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Chapter No: 1

Introduction to Data Communication

Introduction:

Data is defined as information which is stored in the digital form.

Data communication is the process of transferring digital information between two points.

Data can be alphabets, numeric or symbols and it consists of any one or the combination of the following:

Microprocessor op-codes, control codes, user addresses, program data or data base information.

At the source or destination the data are in digital form but during the transmission it may be analog or digital.

A data communication network can be simply consisting of two computers connected to each other a public telecommunication network.

Data Communications

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

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

Q. Define data communications.

Data communications are the exchange of data between two devices via some form of transmission medium such as a wire cable. For data communications to occur, the communicating devices must be part of a communication system made up of a combination of hardware (physical equipment) and software (programs).

Q. Describe the characteristics of data communication system.

Characteristics of Data Communication System:

The effectiveness of a data communications system depends on four **fundamental characteristics**: **Delivery**, **Accuracy**, **Timeliness**, and **Jitter**.

1. **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.

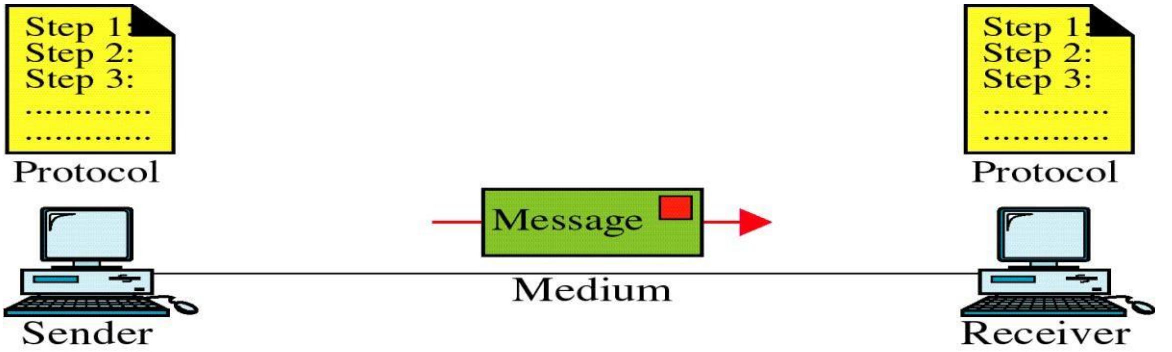
3. **Timeliness:** The system must deliver data in a **timely manner**. Data delivered late are useless. In the case of video and audio, timely delivery means delivering data as they are produced, in the same order that they are produced, and without significant delay. This kind of delivery is called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

3D Ms. If some of the packets arrive with 3D-ms delay and others with 4D-ms delay, an uneven quality in the video is the result.

Q. Draw the components of data communication systems and state the function of each block.

Components of Data Communications System



A data communications system has five components:

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

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

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

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

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

Q. Explain Simplex, Half Duplex and Full Duplex communication with examples.

Communication Modes

Based on whether the system communicates only in one direction or otherwise, the communication systems are classified as

- Simplex systems
- Half duplex systems
- Full duplex systems

1. Simplex Systems

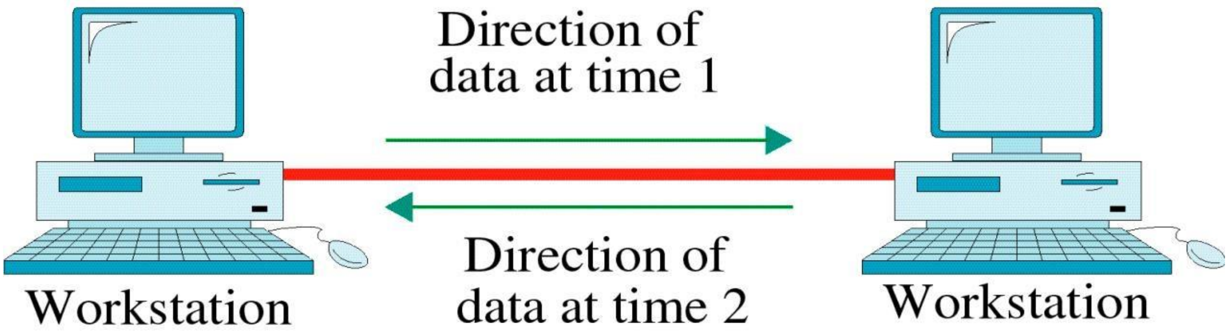
- In **Simplex** mode, the communication is unidirectional, as on a one-way street.
- Only one of the two devices on a link can transmit; the other can only receive.

- **Keyboards** and traditional **monitors** are examples of simplex devices. The keyboard can only introduce input; the monitor can only accept output.
- The simplex mode can use the entire capacity of the channel to send data in one direction.

Simplex mode of Communication

2. Half Duplex Systems

- In **Half-duplex** mode, each station can both transmit and receive, but not at the same time. :
- When one device is sending, the other can only receive, and vice versa.
- The half-duplex mode is like a one-lane road with traffic allowed in both directions. When cars are traveling in one direction, cars going the other way must wait. In a half-duplex transmission, the entire capacity of a channel is taken over by whichever of the two devices is transmitting at the time. **Walkie-talkies** and **CB (citizens band) radios** are both half-duplex systems.
- The half-duplex mode is used in cases where there is no need for communication in both directions at the same time; the entire capacity of the channel can be utilized for each direction.



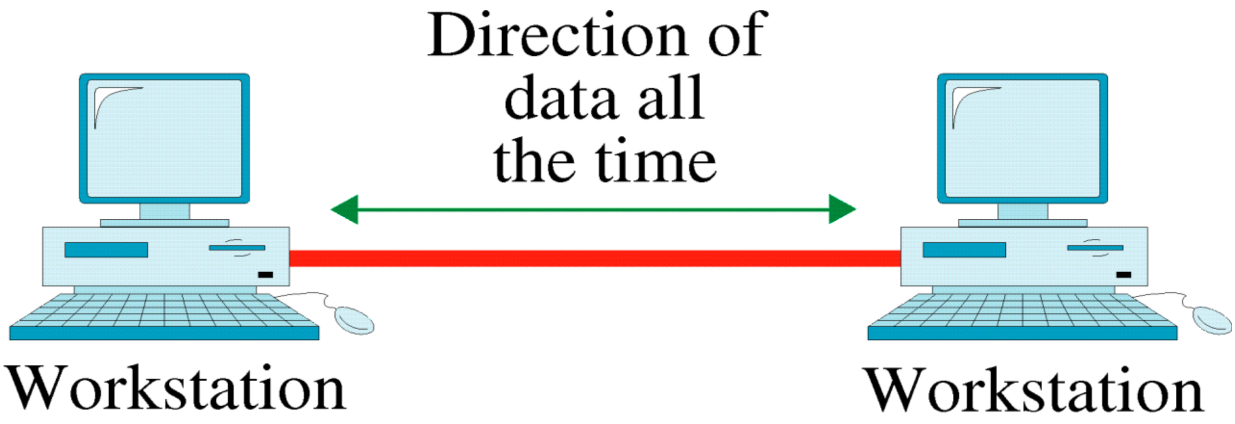
Half Duplex mode of Communication

3. Full-Duplex

- In full-duplex mode (also called duplex), both stations can transmit and receive simultaneously.
- The full-duplex mode is like a two-way street with traffic flowing in both directions at the same time. In full-duplex mode, signals going in one direction share the capacity of the link: with signals going in the other direction. This sharing can occur in two ways: Either the link must contain two physically separate transmission paths, one for sending and the other for

receiving; or the capacity of the channel is divided between signals traveling in both directions.

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



Full Duplex mode of Communication

Q. Define Protocols. Explain key elements of protocols.

Protocol:

- A protocol is a set of rules that govern data communications. A protocol defines what is communicated, how it is communicated, and when it is communicated.
- In computer networks, communication occurs between entities in different systems.
- An entity is anything capable of sending or receiving information. However, two entities cannot simply send bit streams to each other and expect to be understood. For communication to occur, the entities must agree on a protocol.

The key elements of a protocol are: syntax, semantics, and timing.

1) Syntax: (what is to be communicated?)

- The term *syntax* refers to the structure or format of the data, meaning the order in which they are presented.
- For example, a simple protocol might expect the first 8 bits of data to be the address of the sender, the second 8 bits to be the address of the receiver, and the rest of the stream to be the message itself.

2) Semantics: (how it is to be communicated)

- The word *semantics* refers to the meaning of each section of bits.
- How is a particular pattern to be interpreted, and what action is to be taken based on that interpretation? For example, does an address identify the route to be taken or the final destination of the message?

3) Timing: (when it should be communicated)

- The term *timing* refers to two characteristics: when data should be sent and how fast they can be sent. For example, if a sender produces data at 100 Mbps but the receiver can process data at only 1 Mbps, the transmission will overload the receiver and some data will be lost.

Q. Define Standard. Name any four Standard Organizations. Give their functions

Standards:

- Standards provide guidelines to manufacturers, vendors, government agencies, and other service providers to ensure the kind of interconnectivity necessary in today's marketplace and in international communications.

- Standards are essential in creating and maintaining an open and competitive market for equipment manufacturers and in guaranteeing national and international interoperability of data and telecommunications technology and processes.

- Data communication standards fall into two categories: *de facto* (meaning "by fact" or "by convention") and *de jure* (meaning "by law" or "by regulation").

1) De facto:

- Standards that have not been approved by an organized body but have been adopted as standards through widespread use are de facto standards.
- De facto standards are often established originally by manufacturers who seek to define the functionality of a new product or technology.

2) De jure:

- Those standards that have been legislated by an officially recognized body are de jure standards.

Standards Organizations:

- Standards are developed through the cooperation of standards creation committees, forums, and government regulatory agencies.

Standards Creation Committees:

- While many organizations are dedicated to the establishment of standards, data telecommunications in North America rely primarily on those published by the following:

1) International Organization for Standardization

- **(ISO):** The ISO is a multinational body whose membership is drawn mainly from the standards creation committees of various governments throughout the world.
- The ISO is active in developing cooperation in the realms of scientific, technological, and economic activity.

2) International Telecommunication Union-Telecommunication Standards Sector (ITU-T):

- By the early 1970s, a number of countries were defining national standards for telecommunications, but there was still little international compatibility.
- The United Nations responded by forming, as part of its International Telecommunication Union (ITU), a committee, the Consultative Committee for International Telegraphy and Telephony (CCITT).
- This committee was devoted to the research and establishment of standards for telecommunications in general and for phone and data systems in particular. On March 1, 1993, the name of this committee was changed to the International Telecommunication Union - Telecommunication Standards Sector (ITU-T).

3) American National Standards Institute (ANSI):

- The American National Standards Institute is a completely private, nonprofit corporation not affiliated with the U.S. federal government. However, all ANSI activities are undertaken with the welfare of the United States and its citizens occupying primary importance.

4) Institute of Electrical and Electronics Engineers (IEEE):

- The Institute of Electrical and Electronics Engineers is the largest professional engineering society in the world.
- International in scope, it aims to advance theory, creativity, and product quality in the fields of electrical engineering, electronics, and radio as well as in all related branches of engineering.
- As one of its goals, the IEEE oversees the development and adoption of international standards for computing and communications.

5) Electronic Industries Association (EIA):

- The Electronic Industries Association is a nonprofit organization devoted to the promotion of electronics manufacturing concerns.
- Its activities include public awareness education and lobbying efforts in addition to standards development.
- In the field of information technology, the EIA has made significant contributions by defining physical connection interfaces and electronic signaling specifications for data communication.

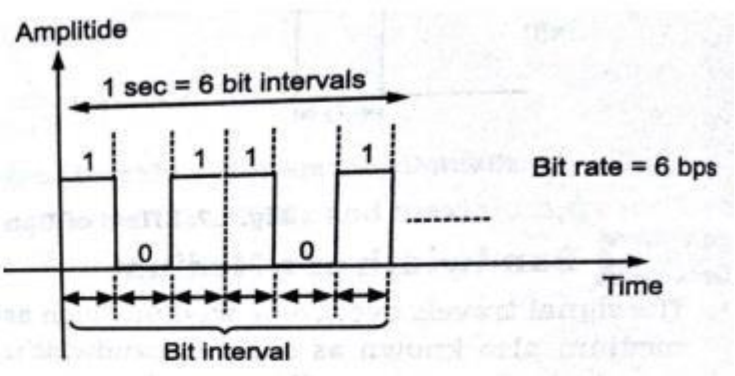
Bandwidth, Data Transmission Rate, Baud Rate and Bits Per Second

Bandwidth is measured as the amount of data that can be transferred from one point to another within a network in a specific amount of time. Typically, bandwidth is expressed as a bitrate and measured in bits per second (bps).

The term bandwidth refers to the transmission capacity of a connection and is an important factor when determining the quality and speed of a network or the internet connection.

Definition of Bit Rate

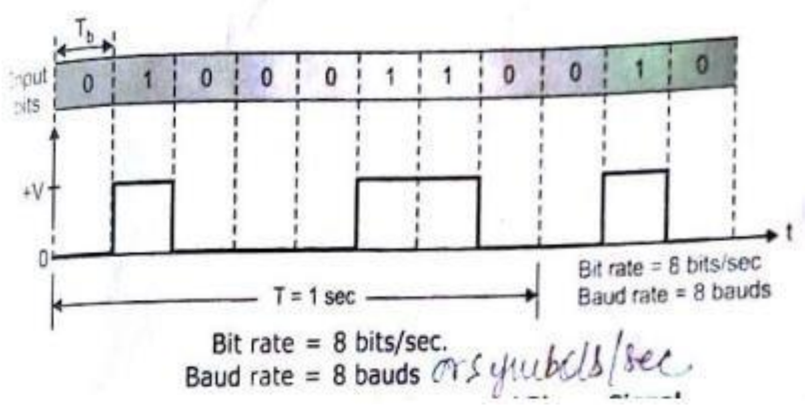
Bit rate can be defined as the number of bit intervals per second. And bit interval is referred to as the time needed to transfer one single bit. In simpler words, the bit rate is the number of bits sent in one second, usually expressed in bits per second (bps). For example, kilobits per second (Kbps), Megabits per second (Mbps), Gigabits per second (Gbps), etc.



Definition of Baud Rate

Baud rate is defined as the number of signal units per second. It is always less than or equal to bit rate. It is represented as bauds or symbols/second.

Baud rate is expressed in the number of times a signal can change on transmission line per second. Usually, the transmission line uses only two signal states, and make the baud rate equal to the number of bits per second that can be transferred.



An example can illustrate it. For example, 1500 baud rate illustrates that the channel state can alter up to 1500 times per second. The meaning of changing state means that channel can change its state from 0 to 1 or from 1 to 0 up to 1500 times per second (in the given case).

- bit: a unit of information
- baud: a unit of signaling speed.
- Bit rate:b
  - Number of bits transmitted per second.
- Baud Rate:s
  - Number of symbols transmitted per second.
- General formula:
  - **b=s*n**
  - Where n is number of bits per symbol.

Key Differences Between Bit Rate and Baud Rate

1. Bit rate is the number bits (0’s and 1’s) transmitted per second.
   On the other hand Baud rate is the number of times a signal is traveling comprised of bits.

2. Baud rate can determine the **bandwidth** of the channel or its required amount to send the signal while through Bit rate it is not possible.Bit Rate can be expressed by the given equation:
   Bit rate = baud rate x the number of bits per signal unit
   In contrary Baud rate is expressed in the given equation:
   Baud rate = bit rate / the number of bits per signal unit

Question: Calculate the baud rate for the given bit rate and type of modulation:
(i) 5000 bps, ASK (ii) 4000 bps, FSK
Answer:

For baud rate (S), we know that the formula is:
S=N/r.
N = S * r
Here, N is Bit rate, S is the Baud rate
r = number of bits in signal elements
So, at first we need to calculate r for each case.
We know, r = $\log_2 L$.

i)For ASK, r = $\log_2 2$ = 1
S=5000 bps/1=5000 baud
ii) For FSK, r = $\log_2 2$ = 1
S=4000bps/1=4000 baud

Question: A signal carries five bits in each signal element. If 1600 signal elements are sent per second, find the baud rate and bit rate in kbps.
Answer:

Baud rate is number of signal elements per second.
Bit rate is the number of bits per second.
We also know that S=N/r where S is the baud rate, N is the bit rate and r is the bits in each signal element.
In this case 1600 signal elements are sent per second.
So baud rate is 1600.
Now S=1600,r=5 and N is unknown.
So N=S*r=1600*5=8000 bps or 8 kbps.
Therefore the bit rate is 8kbps

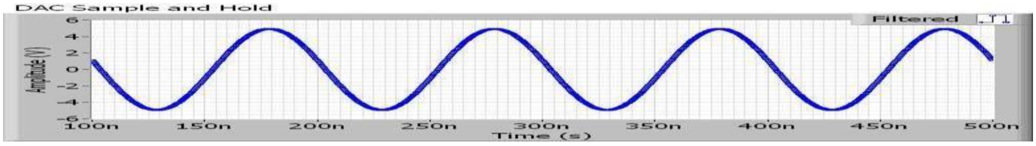
BASIS FOR COMPARISON	BIT RATE	BAUD RATE
Basic	Bit rate is the count of bits per second.	Baud rate is the count of signal units per second.
Meaning	It determines the number of bits traveled per second.	It determines how many times the state of a signal is changing.
Term usually used	While the emphasis is on computer efficiency.	While data transmission over the channel is more concerned.
Bandwidth determination	Can not determine the bandwidth.	It can determine how much bandwidth is required to send the signal.
Equation	Bit rate = baud rate x the count of bits per signal unit	Baud rate = bit rate / the number of bits per signal unit

Analog Signal and Digital Signal

Analog Signal

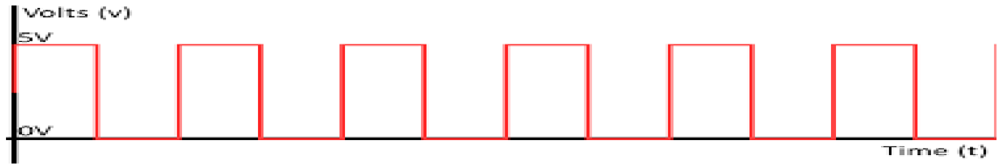
An **analog signal** is a continuous wave denoted by a sine wave (pictured below) and may vary in signal strength (amplitude) or frequency (waves per unit time). The sine wave's amplitude value can be seen as the higher and lower points of the wave, while the frequency value is measured in the sine wave's physical length from left to right.

There are many examples of analog signals around us. The sound from a human voice is analog, because sound waves are continuous, as is our own vision, because we see various shapes and colors in a continuous manner due to light waves. Even a typical kitchen clock having its hands moving continuously can be represented as an analog signal.



