we shall see, constrained by the topology of the LAN. Other factors come into play, including:

1. Capacity: to support the expected network traffic

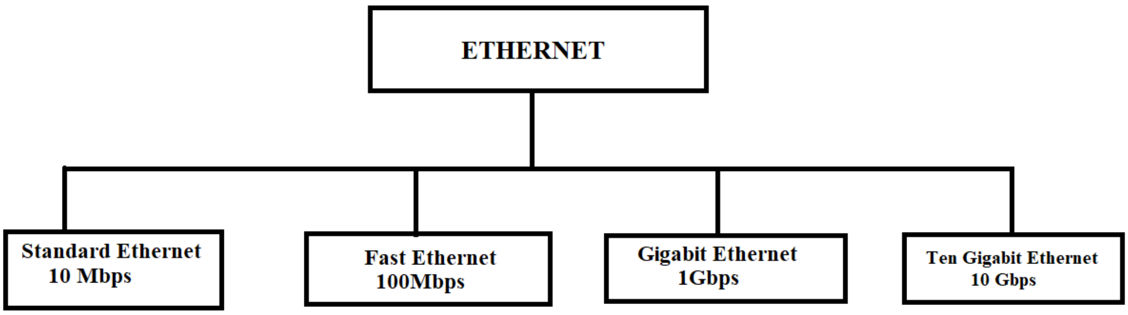
2. Reliability: to meet requirements for availability

3. Types of data supported: tailored to the application

4. Environmental scope: to provide service over the range of environments required

STANDARD ETHERNET

The original Ethernet was created in 1976 at Xerox's Palo Alto Research Center (PARC). Since then, it has gone through four generations: Standard Ethernet (lot Mbps), Fast Ethernet (100 Mbps), Gigabit ethernet (1 Gbps), and Ten-Gigabit Ethernet (10 Gbps), as shown in Figure (a).



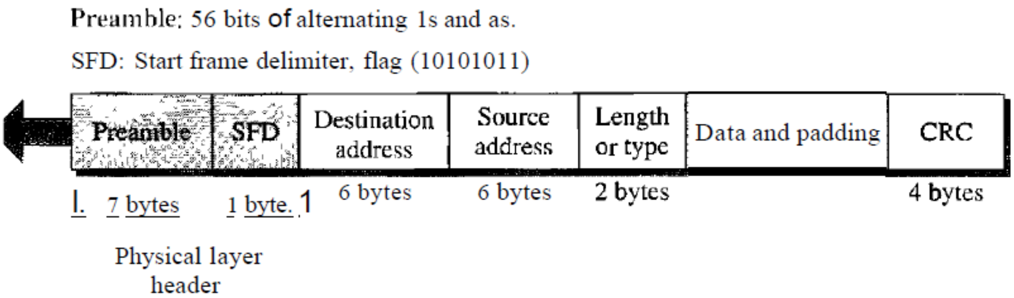
Fig(a) Ethernet Evolution through four Generations

MAC Sublayer

In Standard Ethernet, the MAC sublayer governs the operation of the access method. It also frames data received from the upper layer and passes them to the physical layer.

Frame Format

The Ethernet frame contains seven fields: preamble, SFD, DA, SA, length or type of protocol data unit (PDU), upper-layer data, and the CRe. Ethernet does not provide any mechanism for acknowledging received frames, making it what is known as an unreliable medium. Acknowledgments must be implemented at the higher layers. The format of the MAC frame is shown in Figure (b)



Fig(b) 802.3 Mac Frame

Preamble: The first field of the 802.3 frame contains 7 bytes (56 bits) of alternating Os and is that alerts the receiving system to the coming frame and enables it to synchronize its input timing. The pattern provides only an alert and a timing pulse. The 56-bit pattern allows the stations to miss some bits at the beginning of the frame. The preamble is actually added at the physical layer and is not (formally) part of the frame. Start frame delimiter (SFD). The second field (l byte: 10101011) signals the beginning of the frame. The SFD warns the station or stations that this is the last chance for synchronization. The last 2 bits is 11 and alerts the receiver that the next field is the destination address.
Destination address (DA): The DA field is 6 bytes and contains the physical address of the destination station or stations to receive the packet.
Source address (SA): The SA field is also 6 bytes and contains the physical address of the sender of the packet.
Length or type: This field is defined as a type field or length field. The original Ethernet used this field as the type field to define the upper-layer protocol using the MAC frame. The IEEE standard used it as the length field to define the number of bytes in the data field. Both uses are common today.
***Data*: This field carries data encapsulated from the upper-layer protocols. It is a minimum of 46 and a maximum of 1500 bytes,
CRC: The last field contains error detection information, in this case a CRC-32.