Digital Signal

A **digital signal** - a must for computer processing - is described as using binary (0s and 1s), and therefore, cannot take on any fractional values. As illustrated in the graphic below, digital signals retain a uniform structure, providing a constant and consistent signal. Because of the inherent reliability of the digital signal, technology using it is rapidly replacing a large percentage of analog applications and devices. For example, the wristwatch, showing the time of day, with its minute, hour, and sweeping second hands, is being replaced by the digital watch, which offers the time of day and other information using a numerical display. A typical digital signal is represented below. Note the equally dispersed 1s and 0s.



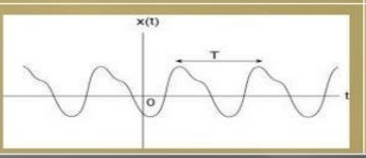
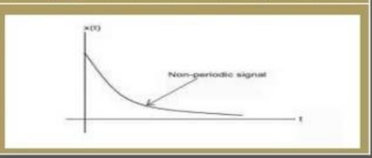
Analog Signal	Digital Signal
An analog signal signifies a continuous signal that keeps changes with a time period.	A digital signal signifies a discrete signal that carries binary data and has discrete values.
Analog signals are continuous sine waves	Digital signal is square waves.
Analog signals describe the behavior of the wave with respect to amplitude, time period, & phase of the signal.	Digital signals describe the behavior of the signal with respect to the rate of a bit as well as bit interval.
Analog signal range will not be set.	Digital signal is limited as well as ranges from 0 to 1.
Analog signal is further horizontal toward distortion during the response to noise	A digital signal has resistance in response toward the noise, therefore, it does not often face distortion.
An analog signal broadcasts the information in the signal form.	A digital signal broadcasts the information in the form of binary that is bits.
The example of an analog signal is the human voice	The example of a digital signal is the data transmission in a computer.

Periodic and Non-periodic signals

- A signal is periodic signal if it completes a pattern within measurable time frame.
- A periodic signal is characterised by **amplitude, frequency and phase.**
- Mathematically: $v(t)=V \sin(2\pi ft+\theta)$
    - V:Peak Amplitude
    - F:frequency
    - t:Time(seconds)
    - θ:Phase(degree or radians)
- **Amplitude** is the highest height of the signal, maximum value or strength of the signal over time; typically, this value is measured in volts.
- **frequency** is the rate [in cycles per second, or Hertz (Hz)] at which the signal repeats., and
- **Phase** is a measure of the relative position in time within a single period of a signal
- An analog signal is not resistant toward the noise, therefore; it faces distortion as well as reduces the transmission quality.

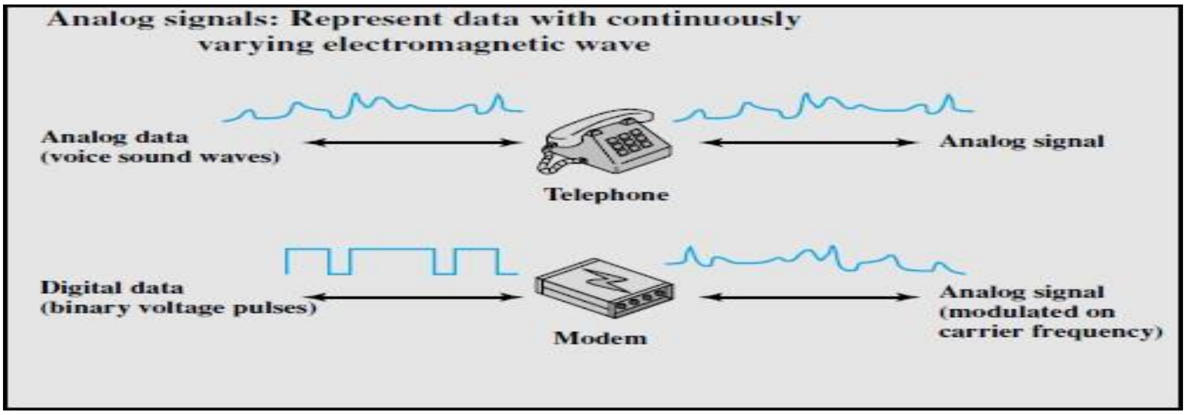
Non-periodic signals

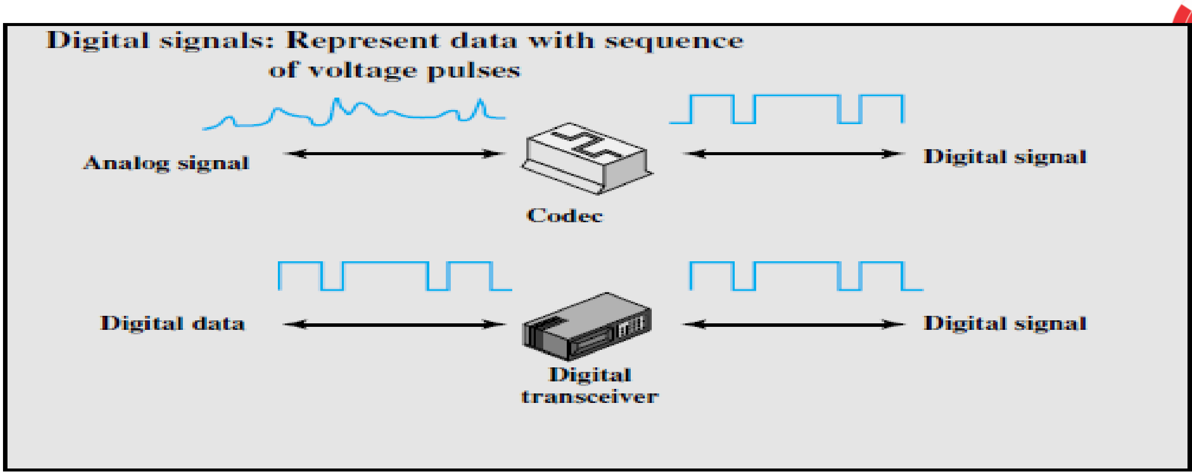
- A signal that does not repeats its pattern over a period is called aperiodic signal or non periodic.
- Both the Analog and Digital can be periodic or aperiodic: but in data communication **periodic analog signals and aperiodic digital** signals are used.

Periodic Signal	Aperiodic Signal
❑ A signal which repeats itself after a specific interval of time is called periodic signal.	❑ A signal which does not repeat itself after a specific interval of time is called aperiodic signal.
❑ A signal that repeats its pattern over a period is called periodic signal	❑ A signal that does not repeats its pattern over a period is called aperiodic signal or non periodic.
❑ They can be represented by a mathematical equation	❑ They cannot be represented by any mathematical equation
❑ Their value can be determined at any point of time	❑ Their value cannot be determined with certainty at any given point of time
❑ They are deterministic signals	❑ They are random signals
❑ Example: sine cosine square sawtooth etc	❑ Example: sound signals from radio , all types of noise signals
❑ Figure: 	❑ Figure: 

Analog and Digital data

- Analog data take on continuous values in time interval.
- For example, voice and video are continuously varying patterns of intensity. Most data collected by sensors, such as temperature and pressure, are continuous valued.
- The most familiar example of analog data is **audio**, which, in the form of acoustic sound waves, can be perceived directly by human beings.
- Digital data take on discrete values; examples are **text and integers**.
- They cannot be easily stored or transmitted by data processing and communications systems in character form.
- Morse code, **International Reference Alphabet (IRA)** are used to translate text into binary.





Analog 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.
- 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.

Digital transmission

- **Digital transmission**, in contrast, assumes a binary content to the signal.
- A digital signal can be transmitted only a limited distance before attenuation.
- 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.

(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.

Both analog and digital information can be encoded as either analog or digital signals. The particular encoding that is chosen depends on the specific requirements to be met and **the media and communications facilities** available.

1. **Digital data, digital signals(Digital data Transmission):**

   - The simplest form of digital encoding of digital data is to assign one voltage level to binary one and another to binary zero.

   - More complex encoding schemes are used to improve performance, by altering the spectrum of the signal.

2. **Digital data, analog signal:**

   - A modem converts digital data to an analog signal so that it can be transmitted over an analog line.

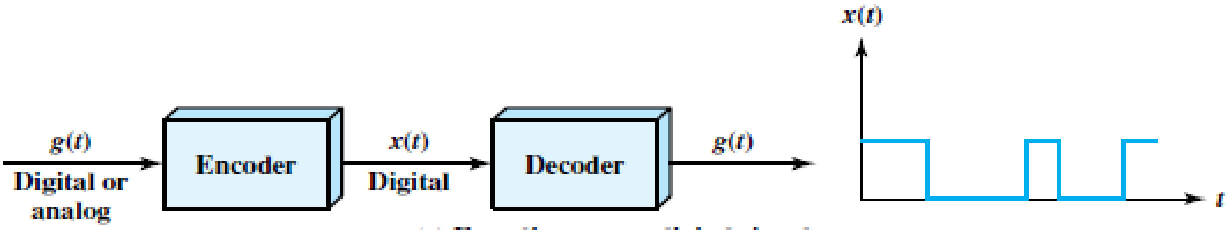
- The basic techniques are **amplitude shift keying (ASK), frequency shift keying (FSK), and phase shift keying (PSK)**.
- All involve altering one or more characteristics of a carrier frequency to represent binary data.

3. **Analog data, digital signals:**

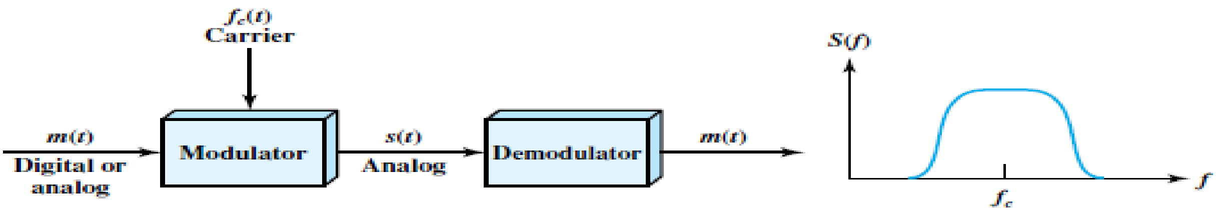
- Analog data, such as voice and video, are often digitized to be able to use digital transmission facilities.
- The simplest technique is **pulse code modulation (PCM)**, which involves sampling the analog data.

4. **Analog data, analog signals:**

- Analog data are modulated by a carrier frequency to produce an analog signal, which can be utilized on an analog transmission system.
- The basic techniques are **amplitude modulation (AM), frequency modulation(FM), and phase modulation (PM).**



(a) Encoding onto a digital signal



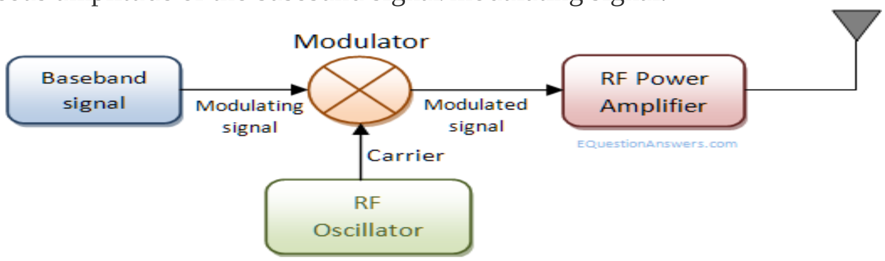
(b) Modulation onto an analog signal

- ## Modulation:

Now we have to develop some way to send the information of message signal via this carrier signal. The carrier signal is a high frequency sinusoidal signal represented by amplitude, frequency and phase. We can vary one of this parameter accordingly with the message information.

- ## What is Modulation?

Modulation is an operation of varying amplitude or frequency or phase of carrier signal according to the instantaneous amplitude of the baseband signal/modulating signal.



Modulation Block Diagram

Here baseband signals comes from a audio/video or computer. Baseband signals are also called modulating signal as it modulates carrir signal. Carrier signals are high frequecy radio waves it generally comes from a radio frequency oscillators. These two signls are combined in modulator. Modulator takes the instantenious amplitute of baseband signal and varies amplitute/frequency/phase of carrier signal. Resultant signal is a modulated signal. It goes to an RF-amplifier for signal power boosting and then feed to antenna or a co-axial cable.

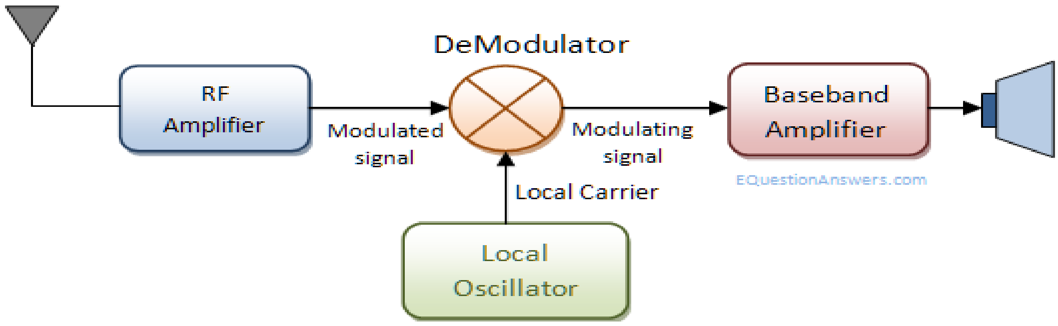
There are two types of modulation analog and digital. Analog modulation delas with the voice, video and regular waves of base band signals. Where as digital modulations are with bit streams or symbols from computing vevices as base band signals.

- **DeModulation:**

Demodulation is the opposite process of modulation. Modulator is a part of signal transmitter where as demodulator is the receiving side. In broadcast system radio transmitting station does to modulation part. A radio receiver acts as a demodulator. A modem receives signals and also transmits signals thus it does modulation and demodulation at the same time. Thus the name modem has been given. A radio antenna receives low power signal. A co-axial cable end point can also take as an signal input. An RF amplifier boosts the signal amplitude. Then the signal goes to a demodulator. demodulator does the reverse of modulation and extracts the backband signal from carrier. Then the base band signal is amplified to feed a audio speaker or video monitor or TTL/CMOS signal levels to match computer inputs.

- **What is De-modulation?**

Demodulation is the opposite process of modulation where the varying amplitude, frequency or phase of carrier signal is extracted to construct the original the message signal.



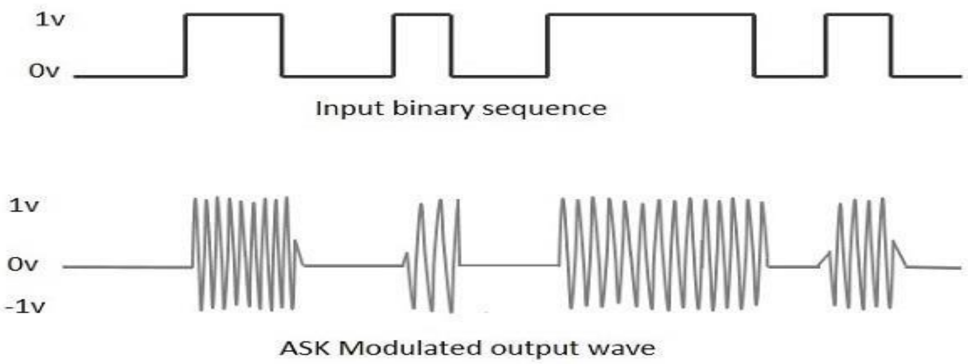
Demodulation Block Diagram

Digital to Analog conversion

- The case of transmitting digital data using analog signals.
- The most familiar use is for transmitting digital data through the public telephone network.
- The telephone network was designed to receive, switch, and transmit analog signals in the voice-frequency range of about 300 to 3400 Hz.
- It is not at present suitable for handling digital signals from the subscriber locations.
- Thus digital devices are attached to the network via a **modem (modulator-demodulator)**, which converts digital data to analog signals, and vice versa.
- Modulation involves operation on one or more of the three characteristics of a carrier signal: **amplitude, frequency, and phase.**
- Accordingly, there are three basic encoding or modulation techniques for transforming digital data into analog signals:
    - Amplitude Shift Keying (ASK),
    - Frequency Shift Keying (FSK), and
    - Phase Shift Keying (PSK).

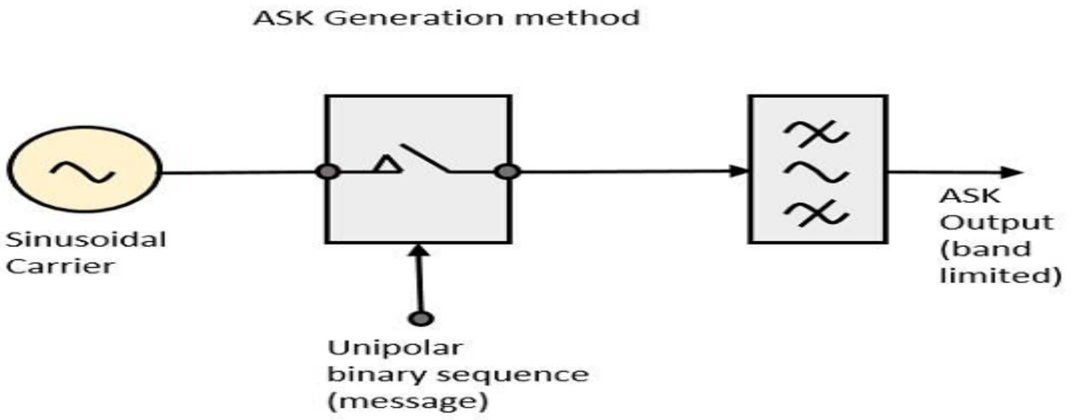
1. Amplitude Shift Keying (ASK)

- ASK is the digital carrier Modulation in which amplitude of carrier will take one of the two values in response to 0 or 1 value of digital data.
- **Amplitude Shift Keying (ASK)** is a type of Amplitude Modulation which represents the binary data in the form of variations in the amplitude of a signal.
- Any modulated signal has a high frequency carrier. The binary signal when ASK modulated, gives a **zero** value for **Low** input while it gives the **carrier output** for **High** input.
- The following figure represents ASK modulated waveform along with its input.



ASK Modulator

The ASK modulator block diagram comprises of the carrier signal generator, the binary sequence from the message signal and the band-limited filter. Following is the block diagram of the ASK Modulator.

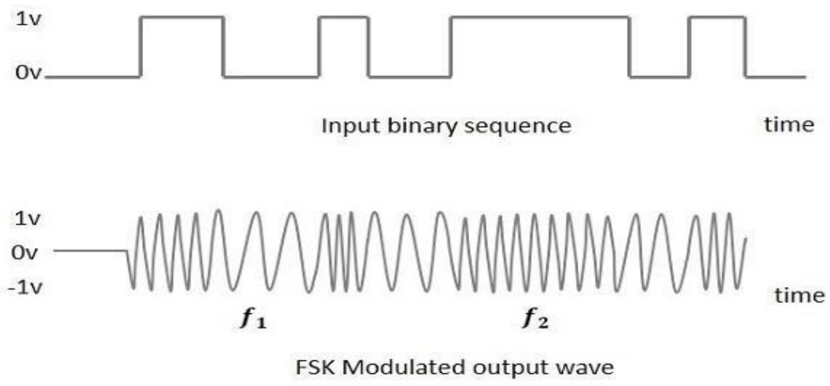


Application:

1. Used in our infrared remote controls
2. Used in fibre optical tranmitter and receiver.

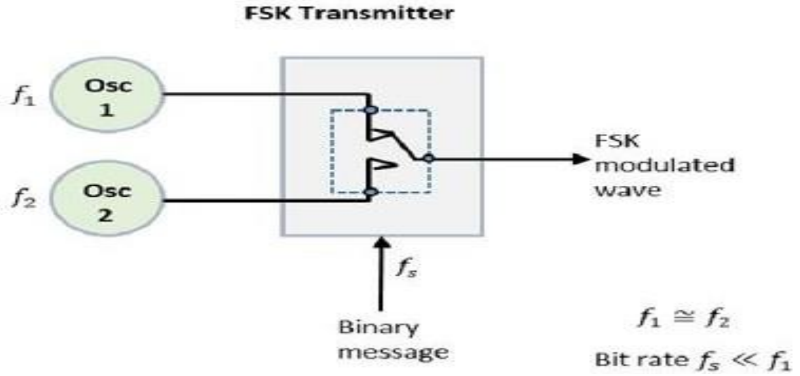
2. Frequency Shift Keying (FSK)

- **Frequency Shift Keying (FSK)** is the digital modulation technique in which the frequency of the carrier signal varies according to the digital signal changes. FSK is a scheme of frequency modulation.
- The output of a FSK modulated wave is high in frequency for a binary High input and is low in frequency for a binary Low input. The binary **1s** and **0s** are called Mark and Space frequencies.
- The following image is the diagrammatic representation of FSK modulated waveform along with its input.



FSK Modulator

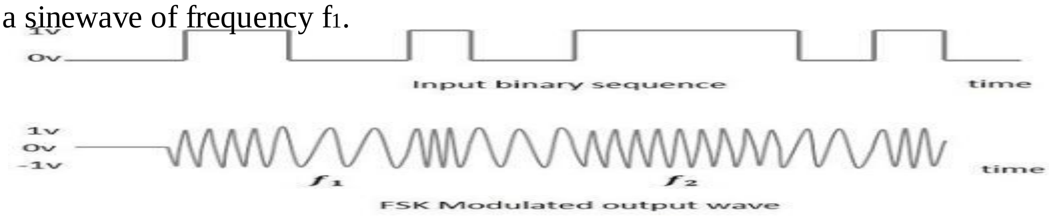
The FSK modulator block diagram comprises of two oscillators with a clock and the input binary sequence. Following is its block diagram.



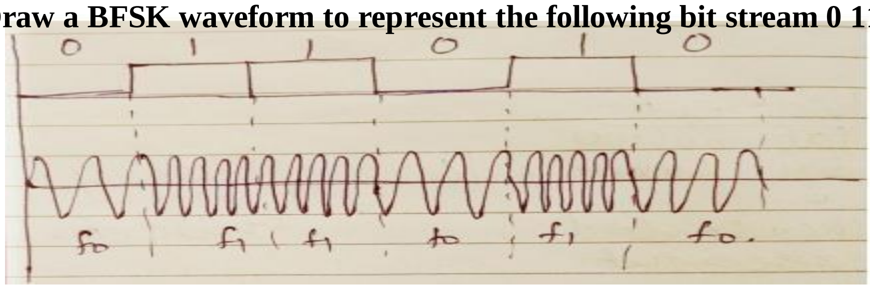
The two oscillators, producing a higher and a lower frequency signals, are connected to a switch along with an internal clock. To avoid the abrupt phase discontinuities of the output waveform during the transmission of the message, a clock is applied to both the oscillators, internally. The binary input sequence is applied to the transmitter so as to choose the frequencies according to the binary input.

Question: Explain the process of FSK modulation with diagram. (4Marks)

Answer: In FSK, frequency of sinusoidal carrier is shifted between two discrete values. One of these frequencies (f_1) represents a binary 1 and other value (f_2) represents binary 0.There is no change in amplitude of carrier. It consists of voltage controlled oscillators (VCO) which produce sine waves at frequencies f_1 and f_0. Corresponding to "binary 0 "input, the VCO produces a sinewave of frequency f_0 whereas corresponding to binary 1 input VCO produces a sinewave of frequency f_1.



Question: Draw a BFSK waveform to represent the following bit stream 0 11 0 1 0.



Application:

1. Many modems used FSK in telemetry systems

3. Phase Shift Keying (PSK)

Phase Shift Keying (PSK) is the digital modulation technique in which the phase of the carrier signal is changed by varying the sine and cosine inputs at a particular time. PSK technique is widely used for wireless LANs, bio-metric, contactless operations, along with RFID and Bluetooth communications.

PSK is of two types, depending upon the phases the signal gets shifted. They are −

Binary Phase Shift Keying (BPSK)

This is also called as 2-phase PSK or Phase Reversal Keying. In this technique, the sine wave carrier takes two phase reversals such as 0° and 180°.

BPSK is basically a Double Side Band Suppressed Carrier (DSBSC) modulation scheme, for message being the digital information.

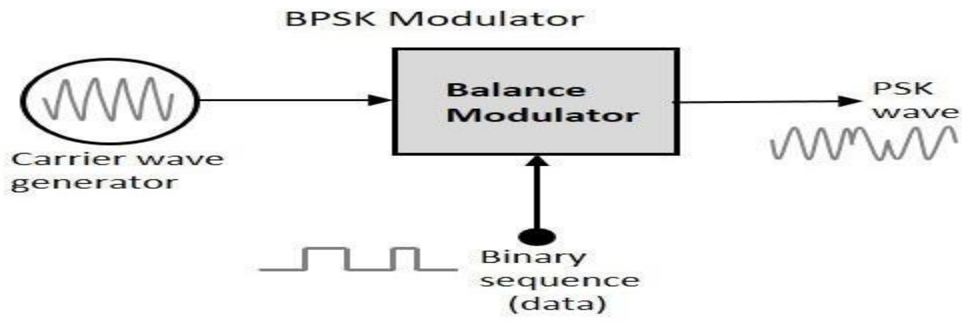
Quadrature Phase Shift Keying (QPSK)

This is the phase shift keying technique, in which the sine wave carrier takes four phase reversals such as 0°, 90°, 180°, and 270°.

If this kind of techniques are further extended, PSK can be done by eight or sixteen values also, depending upon the requirement.

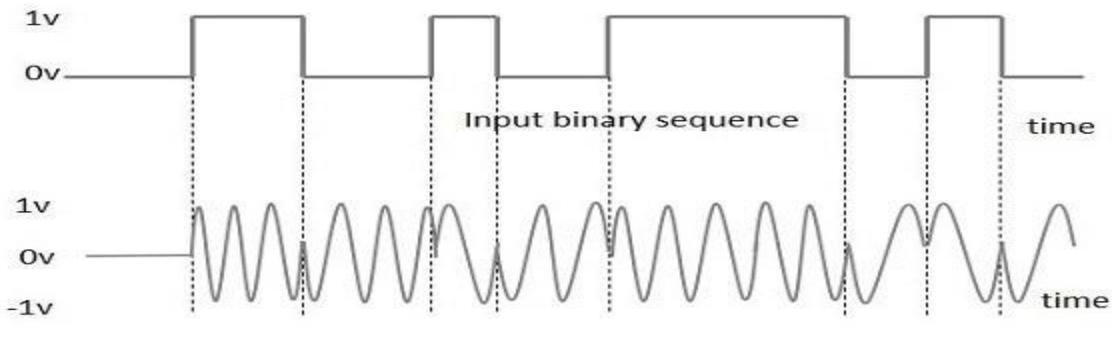
BPSK Modulator

The block diagram of Binary Phase Shift Keying consists of the balance modulator which has the carrier sine wave as one input and the binary sequence as the other input. Following is the diagrammatic representation.



The modulation of BPSK is done using a balance modulator, which multiplies the two signals applied at the input. For a zero binary input, the phase will be **0°** and for a high input, the phase reversal is of **180°**.

Following is the diagrammatic representation of BPSK Modulated output wave along with its given input.

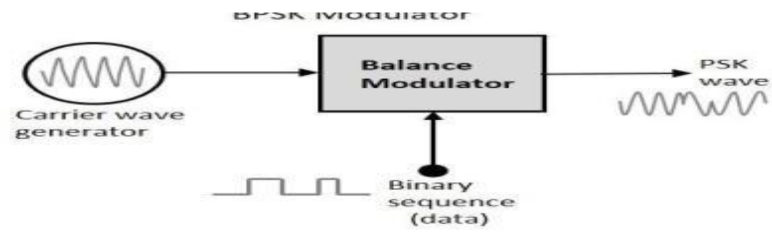


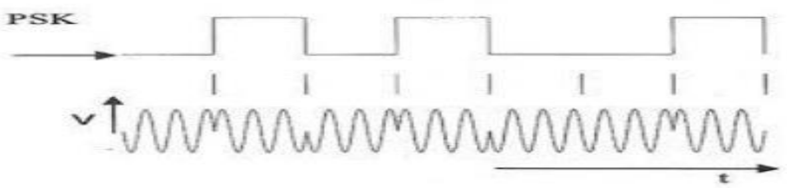
BPSK Modulated output wave

The output sine wave of the modulator will be the direct input carrier or the inverted (180° phase shifted) input carrier, which is a function of the data signal.

Question: Explain process of phase shift keying.(4 Marks)

Answer: Phase-shift keying (PSK) is a digital to analog modulation scheme based on changing, or modulating, the initial phase of a carrier signal. PSK is used to represent digital information, such as binary digits zero (0) and one (1).The modulation of PSK is done using a balance modulator, which multiplies the two signals applied at the input. For a zero binary input, the phase will be 180° and for a high input, the phase reversal is of 0°. Following is the diagrammatic representation of PSK Modulated output wave along with its given input.





The output sine wave of the modulator will be the direct input carrier or the inverted (180° phase shifted) input carrier, which is a function of the data signal. Amplitude and frequency of the original carrier signal is kept constant.

Application:

1. Used in our ADSL broadband modem
2. Used in satellite communication
3. Used in our mobile phones

Comparison of ASK, FSK and PSK

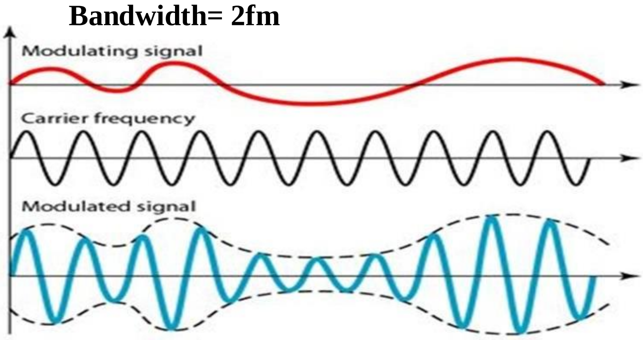
Parameters	**ASK**	**FSK**	**PSK**
Variable characteristics	**Amplitude**	**Frequency**	**Phase**
Bandwidth	**Is proportional to signal rate (B =(1+d)S),d is due to modulation & filtering ,lies between 0 & 1.**	**B=(1+d)×S+2Δf**	**B=(1+d)×S**
Noise immunity	**low**	**High**	**High**
Complexity	**Simple**	**Moderately complex**	**Very complex**
Error probability	**High**	**Low**	**Low**
Performance in presence of noise	**Poor**	**Better than ASK**	**Better than FSK**
Bit rate	**Suitable upto 100 bits/sec**	**Suitable upto about 1200 bits/sec**	**Suitable for high bit rates**

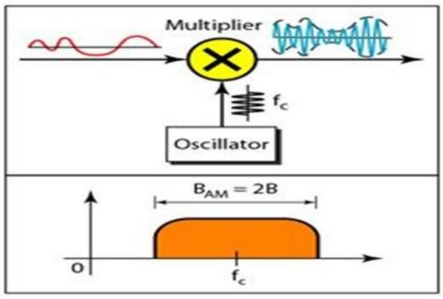
- ## Analog to Analog Conversion

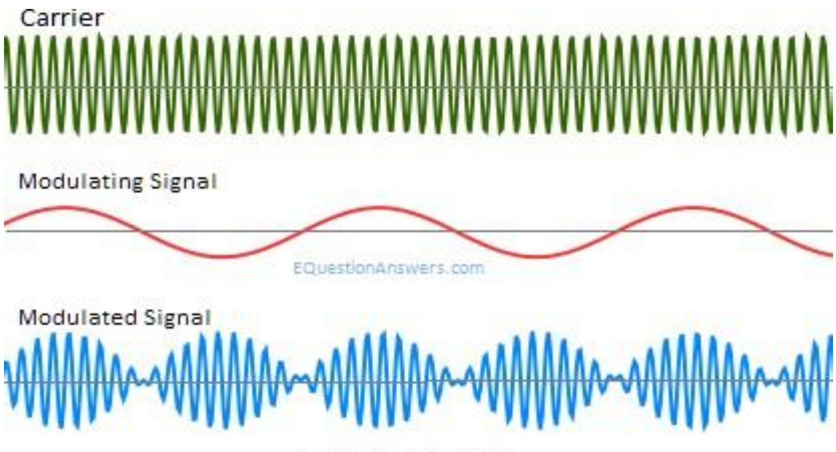
  - Analog-to-analog conversion, or modulation, is the representation of analog information by an analog signal.
  - It is a process by which a characteristic of carrier wave is varied according to the instantaneous amplitude of the modulating signal.
  - Analog to Analog conversion can be done in three ways:
    - **Amplitude Modulation**
    - **Frequency Modulation**
    - **Phase Modulation**

1. AMPLITUDE MODULATION:

- The modulation in which the **amplitude** of the carrier wave is varied according to the instantaneous amplitude of the modulating signal keeping **phase and frequency** as constant.
- AM is normally implemented by using a simple multiplier because the amplitude of the carrier signal needs to be changed according to the amplitude of the modulating signal.
- **AM bandwidth:**
  The modulation creates a bandwidth that is twice the bandwidth of the modulating signal and covers a range centered on the carrier frequency.
  Bandwidth= 2fm







Amplitude Modulation

- ## AM Advantage

  - AM is the simplest type of modulation. Hardware design of both transmitter and receiver is very simple and less cost effective.

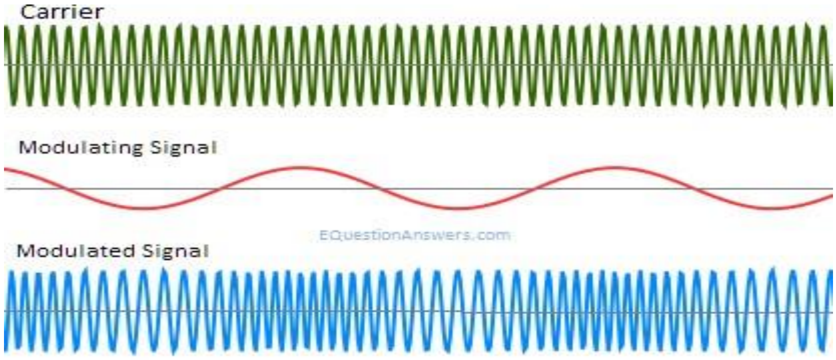
- ## AM Disadvange:

  - AM is very susceptible to noise.

- **Application:**
    - AM radio broad cast is an example

2. Frequency modulation

FM or Frequency modulation is the process of varying the in instantaneous frequency of Carrier signal accordingly with instantaneous amplitude of message signal.



Frequency Modulation

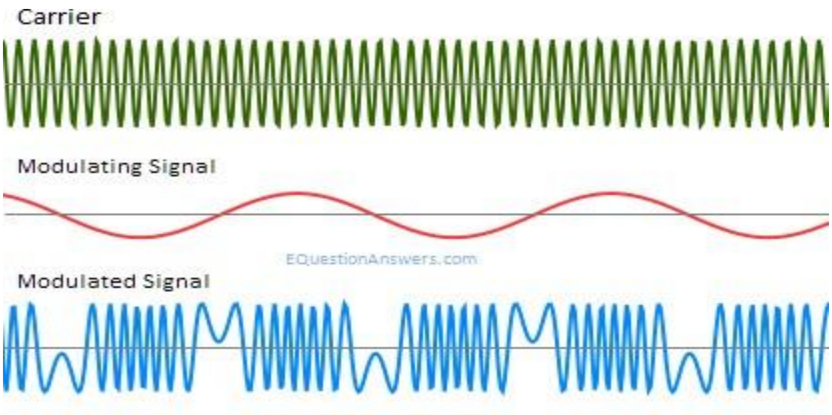
- **FM Advantage**
    - Modulation and demodulation does not catch any channel noise.
- **FM Disadvange:**
    - Circuit needed for FM modulation and demodulation is bit complicated than AM
- **Application:**
    - FM radio broad cast is an example

Question: Compare amplitude modulation and frequency modulation (4 points).
Answer:

Parameter	Amplitude modulation (AM)	Frequency modulation (FM)
Definition	Amplitude modulation (AM) is the process of changing the amplitude of a high frequency carrier signal in proportion with the instantaneous value of the modulating signal keeping frequency &Phase constant.	Frequency modulation (FM) is the process of changing the frequency of carrier signal in proportion with the instantaneous value of the modulating signal keeping Amplitude &Phase constant.
Waveform	AM wave: 	FM wave: 
Bandwidth	BW= $2f_m$(f_m -frequency of modulating signal)	Bandwidth =2 [$\delta + f_m$] (f_m - frequency of modulating signal)
Noise immunity	Less	More
Modulation index	$m_a = \frac{V_m}{V_c}$ V_m - Amplitude of modulating signal V_c - Amplitude of carrier signal	$m_f = \frac{\delta}{f_m}$ δ – frequency deviation f_m - frequency of modulating signal
Frequencies used for transmission	535 – 1700 KHz	88.1 – 108.1 MHz

3. Phase modulation (PM)

PM or Phase modulation is the process of varying the instantaneous phase of Carrier signal accordingly with instantaneous amplitude of message signal.