Physical Layer

The Standard Ethernet defines several physical layer implementations; four of the most common, are shown in Figure (c).

Encoding and Decoding

All standard implementations use digital signaling (baseband) at 10 Mbps. At the sender, data are converted to a digital signal using the Manchester scheme; at the receiver, the received signal is interpreted as Manchester and decoded into data. Manchester encoding is selfsynchronous, providing a transition at each bit interval.

lOBase5: Thick Ethernet

The first implementation is called 10BaseS, thick Ethernet, or Thicknet. The nickname derives from the size of the cable, which is roughly the size of a garden hose and too stiff to bend with your hands. lOBaseS was the first Ethernet specification to use a bus topology with an external transceiver (transmitter/receiver) connected via a tap to a thick coaxial cable. The transceiver is responsible for transmitting, receiving, and detecting collisions. The transceiver is connected to the station via a transceiver cable that provides separate paths for sending and receiving. This means that collision can only happen in the coaxial cable. The maximum length of the coaxial cable must not exceed 500 m, otherwise, there is excessive degradation of the signal. If a length of more than 500 m is needed, up to five segments, each a maximum of SOO-meter, can be connected using repeaters.

10Base2: Thin Ethernet

The second implementation is called lOBase2, thin Ethernet, or Cheapernet. IOBase2 also uses a bus topology, but the cable is much thinner and more flexible. The cable can be bent to pass very close to the stations. In this case, the transceiver is normally part of the network interface card (NIC), which is installed inside the station. Note that the collision here occurs in the thin coaxial cable. This implementation is more cost effective than 10BaseS because thin coaxial cable is less expensive than thick coaxial and the tee connections are much cheaper than taps. Installation is simpler because the thin coaxial cable is very flexible. However, the length of each segment cannot exceed 185 m (close to 200 m) due to the high level of attenuation in thin coaxial cable.

lOBase-T: Twisted-Pair Ethernet

The third implementation is called lOBase-T or twisted-pair Ethernet. 1OBase-T uses a physical star topology. The stations are connected to a hub via two pairs of twisted cable, Note that two pairs of twisted cable create two paths (one for sending and one for receiving) between the station and the hub. Any collision here happens in the hub. Compared to lOBaseS or lOBase2,since hub actually replaces the coaxial cable as far as a collision is concerned. The maximum length of the twisted cable here is defined as 100 m, to minimize the effect of attenuation in the twisted cable.

lOBase-F: Fiber Ethernet

Although there are several types of optical fiber lO-Mbps Ethernet, the most common is called 10Base-F. lOBase-F uses a star topology to connect stations to a hub. The stations are connected to the hub using two fiber optic cables

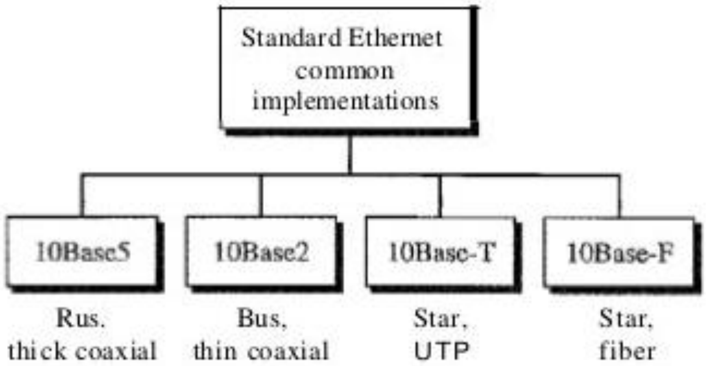


Fig (c) Categories of Standard Ethernet

Fast Ethernet

Fast Ethernet is a version of Ethernet with a 100Mbps data rate.Fast Ethernet was designed to compete with LAN protocols such as FDDI or Fiber channel. IEEE created fast Ethernet under the name 802.30. The 802.u or the fast Ethernet, as it is commonly known, was approved by the IEEE 802 Committee in June 1995. It may not be considered as a new standard but an addendum to the existing 802.3 standard.