Phase Modulation

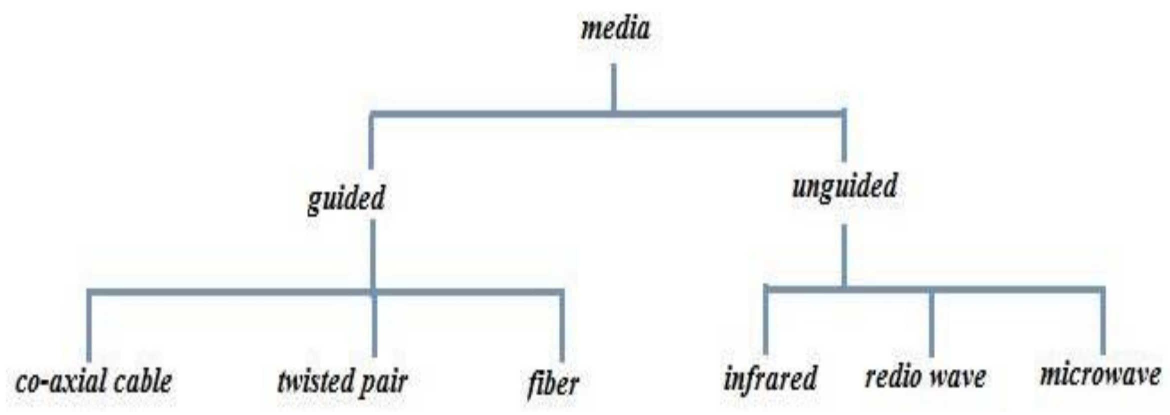
- **PM Advantage**
  - Modulation and demodulation does not catch any channel noise.
- **PM Disadvange:**
  - Circuit needed for PM modulation and demodulation is bit complicated than AM and FM
- **Application:**
  - Satellite communication.

Chapter No: 2

TRANSMISSION MEDIA

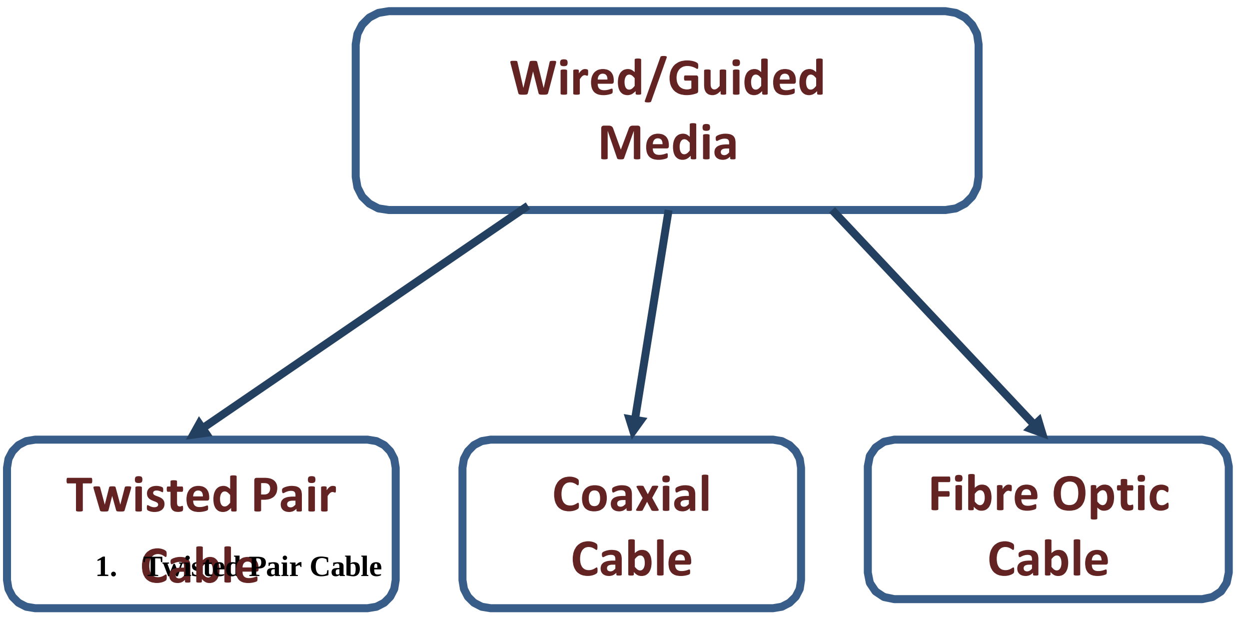
Transmission Media

- Transmission medium is the way in which data is transmitted from one place to another.
- It provide a pathway over which the message can travel from sender-to-receiver.
- Each of the message can be sent in the form of data by converting them into binary digits.
- These binary digits are then encoded into a signal that can be transmitted over the appropriate medium.

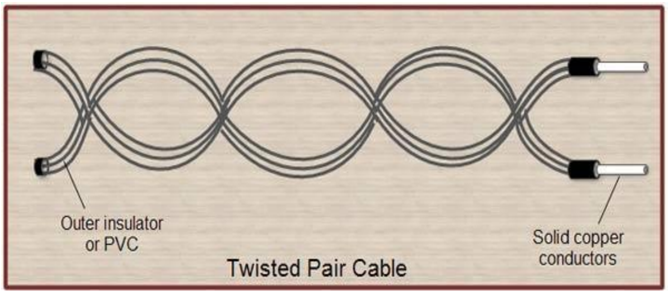


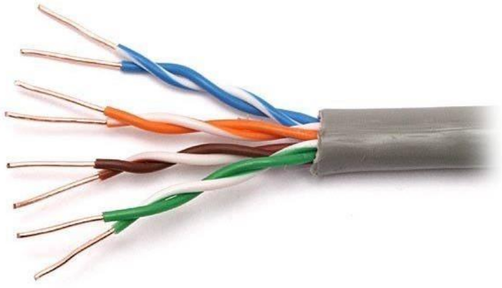
I. Wired/Guided Transmission Media

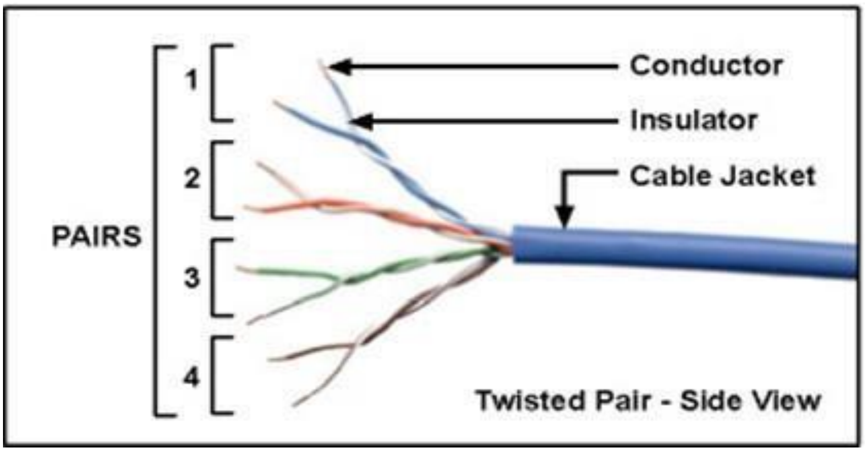
- Guided transmission media are the cables that are tangible or have physical existence.
- Bounded transmission means having connectivity between a source and destination using cables or wires. The signals have to travel through this channel i.e. physical media



1. Twisted Pair Cable







- A twisted pair cable is a pair of copper wires.
- Copper wires are the most common wires used for transmitting signals because of good performance at low costs.
- A twisted pair consists of two conductors (normally copper), each with its own plastic insulation, twisted together to form a single media.
- Out of these two wires, only one carries actual signal and another is used for ground reference.
- To identify every cable, these cables are colour coated.
- The twists between wires are helpful in reducing noise (electro-magnetic interference) and crosstalk.
- This type of cable is used in telephone lines to provide voice and data channels.

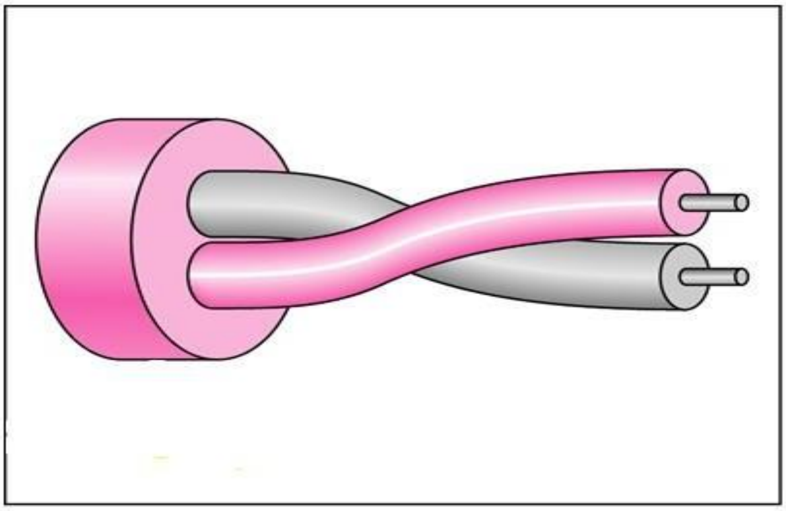
Types of Twisted Pair

The two types of twisted pairs are:

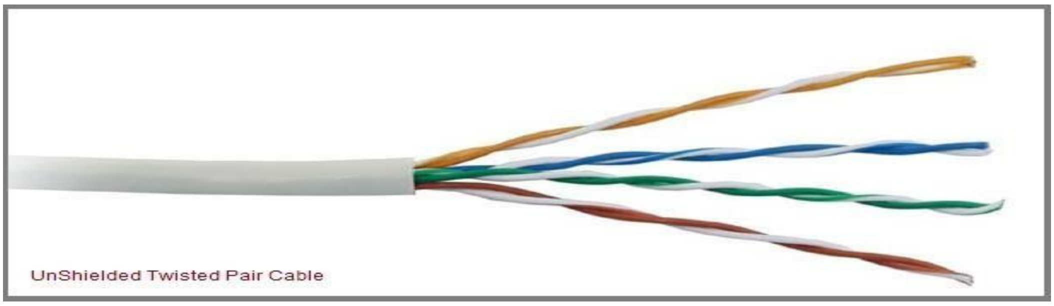
1. Unshielded twisted pair (UTP)

2. Shielded twisted pair (STP)

1. **Unshielded twisted pair (UTP):-**



- UTP is more common.
- UTP cost less than STP and easily available due to its many use.
- Due to its low cost, UTP cabling is used extensively for local-area networks (LANs) and telephone connections.
- UTP cables consist of 2 or 4 pairs of twisted cable.
- Cable with 2 pair use RJ-11 connector and 4 pair cable use RJ-45 connector.



Advantages of UTP:

- Easy installation and setup.
- Capable of high speed for LAN.
- Low cost.
- UTP is very flexible.

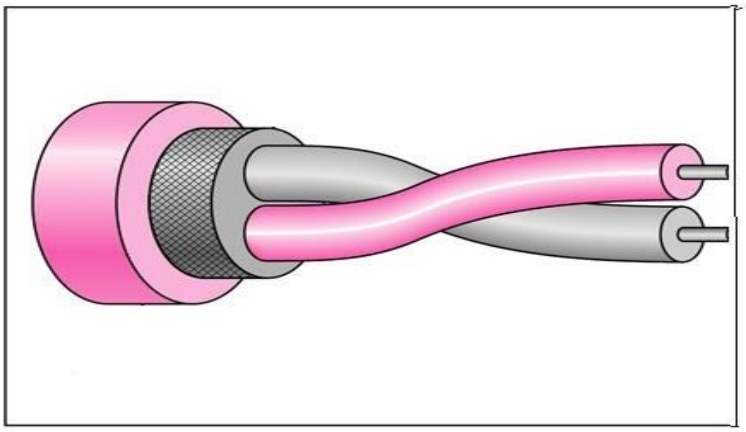
Disadvantages of UTP:

- Short distance due to attenuation.
- Limited bandwidth.

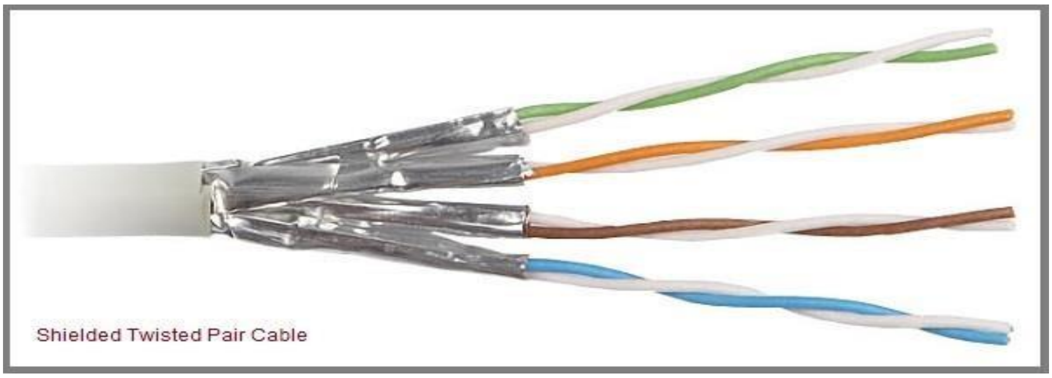
Application

- Commonly used in telephone lines.

2. **Shielded twisted pair (STP):-**



- This type of cable has a metal foil covering which encases each pair of insulator conductors.
- Electromagnetic noise penetration is prevented by metal casing. Shielding also eliminates crosstalk.
- It is similar to UTP but has a mesh shielding that's protects it from EMI which allows for higher transmission rate.
- It is more expensive than coaxial and unshielded twisted pair.



Advantages of STP:

- STP reduces interference.
- Faster than UTP and coaxial cable.
- Better performance at higher data rates.

Disadvantages of STP:

- More expensive than UTP and coaxial cable.
- More difficult installation and setup.
- High attenuation rate.
- High cost.

Question: Explain the construction of Shielded Twisted Pair Cable.

- STP is similar to UTP but with each pair covered by an additional copper braid jacket or foil wrapping. This shielding helps to protect the signals on the cables from external interference.

- Shielding provides a means to reflect or absorb electric fields that are present around cables. Shielding comes in a variety of forms from copper braiding or copper meshes to aluminized.
- STP is more expensive than UTP but has the benefit of being able to support higher transmission rates over longer distances.
- STP is heavier and more difficult to manufacture, but it can greatly improve the signaling rate in a given transmission scheme Twisting provides cancellation of magnetically induced fields and currents on a pair of conductors.
- Magnetic fields arise around other heavy current-carrying conductors and around large electric motors. Various grades of copper cables are available, with Grade 5 being the best and most expensive.
- STP is used in IBM token ring networks.

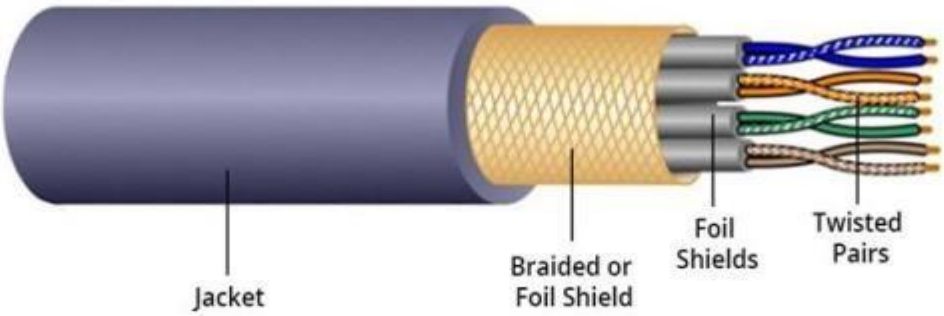
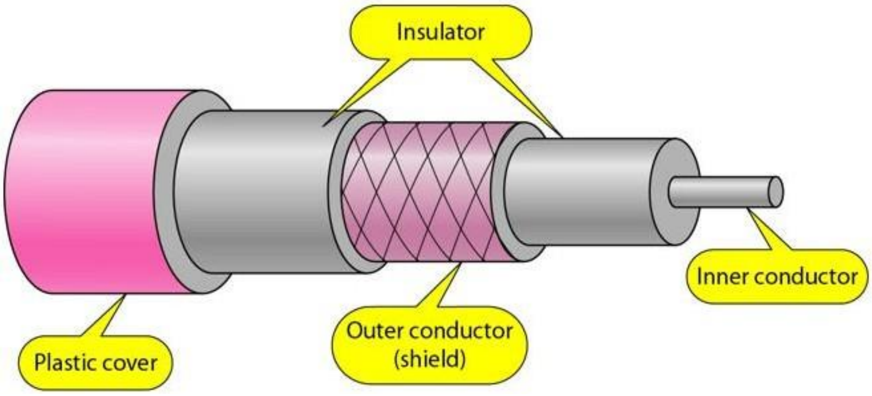


Figure: Construction of Shielded Twisted Pair

- RJ-45 connectors is used with Ethernet cables in computer networking.
- RJ-11 connectors is used in connecting telephone units.

Coaxial Cable

- **Coaxial cables** are copper cables with better shielding than twisted pair cables, so that transmitted signals may travel longer distances at higher speeds.
- The shield minimizes electrical and radio frequency interference.
- Coaxial cabling is the primary type of cabling used by the cable television industry and is also widely used for computer networks, such as Ethernet.



- Coaxial cable has two wires of copper.
- The core/inner copper wire in centre and is made of solid conductor. It is enclosed in an insulating sheath.
- The second/outer copper wire is wrapped around, and is used to protect from external electromagnetic interference (Noise).
- This all is covered by plastic cover used to protect the inner layers from physical damage such as fire or water.

Coaxial Cable Standards

- Coaxial cables are categorized by their Radio Government (RG) ratings. Each RG number denotes a unique set of physical specifications
    - 50-Ohm RG-7 or RG-11 : used with thick Ethernet.
    - 50-Ohm RG-58 : used with thin Ethernet
    - 75-Ohm RG-59 : used with cable television



Advantages of Co-axial Cable:

- Low cost due to less total footage of cable, hubs not needed.
- Lower attenuation than twisted pair.
- Supports high bandwidths.
- Can support high data rates.

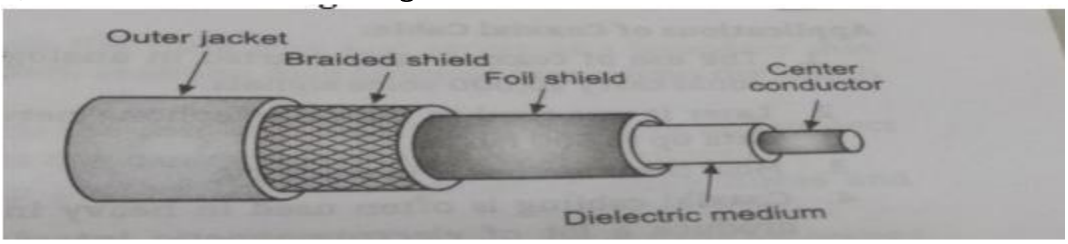
Disadvantages of Co-axial Cable:

- Limited in network speed.
- Limited in size of network.
- One bad connector can take down entire network.
- Coax is among the most expensive types of wire cables

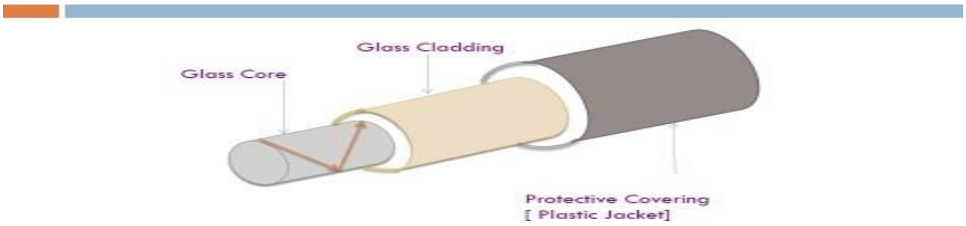
Applications of Co-axial Cable:

- Digit Analog telephone n/w.
- Analog telephone n/w.
- Cable TV.
- Ethernet LAN's – Thick and Thin.
- Digital transmission.
- Long distance telephone transmission– can carry 10,000 voice calls.

Question: Draw a labeled diagram of coaxial cable.

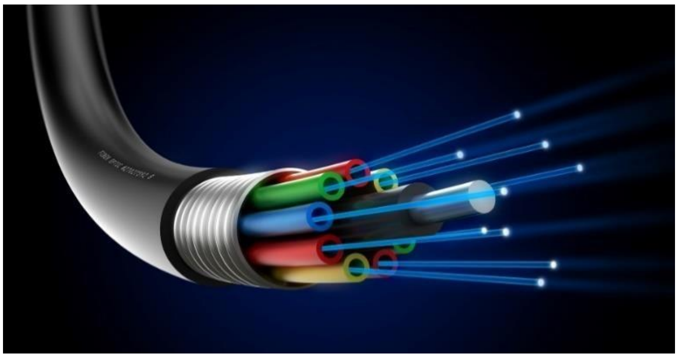


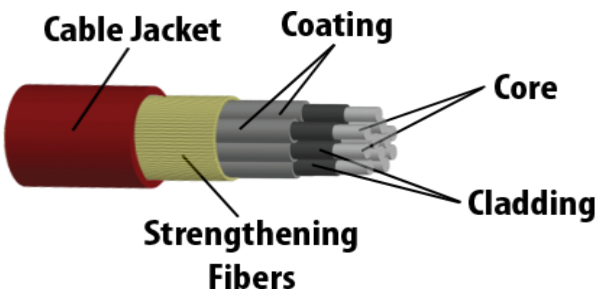
Fibre Optics Cable

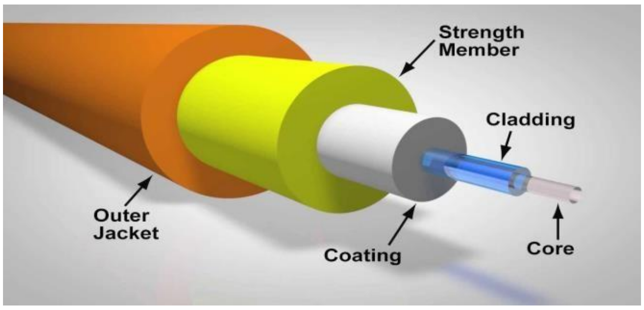


- A fibre optic cable is made of high quality of thin glass or plastic and is used to transfer digital data signals in the form of light up to distance of thousands of miles.
- Fibre optic cables are not affected by electromagnetic interference, so noise and distortion is very less.
- Fibre optic cables carry communication signals using pulses of light generated by small lasers or light-emitting diodes (LEDs).

- The cable consists of one or more strands of glass, each only slightly thicker than a human hair. The centre of each strand is called the core, which provides the pathway for light to travel. The core is surrounded by a layer of glass called cladding that reflects light inward to avoid loss of signal and allow the light to pass through bends in the cable. No light escapes the glass core because of this reflective **cladding.**







Advantages of Optical Fibre:-

- Fibre optic cables have a much High bandwidth than metal cables. This means that they can carry more data.
- Smaller Size and Lighter weight.
- low attenuation
- Not affected electromagnetic interference (No EMI interference)
- Signals carrying data can travel long distances without weakening
- Suitable for industrial and noisy areas

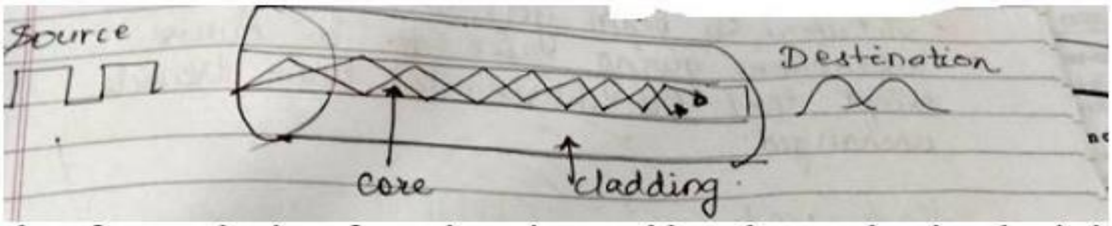
Disadvantages of Optical Fibre:-

- Optical fibre cables are expensive
- Difficult to install
- Maintenance is expensive and difficult

Question:Explain propagation modes in fiber optic cable with neat diagram.

The different propagation modes in fiber optic cable are as follows:

- **Multimode step index fiber**: In multimode step index fiber, the core has one density and the cladding has another density.



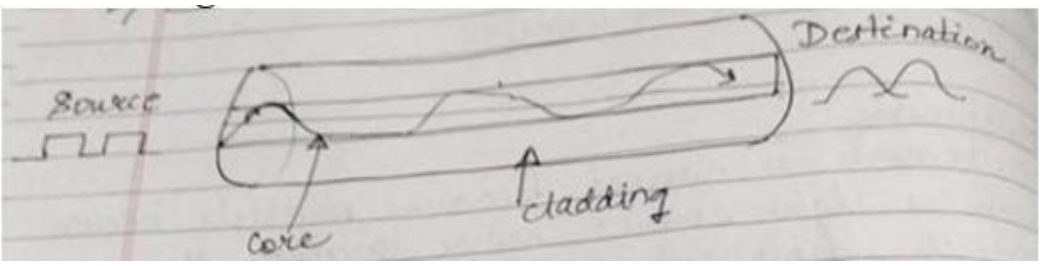
- Therefore at the interface, there is a sudden change that is why it is called step index. Multiple beams take different paths on reflection as shown in figure.
- The beam that strikes core at a smaller angle that has to be reflected many more times than the beam that shifted the core at a larger angle to reach other end. This means that at the destination, all beams do not reach simultaneously. It is used for short distances.

➢ **Multimode graded-index fiber:**

In this, core itself is made of a material of varying densities.

• The density is the highest at the core and gradually decreases towards the edge.

• Therefore, a beam gas through gradual refraction giving rise to a curve except that the horizontal beam travels unchanged.

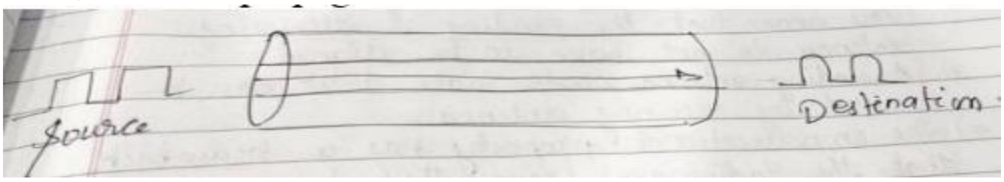


➢ **Single-mode:**

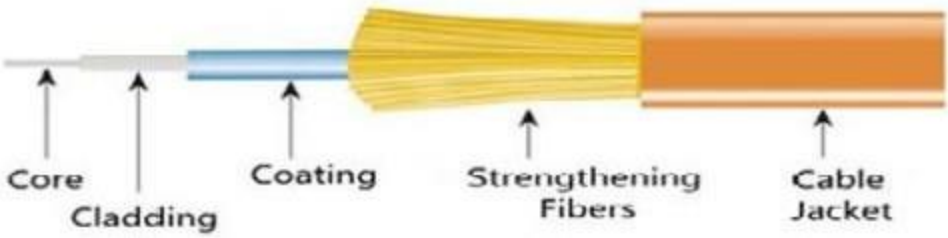
• It uses step-index fiber and a highly focused source of light that limits beam to a small range of angles, all close to horizontal.

• It is manufactured with much smaller diameter than that of multimode fiber and with substantially lower density.

• The decrease in density results in a critical angle i.e. close enough to 90 to make propagation of beams almost horizontal.



Question:Draw a labeled diagram of fiber optic cable and state its advantages.



Advantages of fiber optic cable:

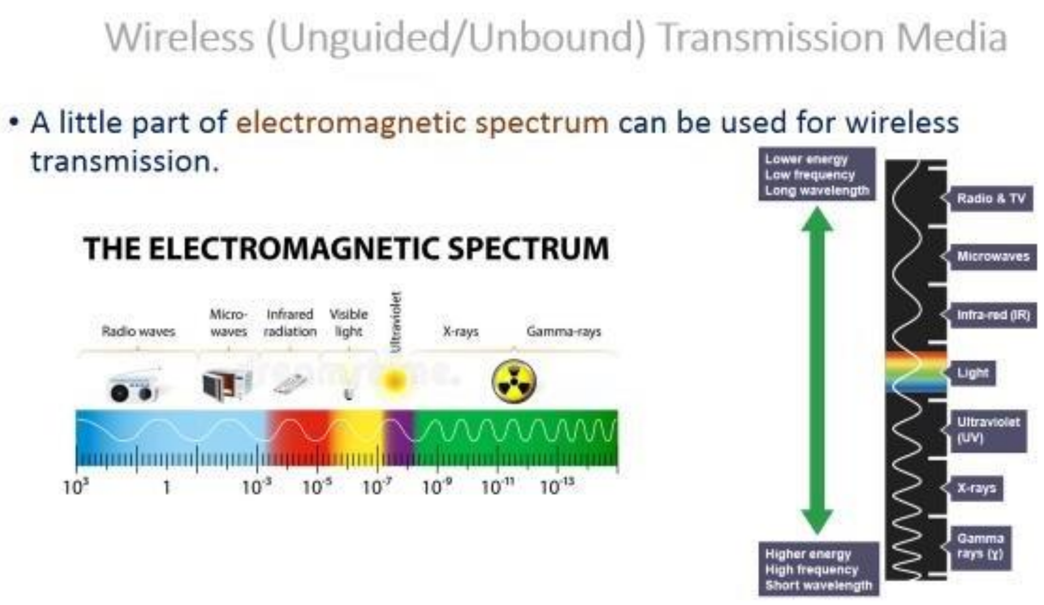
1.Higher data rate
2.Large Bandwidth
3.Less signal attenuation
4.Light weight.
5.More reliability
6.Long distance.
7.Higher security.

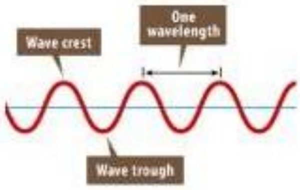
Question: Differentiate between twisted pair coaxial cable and fiber optic cable (any 4 points).

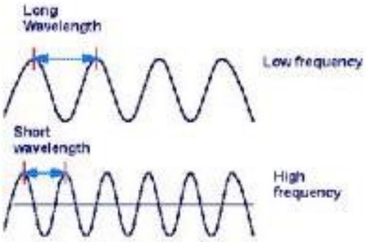
Sr. No.	Twisted pair cable	Coaxial cable	Fiber optic cable
1	Transmission of signals of takes place in the electrical form over the metallic conducting wires.	Transmission of signals takes place in the electrical form over the inner conductor of the cable.	Signal transmission takes place in an optical form over a glass fiber.
2	In this medium the noise immunity is low.	Coaxial having higher noise immunity than twisted pair cable.	Optical fiber has highest noise immunity as the light rays are unaffected by the electrical noise.
3	Twisted pair cable can be affected due to external magnetic field.	Coaxial cable is less affected due to external magnetic field.	Not affected by the external magnetic field.
4	Cheapest medium	Moderate Expensive	Expensive
5	Low Bandwidth	Moderately high bandwidth	Very high bandwidth
6	Attenuation is very high	Attenuation is low	Attenuation is very low
7	Installation is easy	Installation is fairly easy	Installation is difficult

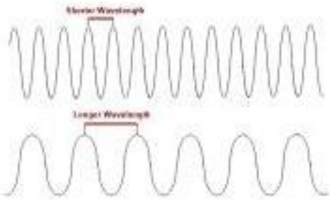
II. Wireless (Unguided/Unbound) Transmission Media

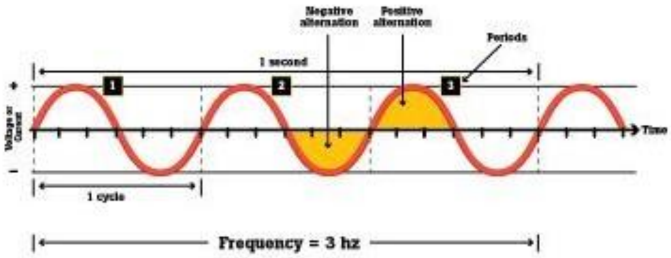
- A wave can be described as a disturbance that travels through a medium from one location to another location.
- A **wave** is a transfer of energy, usually through a form of matter called a **medium**.
- There are a special type of wave that can travel without a medium, called **electromagnetic waves (**also called **EM** waves)**,** which are waves like radio waves and microwaves.
- Unlike sound waves and water waves, electromagnetic waves don't need a fluid, or a solid, or even air to help them travel from one place to another. EM waves can travel across the great vacuum of space, which is why we see light from distant stars and planets.
- Electromagnetic waves are formed when an electric field comes in contact with a magnetic field. They are hence known as 'electromagnetic' waves.
- Electromagnetic (EM) radiation is a form of energy that is all around us and takes many forms, such as radio waves, microwaves, X-rays and gamma rays.
- Sunlight is also a form of EM energy. Electromagnetic energy from the sun comes to Earth in the form of radiation.
- The Electromagnetic Spectrum describes a wide range of different electromagnetic waves.



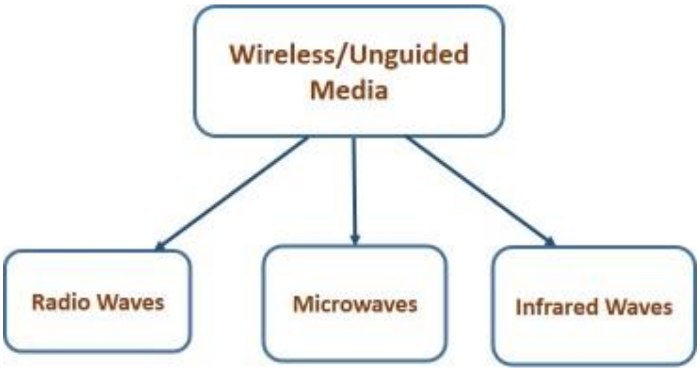








Band	Frequency range	Wavelength range
Extremely Low Frequency (ELF)	<3 kHz	>100 km
Very Low Frequency (VLF)	3 to 30 kHz	10 to 100 km
Low Frequency (LF)	30 to 300 kHz	1 m to 10 km
Medium Frequency (MF)	300 kHz to 3 MHz	100 m to 1 km
High Frequency (HF)	3 to 30 MHz	10 to 100 m
Very High Frequency (VHF)	30 to 300 MHz	1 to 10 m
Ultra High Frequency (UHF)	300 MHz to 3 GHz	10 cm to 1 m
Super High Frequency (SHF)	3 to 30 GHz	1 to 1 cm
Extremely High Frequency (EHF)	30 to 300 GHz	1 mm to 1 cm



Radio Waves Transmission

- Radio waves are EM (Electromagnetic) waves that have wavelengths between 1 millimetre and 100 kilometres (or 300 GHz and 3 kHz in frequency).
- Radio frequency is easy to generate because it has large wavelength and can travel long distance.
- Radio waves are generated by radio transmitters and received by radio receivers.
- Radio stations transmit radio waves using transmitters, which are received by the receiver installed in our devices. Both transmitters and receivers use antennas to radiate or capture radio signals
- It can penetrate walls easily, so these waves are widely used for communication both indoors and outdoors.
- Radio waves are omnidirectional means they travel in all the directions from the source.
- When an antenna transmits radio waves, they are propagated in all directions.
- A sending antenna send waves that can be received by any receiving antenna. The omnidirectional property has disadvantage, too. The radio waves transmitted by one antenna are susceptible to interference by another antenna that may send signal using the same frequency or band.

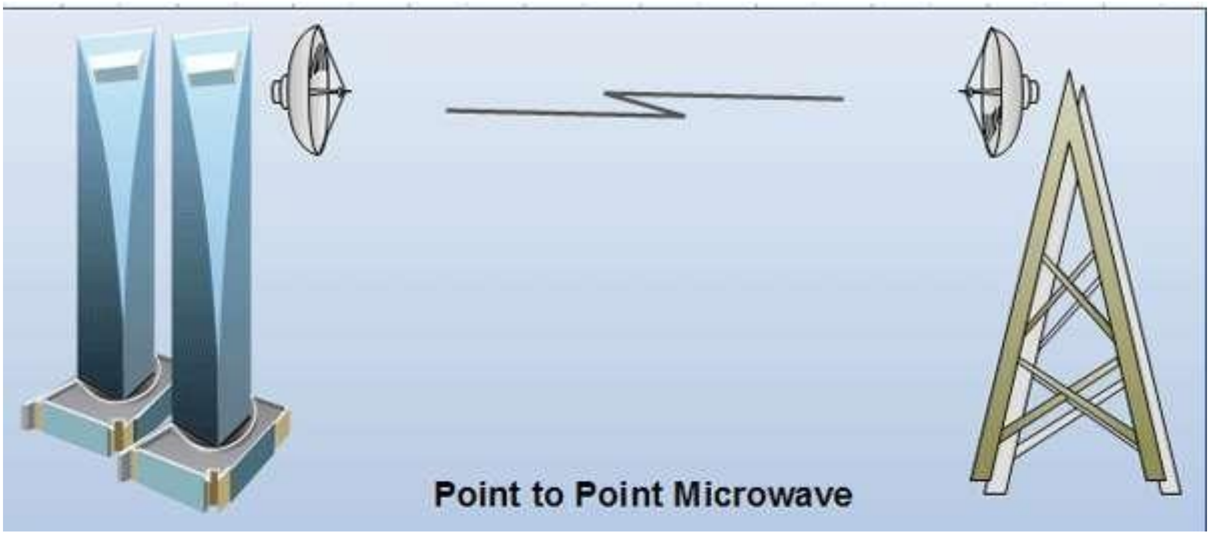
- It is Used Mobile, AM/FM radio, television

Radio Spectrum

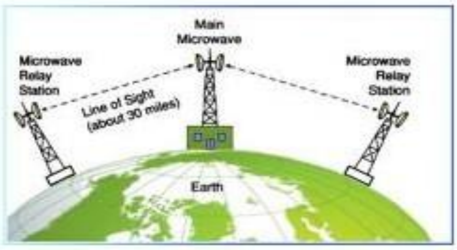
Radio Band	Frequency	Some Applications
Very Low Frequency VLF	3 KHZ to 30 KHz	Radio Navigation
Low Frequency LF	30 KHz to 300 KHz	Long Wave Radio
Medium Frequency MF	30 KHz to 3 MHz	AM Radio
High Frequency HF	3 MHz to 30 MHz	CB Radio (HAM) Point to Point Radio Search and Rescue Services
Very High Frequency VHF	30 MHz to 300 MHz	FM radio 88-108 MHz VHF Broadcast TV
Ultra High Frequency UHF	300 MHz to 3 GHz	UHF Broadcast TV Cellular Phones Microwave Links Wi-Fi in 2.4 Band Satellite Communications
Super High Frequency SHF	3 GHz to 30 GHz	Microwave Links Wi-Fi in 5 GHz Band Satellite Communications
Extra High Frequency EHF	30 GHz to 300 GHz	Microwave Links

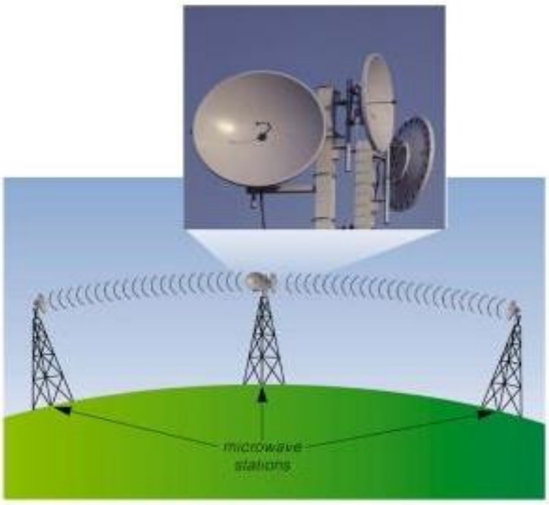
Micro Waves Transmission

- Microwaves are a type of radio waves with high frequencies. It can be classified as a subclass of radio waves. The frequency of microwaves lies in the 300 MHz to 300 GHz.
- Unlike radio waves, microwaves are unidirectional, in which the sending and receiving antennas need to be aligned.
- Microwaves are widely used for point-to-point communications because their small wavelength, which means that the signal is focused into a narrow beam. Additionally, each antenna must be within line of sight of the next antenna.