MAC sublayer

It makes use of only star topology. Two different operations are possible: half duplex and full duplex. In half duplex stations are connected a hub. In full duplex connection are made via switch.

Frame Format

The frame format of fast Ethernet is same as that of the traditional Ethernet but uses a data transfer rate of 100 Mb/s instead of 10 Mb/s

Addressing

Addressing scheme of fast Ethernet is same as that of traditional Ethernet.

Access Method

The access method is also same i.e. CSMA/CD for the half duplex operation. For full duplex fast Ethernet, there is no need for CSMA/CD. The CSMA/CD is used for backward compatibility with the traditional Ethernet.

Auto-negotiation

It is the new feature added to fast Ethernet. It allows two devices to negotiate the mode or data or data rate of operation. It allows two incompatible devices to connect to each other.

Physical Layer functions

If two stations are there, then point to point topology is used. If multiple stations are connected then Start topology with a hub or switch is used. IEEE has designed two categories of Fast Ethernet: 100Base-X and 100Base-T4. 100Base-X uses two-wire interface between a hub and a station while 100Base-T4 uses four-wire interface. 100-Base-X itself is divided into two: 100Base-TX and 100base-FX as shown in Fig below.

100 BASE TX: This option uses two pairs of category 5 UTP or two shielded twisted-pair (STP) cable to connect a station to hub. One pair is used to carry frames from the hub to the station and other to carry frames from station to hub.

100 BASE FX: This option uses two Fiber optic cables, one carry frames from station to hub and other from hub to station.

GIGABIT ETHERNET

As applications increased, the demand on the network, newer, high-speed protocols such as FDDI and ATM became available. However, in the last couple of years, Fast Ethernet has become the backbone of choice because it's simplicity and its reliance on Ethernet. The primary goal of Gigabit Ethernet is to build on that topology and knowledge base to build a higher-speed protocol without forcing customers to throw away existing networking equipment. In March 1996, the IEEE 802.3 committee approved the 802.3z Gigabit Ethernet Standardization project. At that time as many as 54 companies expressed there intent to participate in the standardization project. The Gigabit Ethernet Alliance was formed in May 1996 by 11 companies. The Alliance represents a multi-vendor effort to provide open and inter-operable Gigabit Ethernet products. The objectives of the alliance are:

• Supporting extension of existing Ethernet and Fast Ethernet technology in response to demand for higher network bandwidth.
• Developing technical proposals for the inclusion in the standard
• Establishment of inter-operability test procedures and processes

Gigabit Ethernet provides the data rate of 1Gbps or 1000Mbps. IEEE created Gigabit Ethernet under the name 802.32.It is compatible with standard or fast Ethernet. It also uses 48 bit hexadecimal addressing scheme. The frame format is also similar to standard Ethernet. It operates in both half-duplex and full duplex mode. In half- duplex mode, CSMA/CD access method is used whereas in full duplex mode CSMA/CD is not required.

(i)The cabling requirement of gigabit Ethernet is very different. The technology is based on fiber optic cable. Multi-mode fiber is able to transmit at gigabit rate to at least 580 meters and with single-mode runs exceeding 3 km. Fiber optic cabling is costly. In order to reduce the cost of cabling, the 802.3z working group also proposed the use of twistedpair or cable or coaxial cable for distances up to 30 meters.
(ii) Gigabit Ethernet also relies on a modified MAC layer. At gigabit speed, two stations 200 meters apart will not detect a collision, when both simultaneously send 64-byte frames. This inability to detect collision leads to network instability. A mechanism known as carrier extension has been proposed for frames shorter than 512 bytes. The number of repeater hops is also restricted to only one in place of two for 100 Base-T.
(iii) Flow Control is a major concern in gigabit Ethernet because of buffer overflow and junked frames in heavily loaded condition. The solution proposed by IEEE subcommittee is the 802.3x. The X-on/X-off protocol works over any full-duplex Ethernet, fast Ethernet or gigabit Ethernet link. When a switch buffer is close to capacity, the receiving device signals the sending station and tells it to stop transmitting until the buffer becomes empty.