- Electromagnetic waves above 100 MHz tend to travel in a straight line and signals over them can be sent by beaming those waves towards one particular station. Because Microwaves travels in straight lines, both sender and receiver must be aligned to be strictly in line-of-sight.





- Microwaves have higher frequencies and do not penetrate wall like obstacles.
- It is used for satellite communication, navigation, radar, remote sensing and other short distance communication systems.\

Question: Explain the reason for using different frequency bands for uplink and downlink in satellite communication.

- The **uplink** frequency is the frequency which is used for transmission of signals from earth station transmitter to the satellite.
  The **downlink** frequency is the frequency which is used for transmission of signals from the satellite to the earth station receiver

- Uplink frequency is different from downlink frequency for following reason:
  - The satellite transmitter generates a signal that would jam its own receiver; if both uplink and downlink shared the same frequency.
  - Trying to receive and transmit an amplified version of the same uplink waveform at same satellite will cause unwanted feedback or ring around from the downlink antenna back into the receiver.
  - Frequency band separation allows the same antenna to be used for both receiving and transmitting, simplifying the satellite hardware.
- To overcome the above-mention difficulties satellite repeaters must involve some form of frequency translation before power amplification.
  So, Uplink frequency is different from downlink frequency.

Question: Explain Microwave transmission with its advantages and disadvantages.

- **Microwave:**
  Electromagnetic waves having frequencies between 1 and 300GHz are called microwaves.
- Microwaves are unidirectional. When an antenna transmits microwave waves, they can be narrowly focused. This means that the sending and receiving antennas need to be aligned.
- The unidirectional property has an obvious advantage. A pair of antennas can be aligned without interfering with another pair of aligned antennas.

The following describes some characteristics of microwave propagation:

- Microwave propagation is line-of-sight.
- Very high-frequency microwaves cannot penetrate walls. This characteristics can be a disadvantage if receivers are inside buildings.
- The microwave band is relatively wide, almost 299 GHz. Therefore wider sub bands can be assigned, and a high data rate is possible.
- Use of certain portions of the band requires permission from authorities

Applications:

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

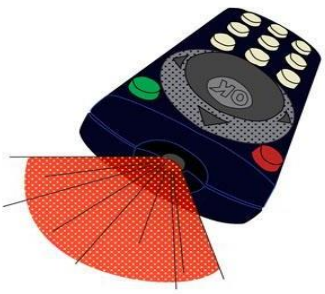
Advantages:

- Installation of towers and associated equipment's is cheaper than laying down a cable of 100KM length.
- Less maintenance as compared to cables.
- Repeaters can be used. So effect of noise is reduced.
- No adverse effects such as cable breakage.
- Due to the use of highly directional antenna no interference is there.
- Size of transmitter and receiver reduces due to the use of high frequency.

Disadvantages:

- Signal strength at the receiving antenna reduces due to multipath reception.

Infrared Waves Transmission



- Infrared signals have frequencies between 300 GHz to 400 THz. They are used for short-range communication.
- Infrared waves are used for very short distance communication like TV remote, wireless speakers, automatic doors, hand held devices etc.
- Infrared waves having high frequencies prevents interference b/w one system to another.

- Infrared signals have high frequencies and cannot penetrate walls. Due to its short-range communication system, the use of an infrared communication system in one room will not be affected by the use of another system in the next room. This is why using an infrared TV remote control in our home will not interfere with the use of our neighbour's infrared TV remote control.

THE DISADVANTAGES OF USING INFRARED

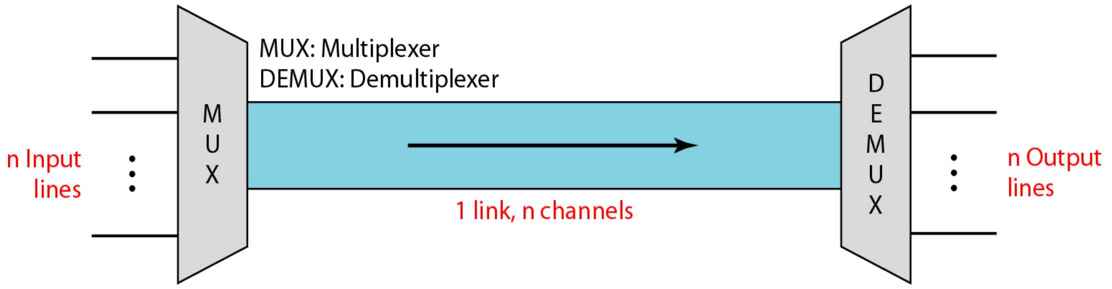
- Infrared signals cannot be used for long distance communication. In addition, we cannot use infrared waves outside a building because sun's rays contain infrared waves that can interfere with communication.

Chapter No: 3

MULTIPLEXING AND SWITCHING

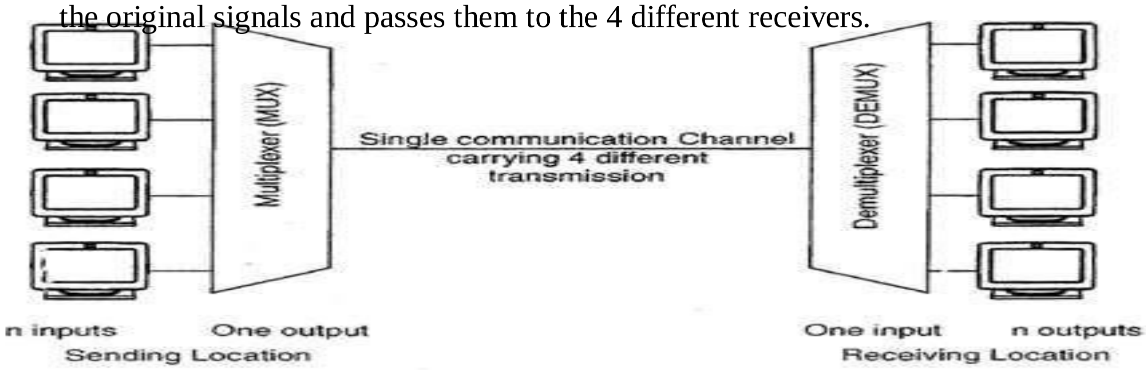
Multiplexing and De-multiplexing

- To combine multiple signals (analog or digital) for transmission over a single line or media.
- A common type of multiplexing combines several low-speed signals for transmission over a single high-speed connection.
- **Multiplexing** is done by using a device called **Multiplexer** (MUX) that combines ***n*** input lines to generate ***one*** output line i.e. (**many to one**). Therefore multiplexer (MUX) has several inputs and one output.
- At the receiving end, a device called **Demultiplexer** (DEMUX) is used that separates signal into its component signals. So DEMUX has ***one*** input and ***several*** outputs.



Concept of Multiplexing

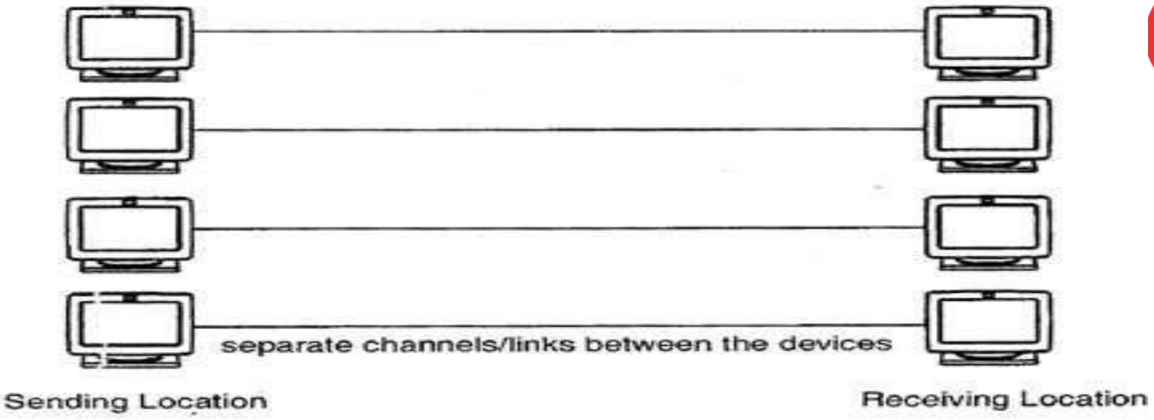
- ☐ As shown in fig multiplexer takes 4 input lines and diverts them to single output line.
- ☐ The signal from 4 different devices is combined and carried by this single line.
- ☐ At the receiving side, a demultiplexer takes this signal from a single line & breaks it into the original signals and passes them to the 4 different receivers.



Multiplexing

Advantages of Multiplexing

- ☐ If no multiplexing is used between the users at two different sites that are distance apart, then separate communication lines would be required as shown in fig.
- ☐ This is not only costly but also become difficult to manage. If multiplexing is used then, only one line is required. This leads to the reduction in the line cost and also it would be easier to keep track of one line than several lines.
- ☐ More than one signal can be sent over a single medium.
- ☐ The bandwidth of a medium can be utilized effectively.

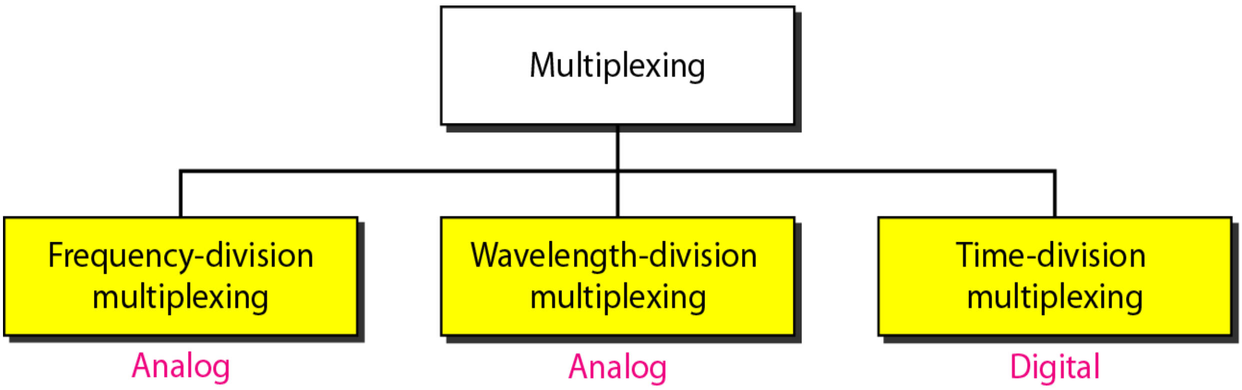


No multiplexing used

Why to use Multiplexing?

- If there are multiple signals to share one medium, then the medium must be divided in such a way that each signal is given some portion of the available bandwidth.
- For example: If there are 10 signals and bandwidth of medium is100 units, then the 10 unit is shared by each signal.
- When multiple signals share the common medium, there is a possibility of collision. Multiplexing concept is used to avoid such collision.

Types of Multiplexing



Question: Define Multiplexing. State its Types. (2 marks)
Answer:

- Multiplexing is the process in which multiple data streams, coming from different sources, are combined and transmitted over a single data channel or data stream.
- The following three major multiplexing techniques:
    - Frequency division multiplexing
    - Wavelength division multiplexing
    - Time division multiplexing

Question: State advantages of multiplexing. (2 Marks)
Answer: Advantages of multiplexing:

1. Simple and easy
2. Large capacities and scalable.
3. Signals from different sources can be sent together through a single common channel.
4. Signals may have varying speed.

Frequency Division Multiplexing (FDM)

- **Frequency-Division Multiplexing (FDM**) is a scheme in which numerous signals are combined for transmission on a single communications line or channel.

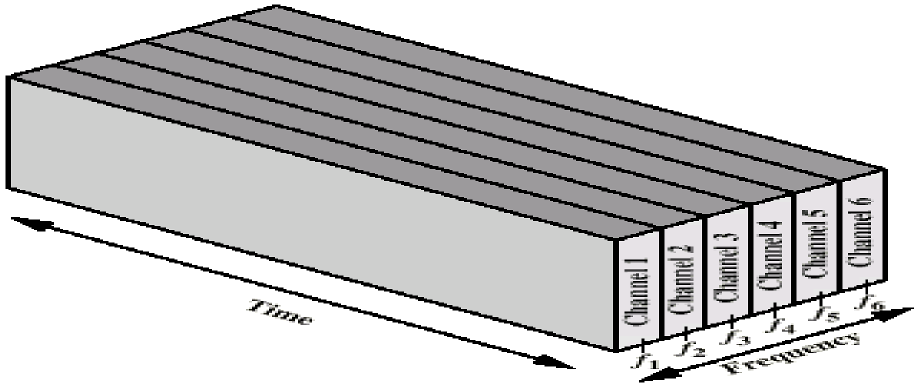
- It is analog technique. Each signal is assigned a different frequency (sub channel) within the main channel.

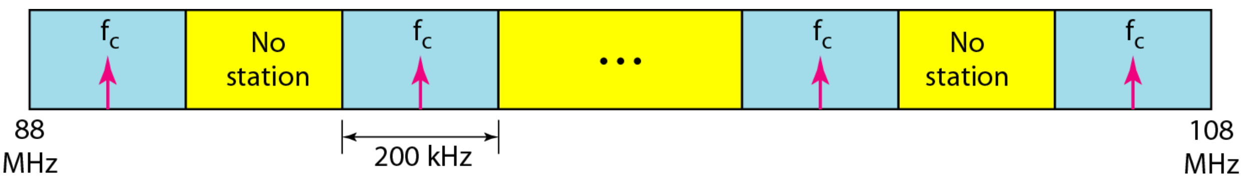
- FDM requires that the bandwidth of a link should be greater than the combined bandwidths of the various signals to be transmitted. Thus each signal having different frequency forms a particular logical channel on the link and follows this channel only. These channels are then separated by the strips of unused bandwidth called guard bands. These guard bands prevent the signals from overlapping as shown in Fig.

- In FDM, signals to be transmitted **must be analog signals**. Thus digital signals need to be converted to analog form, if they are to use FDM.

- A typical analog Internet connection via a twisted pair telephone line requires approximately three kilohertz (3 kHz) of bandwidth for accurate and reliable data transfer.

- Twisted-pair lines are common in households and small businesses. But major telephone cables, operating between large businesses, government agencies, and municipalities, are capable of much larger bandwidths.





FDM Process

- In FDM, signals generated by each sending device modulate different carrier frequencies. These modulated signals are then combined into a single composite signal that can be transported by the link.
- Carrier frequencies are separated by sufficient bandwidth to accommodate the modulated signal.
- These bandwidth ranges are the channels through which the various signals travel.
- Channels can be separated by strips of unused bandwidth **guard bands** to prevent signals from overlapping.
-

Figure: Multiplexing Process :

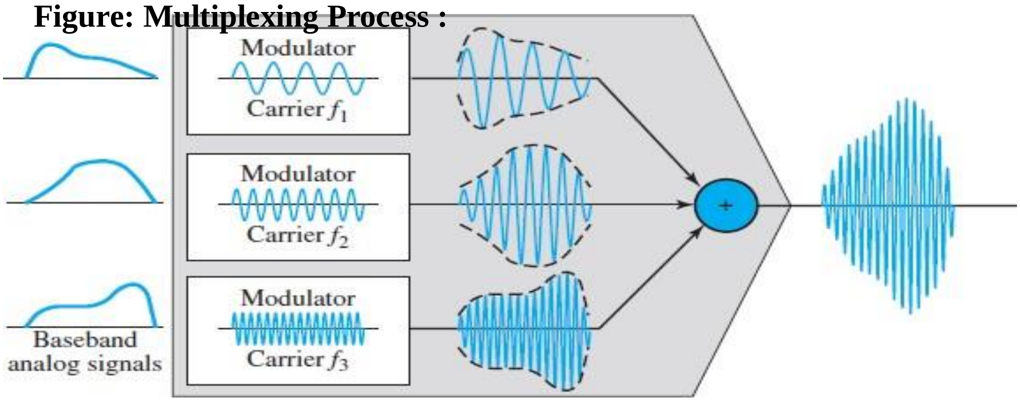
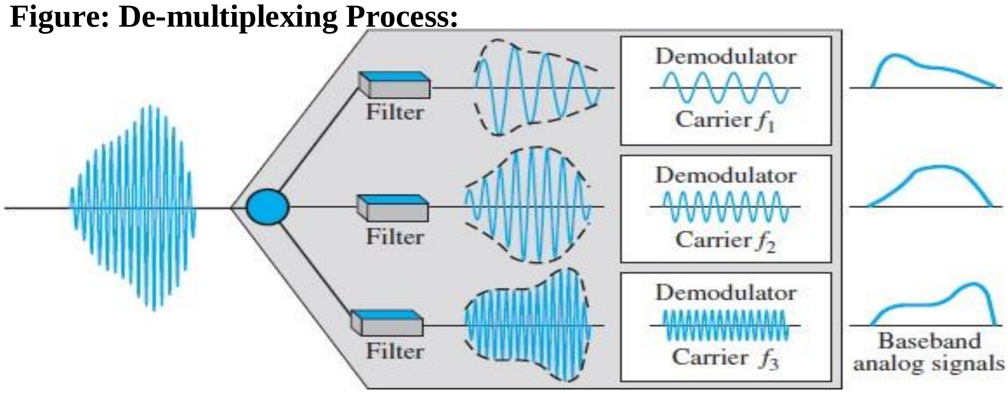


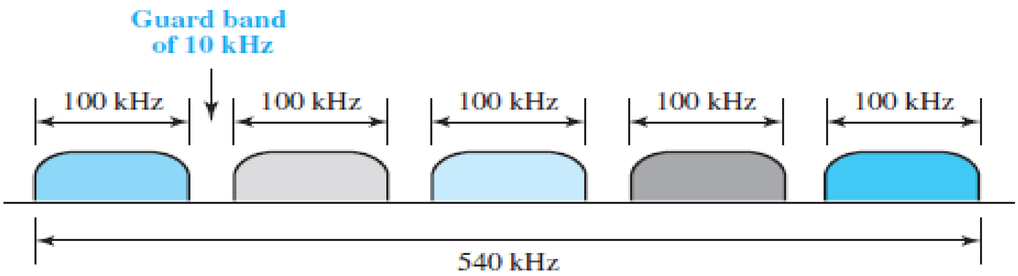
Figure: De-multiplexing Process:



Question: Five channels, each with a 100-kHz bandwidth, are to be multiplexed together. What is the minimum bandwidth of the link if there is a need for a guard band of 10 kHz between the channels to prevent interference? (4 Marks)

Answer:

- **For five channels, we need at least four guard bands. This means that the required bandwidth is at least**
  - **$5 \times 100 + 4 \times 10 = 540$ kHz**



Question: Five channels each with 200kHz bandwidth are multiplexed using FDM. Find minimum bandwidth of the link if guard band of 10kHz is used. (4Marks)

Answer:

- Five channels each with 200 kHz bandwidth are multiplexed using FDM.
- For five channels, we need at least four guard bands.
- Guard Bands of 10 KHz is used.
- This means that the required bandwidth is at least :
    - **5*200+4*10=1040 KHz.**

Advantages of FDM:

1. A large number of signals (channels) can be transmitted simultaneously.
2. FDM does not need synchronization between its transmitter and receiver for proper operation.
3. Demodulation of FDM is easy.
4. Due to slow narrow band fading only a single channel gets affected.

Disadvantages of FDM:

1. The communication channel must have a very large bandwidth.
2. Intermodulation distortion takes place.
3. Large number of modulators and filters are required.
4. FDM suffers from the problem of crosstalk.
5. All the FDM channels get affected due to wideband fading.

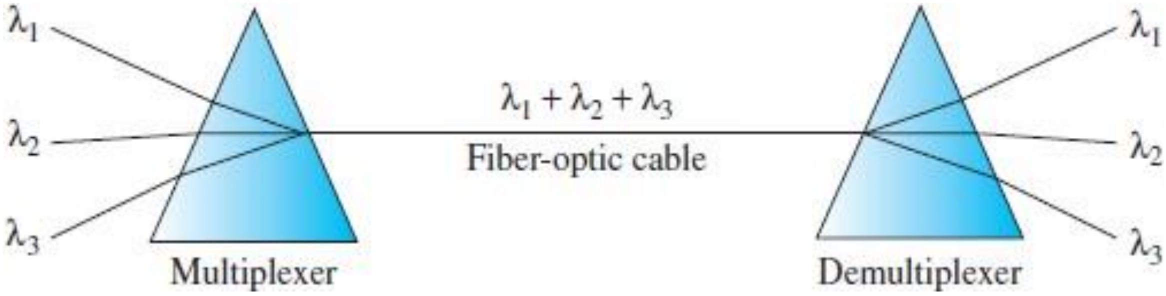
Applications of FDM

- FDM is used for FM & AM radio broadcasting.
- FDM is used in television broadcasting.
- First generation cellular telephone also uses FDM.

Wavelength-Division Multiplexing

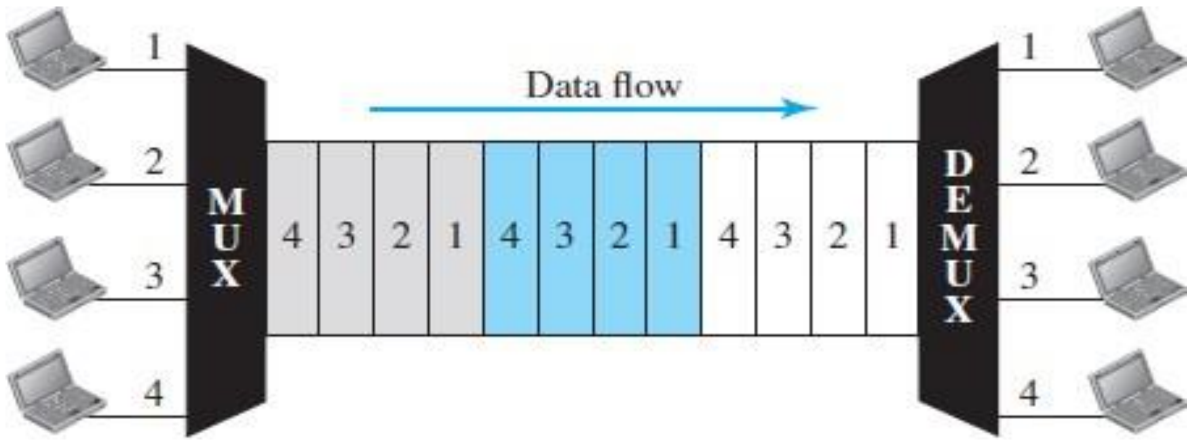
- Wavelength-division multiplexing (WDM) is designed to use the high-data-rate capability of fiber-optic cable.
- The optical fiber data rate is higher than the data rate of metallic transmission cable, but using a fiber-optic cable for a single line wastes the available bandwidth.
- WDM is conceptually the same as FDM, except that the multiplexing and Demultiplexing involve optical signals transmitted through fiber-optic channels. The difference is that the frequencies are very high.
- WDM is an analog multiplexing technique.
- In WDM different signals are *optical or light* signals that are transmitted through optical fiber.

- Various light waves from different sources are combined to form a composite light signal that is transmitted across the channel to the receiver.
- At the receiver side, this composite light signal is broken into different light waves by Demultiplexer.
- The Combining and the Splitting of light waves is done by using a PRISM. Prism bends beam of light based on the angle of incidence and the frequency of light wave.



Time Division Multiplexing (TDM):

- **TDM** is the digital multiplexing technique.
- **In TDM**, **the channel/link is divided** on the **basis of time**.
- Total time available in the channel is divided between several users. Each user is allotted a particular a time interval called time slot or time slice during which the data is transmitted by that user.
- Thus each sending device **takes control of entire bandwidth** of the channel for fixed amount of time.
- Each user is allotted a particular time interval **called *time slot* or *slice.***
- In TDM the data rate capacity of the transmission medium should be greater than the data rate required by sending or receiving devices.
- All the signals to be transmitted are not transmitted simultaneously. Instead, they are transmitted one-by-one. Thus each signal will be **transmitted for a very short time**. One cycle or frame is said to be complete when all the signals are transmitted once on the transmission channel.
- The TDM system can be used to multiplex analog or digital signals, however it is **more suitable for the digital signal multiplexing**.
- The TDM signal in the form of frames is transmitted on the common communication medium.



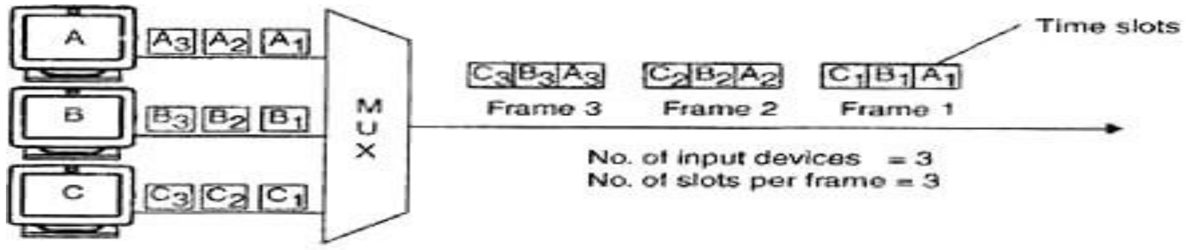
Types of TDM

1. **Synchronous TDM and**
2. **Statistical (Asynchronous) TDM**

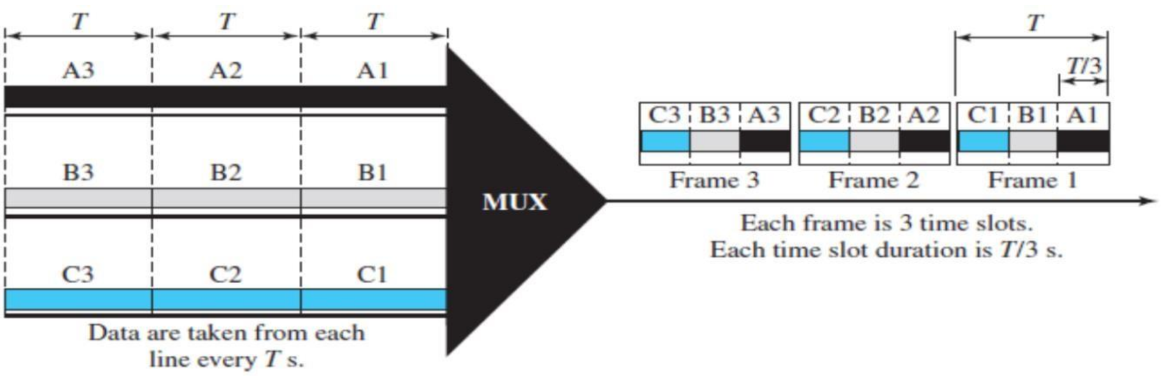
Question: Explain process of synchronous time division multiplexing with its advantages. (4 Marks)

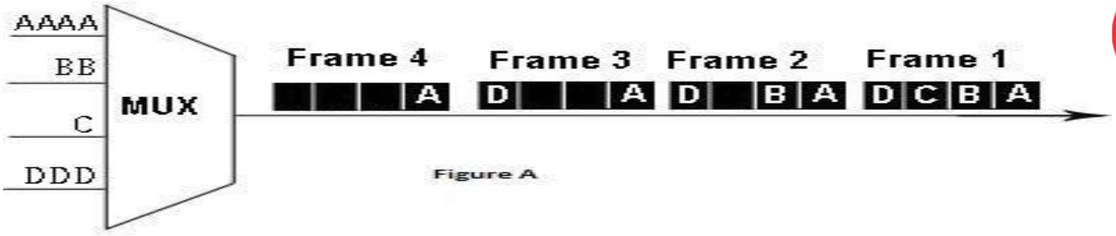
1. Synchronous TDM (STDM)

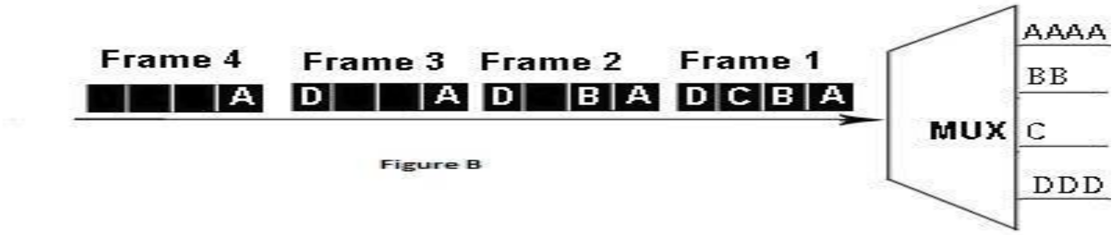
- In synchronous TDM, **each device is given same time slot** to transmit the data over the link, irrespective of the fact that the device has any data to transmit or not. Hence the name Synchronous TDM.
- Synchronous TDM requires that the total speed of various input lines should not exceed the capacity of path.
- Each device places its data onto the link when its **time slot** arrives *i.e.* each device is given the possession of line turn by turn.
- If any device does not have data to send then its **time slot remains empty**.
- The various time slots are organized into **frames** and each frame consists of one or more time slots dedicated to each sending device.
- If there are *n* sending devices, there will be *n* slots in frame *i.e.* one slot for each device. As show in fig, there are 3 input devices, so there are 3 slots in each frame.
- If there is no data to be transmitted, the buffer will be empty but still the turn of the node will come.



Synchronous TDM







Advantages of Synchronous TDM :

- Relatively simple
- An order of data is maintained
- No addressing information is required, channel capacity should be large.
- Commonly used with ISDN (Integrated Services Digital Network).

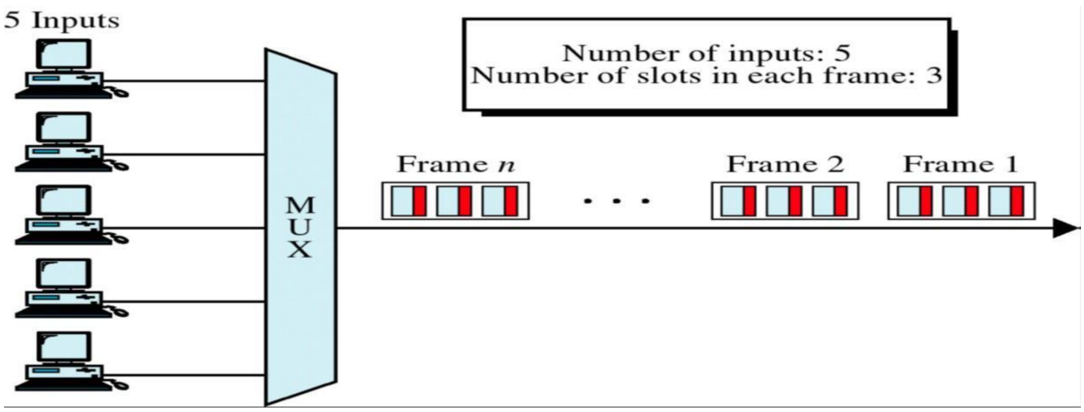
Disadvantages of Synchronous TDM :

1. The channel capacity cannot be fully utilized. Some of the slots go empty in certain frames.
2. The capacity of single communication line that is used to carry the various transmission should be greater than the total speed of input lines.

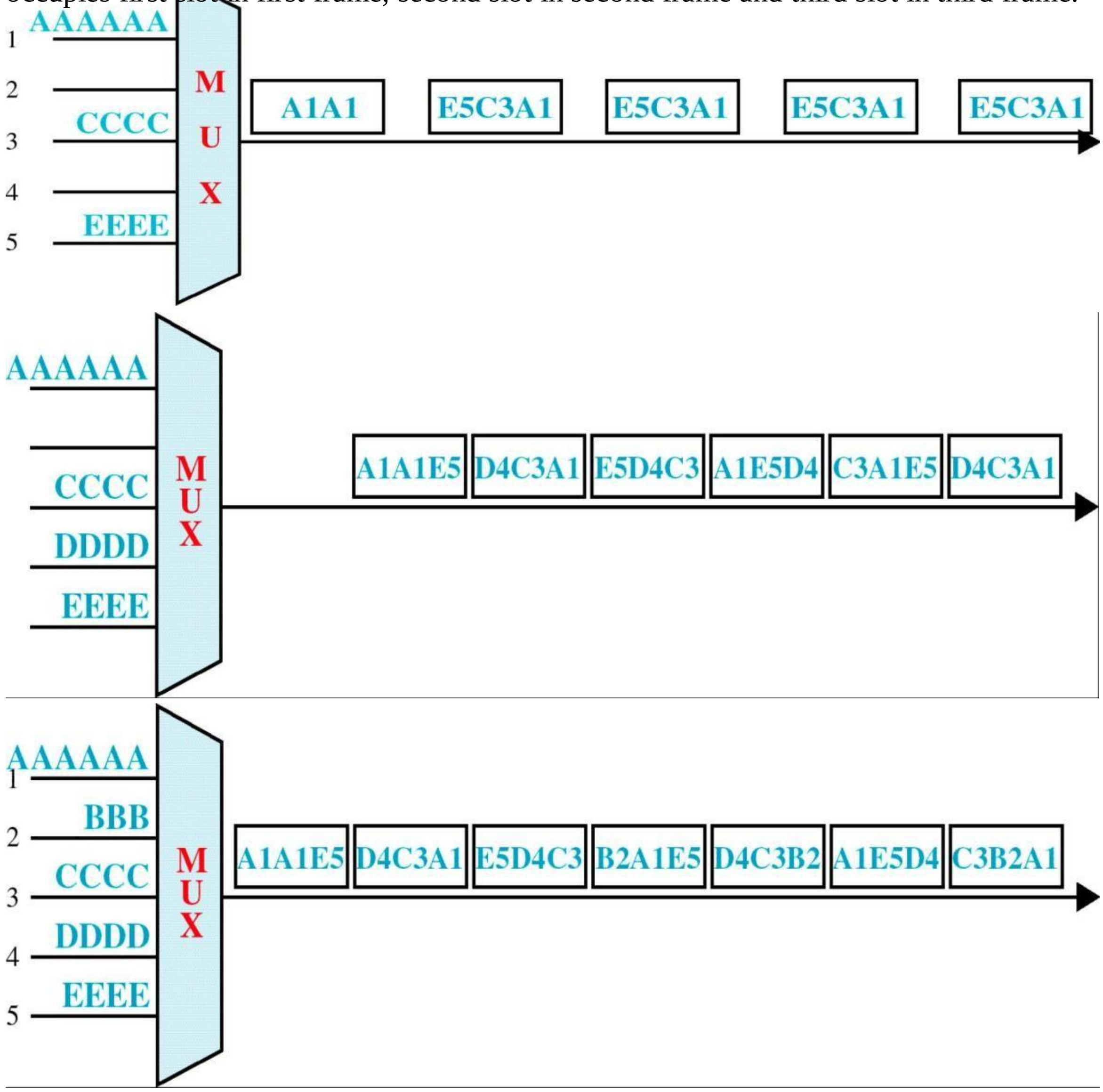
2. Asynchronous TDM or statistical TDM

Question: Explain the process of asynchronous TDM with example.

- It is also known as **statistical** time division multiplexing.
- Asynchronous TDM is called so because is this type of multiplexing, **time slots are not fixed** *i.e.* the slots are flexible. Here, the total speed of input lines can be greater than the capacity of the path.
- In synchronous TDM, if we have *n* input lines then there are *n* slots in one frame. But in asynchronous it is not so. If we have *n* input lines then the frame contains not more than *m* slots, with *m* less than *n* $(m < n)$.
- The number of time slots in a frame is based on a statistical analysis of number of input lines.