(iv) Finally, one important feature, which Ethernet technology lacks, is the Quality of Service (QoS). The gigabit Ethernet is a connectionless technology that transmits variable length frames. As such, it simply cannot guarantee that the real-time packets get the preferential treatment they require. The IEEE subcommittee developed two specifications that will help Ethernet provide the required QoS. 802.lq tags traffic for VLANs and for prioritization. 802.lp is a signaling scheme that lets end station request priority and allows switches to pass these requests along the path. The gigabit Ethernet comes into its own as an internetworking switch link (ISL) that aggregates 10-and100-Mb/s feeds from the desktops and servers. Presently, gigabit Ethernet is already matured with a large installation base as a backbone network technology.

MAC Sublayer

MAC sublayer remains almost same in Gigabit Ethernet. There are two distinct approaches for medium access: Half duplex and Full duplex.

Physical Layer

Two stations can be connected in Point-to-Point fashion and multiple stations can be connected in a star topology with a switch or hub.

The cabling used is 1000 BASE X(2 wire) and 1000 BASE-T(4 wire).

TEN GIGABIT ETHERNET

IEEE created Ten Gigabit Ethernet under the name 802.3ae. It provides the data rate of 10Gbps. It is compatible with standard, Fast and Gigabit Ethernet. It uses 48 bit hexadecimal addressing scheme. The frame format is same as that of standard Ethernet. It allows interconnection of existing LAN's into Metropolitian area network(MAN) and Wide are network(WAN).

MAC Sublayer

Ten Gigabit Ethernet operates only in Full duplex mode which means there is no need for contention. Therefore CSMA/CD is not used in it.

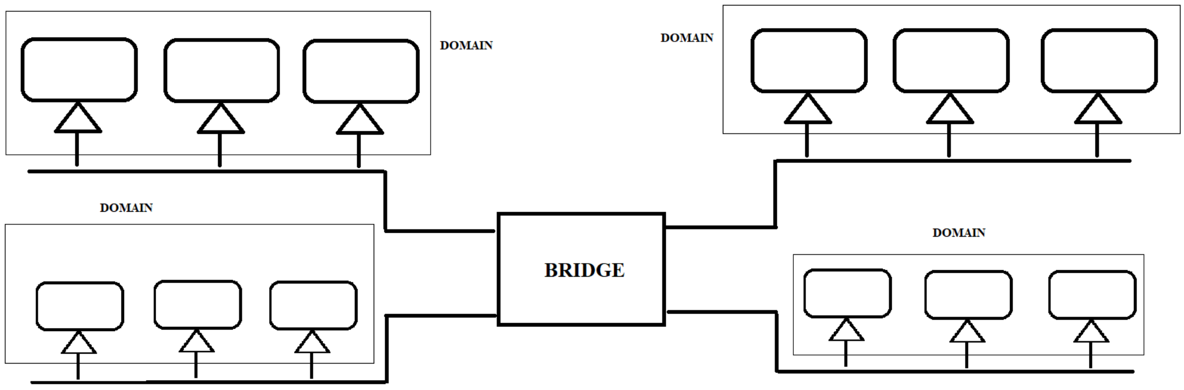
Physical Layer

There are three basic implementations of Ten Gigabit Ethernet: 10GBase-5, 10GBase L and 10GBase-E.

Other Implementations of Ethernet

1. Bridged Ethernet

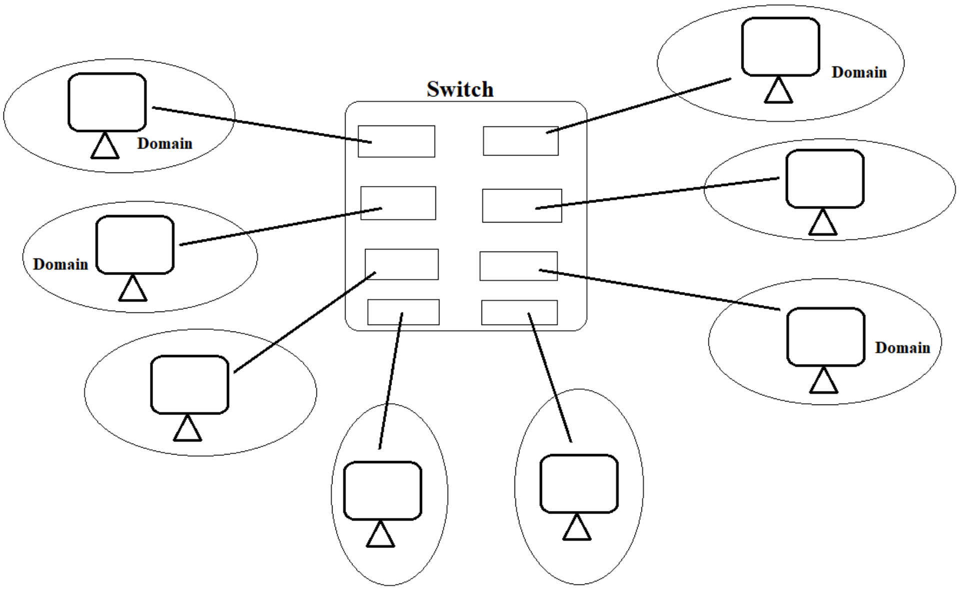
Bridged Ethernet is the Ethernet in which the LAN is divided using a bridge. A bridge is a networking device that is used to divide a network into two or more network segments such that each network (segment) is independent. When such a Ethernet is created the bandwidth of individual segments is increased. Also the collision domains are separated. In such networks collision domain becomes much smaller and probability of collision is reduced.



BRIDGED ETHERNET

2. Switched Ethernet

In switched Ethernet a switch having N ports is used. To this switch, N stations are connected as:

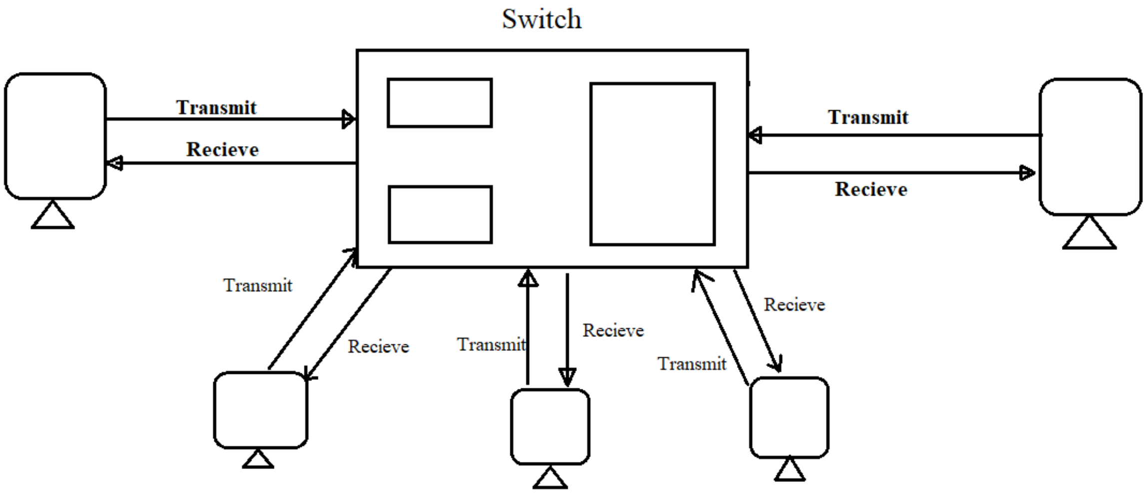


SWITCHED ETHERNET

In such a case, the bandwidth is shared only between the station and the switch. The collision domain is also divided into N domains.

3. Full Duplex Ethernet

Full Duplex Ethernet is further extension of switched Ethernet. It makes use of switch. All the stations are connected to this switch. In this mode, the stations support full duplex operations i.e. they can send and receive simultaneously. Full duplex mode increases the capacity of each domain from 10 to 20 Mbps. Such a system uses to links: one to transmit and one to receive. In full duplex Ethernet there is no need for CSMA/CD as each station is connected to switch via two separate links. Each station can send and receive independently without collisions.



Full Duplex Ethernet

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