- In this system slots are not predefined, the slots are allocated to any of device that has data to send.
- The multiplexer scans the various input lines, accepts the data from the lines that have data to send, fills the frame and then sends the frame across the link.
- If there are not enough data to fill all the slots in a frame, then the frames are transmitted partially filled.
- Asynchronous Time Division Multiplexing is depicted in fig. Here we have five input lines and three slots per frame. In **Case 1**, only three out of five input lines place data onto the link *i.e.* number of input lines and number of slots per frame are same. In **Case 2**, four out of five input lines are active. Here number of input line is one more than the number of slots per frame.
- In **Case 3**, all five input lines are active. In all these cases, multiplexer scans the various lines in order and fills the frames and transmits them across the channel.
- The distribution of various slots in the frames is not symmetrical. In case 2, device 1 occupies first slot in first frame, second slot in second frame and third slot in third frame.



Advantages of TDM :

1. Full available channel bandwidth can be utilized for each channel.
2. lnter modulation distortion is absent.
3. TDM circuitry is not very complex.
4. The problem of crosstalk is not severe.

Disadvantages of TDM :

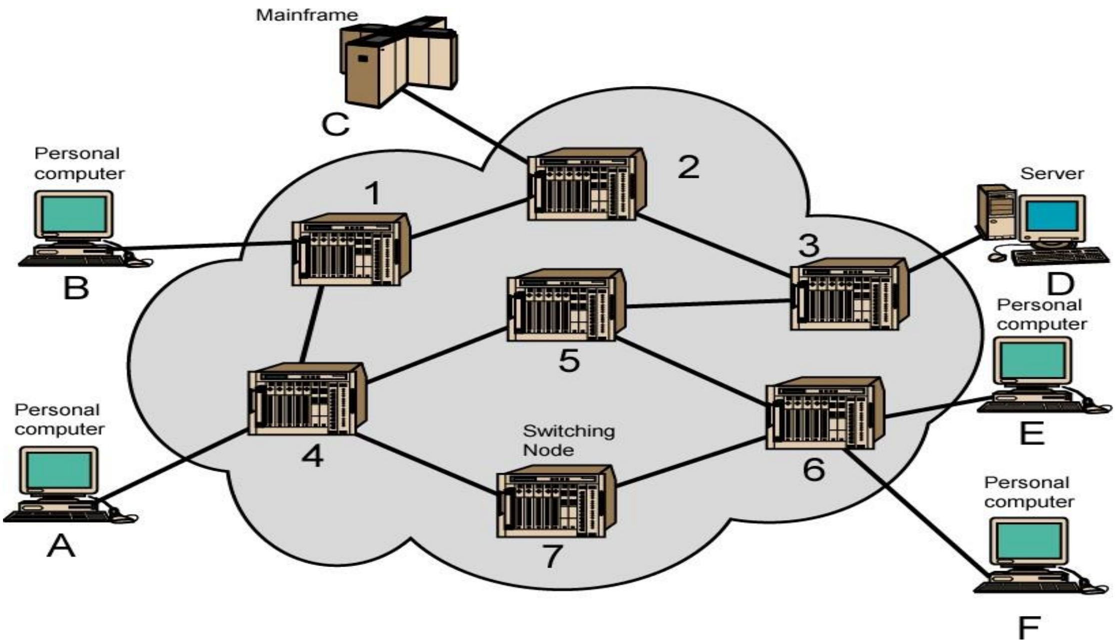
1. Synchronization is essential for proper operation.
2. Due to slow narrowband fading, all the TDM channels may get wiped out.

Comparison of FDM and TDM

PARAMETER	TDM	FDM
Definition	TDM is the transmission technique in which different signal are transmitted over a common channel and each signal occupies entire range of bandwidth in the time domain.	FDM is the transmission technique in which different signal are transmitted over a common channel and each signal occupies different slot within that bandwidth of the frequency domain.
Stands For	Time-Division Multiplexing	Frequency-Division Multiplexing
Useful for	TDM can be used for both Analog and Digital signals.	FDM can be used for Analog signals only.
Synchronization	TDM requires Synchronization.	not required Synchronization.
Circuit	circuitry is very simple to built.	FDM circuitry is very complex.
Cross Talk	TDM is not sensitive for Cross Talk (Noise Immunity)	FDM suffers from the cross talk immunity due to Bandpass Filter.
Requirement	TDM requires sync pulse for its operation.	FDM requires Guard bands for its operation.
Effiecient	TDM is more efficient and is widely used technique in multiplexing.	FDM is less efficient compared to TDM.
Applications	TDM is used in Pulse code modulation.	FDM is used in TV and RADIO broadcasting.

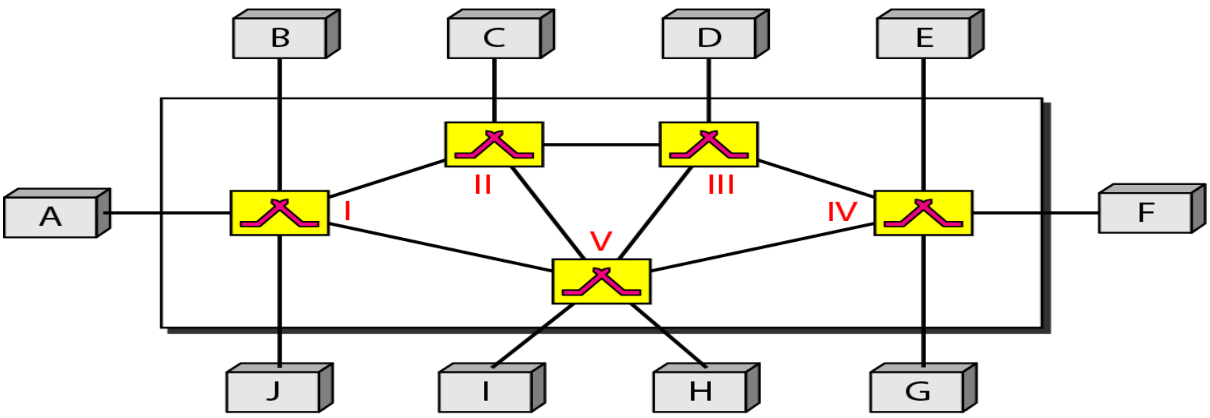
Switched network

- Transmission of data beyond a local area, communication is typically achieved by transmitting data from source to destination through a network of intermediate switching nodes.
- Networks are used to interconnect many devices or stations.
- The stations may be computers, terminals, telephones, or other communicating devices.
- Long distance transmission between stations is typically done over a network of **switching nodes**.
- **Switching nodes** do **not concern with content of data**. **Their purpose is to** provide a switching facility that will **move the data from node to node** until they reach their destination (the end device).
- In a switched communications network, data entering the network from a station are **routed** to the destination by being switched from node to node.

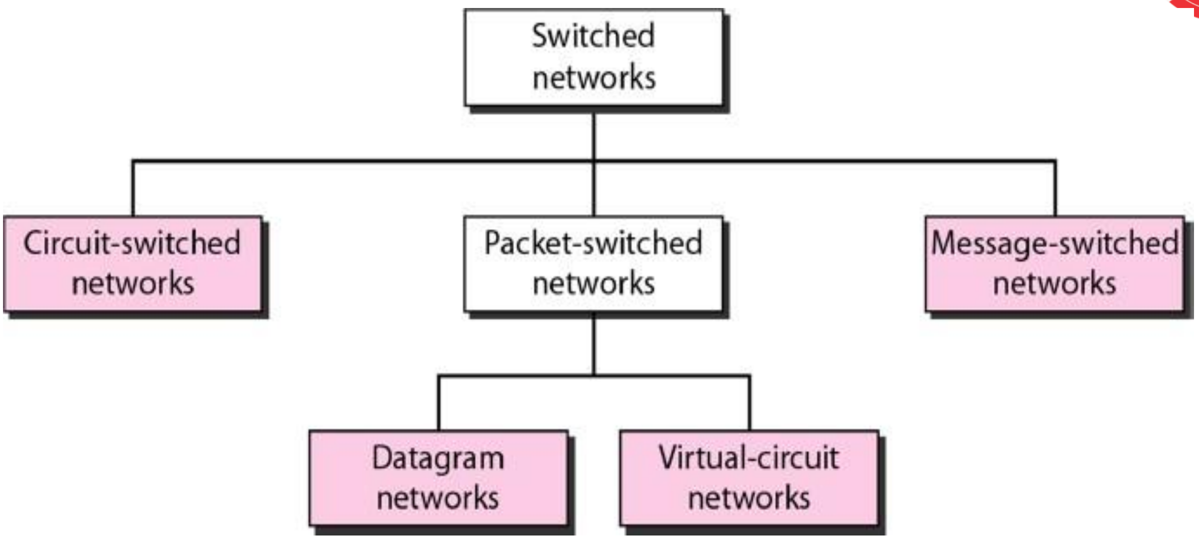


Switching Nodes:

- Nodes may connect to other nodes, or to some stations.
- Network is usually partially connected
    - there is not a direct link between every possible pair of nodes.
- However, some redundant connections are desirable for reliability

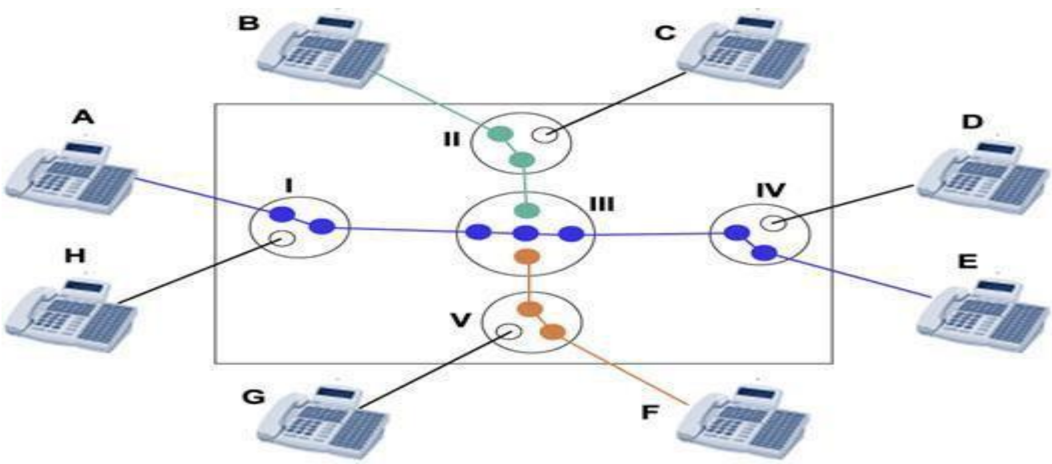


Types of Switching



Circuit Switching

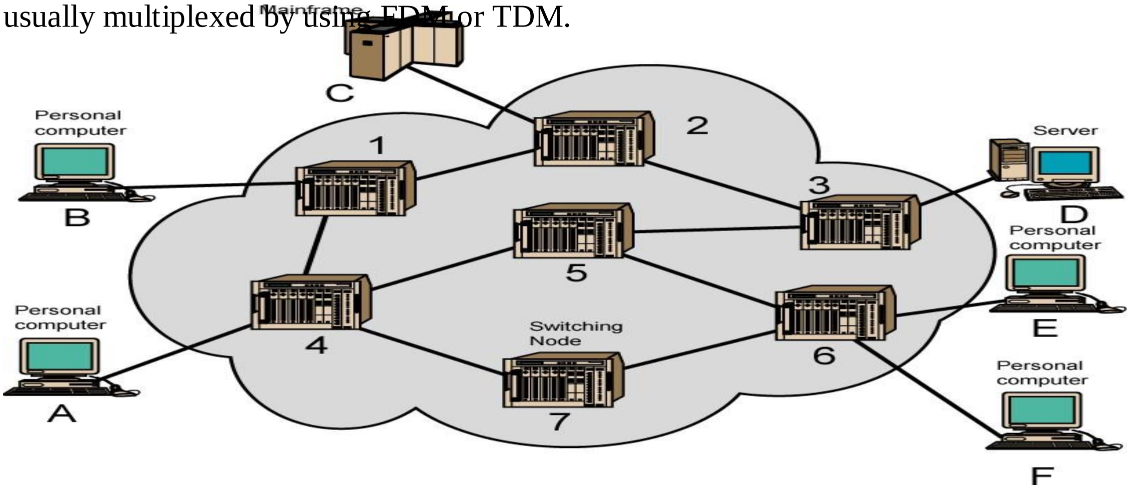
- Circuit Switching is used in public telephone networks.
- Telephone network provides telephone service which involves the two way, real-time transmission of voice signals across a network.
- The network connection allows electrical current and the associated voice signal to flow between the two users.
- These networks are **connection oriented** because they require setting up of a connection before the actual transfer of information can take place.
- The transfer mode of a network that involves setting up a dedicated end to end connection is called **Circuit Switching.**
- Communication via circuit switching has **three phases:**
    1. **Circuit establishment (link by link)**
        - Routing & resource allocation (FDM or TDM)
    2. **Data transfer**
    3. **Circuit disconnect**
        - Deallocate the dedicated resources



Phases of Operation in Circuit Switching

Communication via Circuit switching takes place over three phases of operation:

1. **Circuit Establishment** – In a circuit switching network, before any signal is transmitted, it is necessary to establish an end-to-end link. The node to node links are usually multiplexed by using FDM or TDM.



   - For example consider above figure, station A sends a request to node-4 requesting a connection to station E.
   - Typically, the link from A to 4 is a dedicated line node 4 must find the next route leading to E node 4 selects the link to node 5 and so on then sends a message requesting connection to E.
   - Thus, a dedicated path has been established from A-4-5-6-E

2. **Data Transfer -**After establishing a connection actual transfer of information can take place. It can be analog or digital depending on the nature of network.
   - Data can now be transmitted from A through the network to E.
   - The path is A-4 link, internal switching through 4, 4-5 channel, internal switching through 5, 5-6 channel, internal switching through 6, 6-E link. Generally, the connection is full duplex.

3. **Circuit disconnect (Teardown) :** After some time the connection between two users is terminated usually by the action of one or two stations.Signals must be propagated to nodes 4, 5, and 6 to deallocate the dedicated resources.

Advantages :

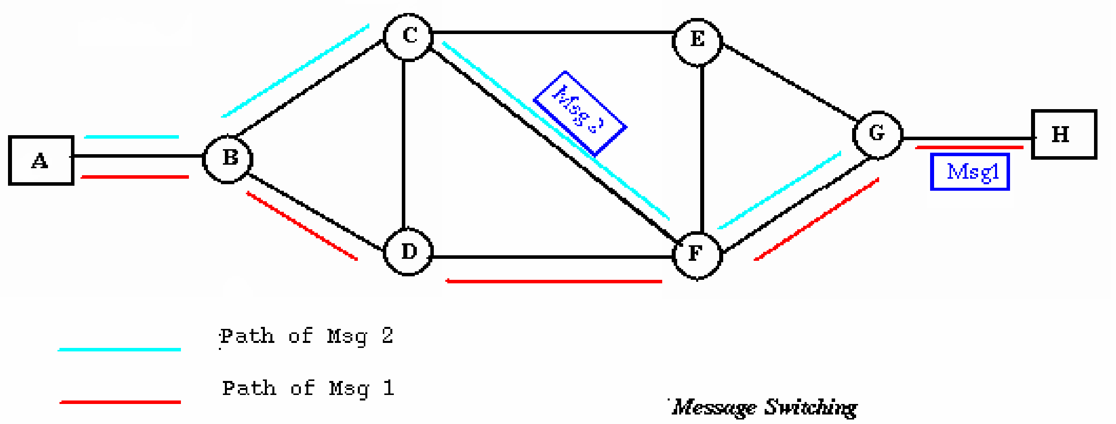
- The dedicated transmission channel provides a guaranteed data rate.
- Because of dedicated path there is no delay in data flow.
- This method is **suitable for long continuous transmission**.

Disadvantages :

- Since the connection is dedicated it cannot be used to transmit any other data even if the channel is free.
- Dedicated channels require more bandwidth.
- It takes more time to establish connection.

Message Switching

- With message switching there is no need to establish a dedicated path between two stations.
- When a station sends a message, the destination address is appended to the message.
- The message is then transmitted through the network, in its entirety, from node to node.
- Each node receives the entire message, stores it in its entirety on disk, and then transmits the message to the next node.
- This type of network is called a store-and-forward network.



Message Switching

- A message-switching node is typically a computer.
- The device needs sufficient secondary-storage capacity to store the incoming messages.
- A time delay is introduced using this type of scheme due to store- and-forward time, plus the time required to find the next node in the transmission path.

Advantages:

- Channel efficiency can be greater compared to circuit switched systems, because more devices are sharing the channel.
- Traffic congestion can be reduced, because messages may be temporarily stored in route.
- Message priorities can be established due to store-and-forward technique.
- Message broadcasting can be achieved with the use of broadcast address appended in the message.

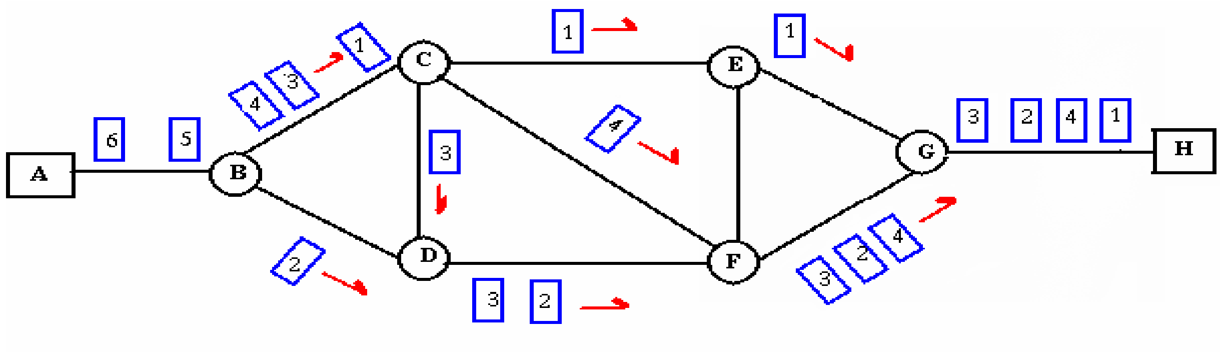
Disadvantages

- Message switching is not compatible with interactive applications.
- Store-and-forward devices are expensive, because they must have large disks to hold potentially long messages.

Packet Switching

- In Packet Switching, messages are broken up into **packets**, each of which includes a header with source, destination and intermediate node address information.
- *Packet switching* can be seen as a solution that tries to combine the advantages of message and circuit switching.
- There are two methods of packet switching:
    - **Datagram and**
    - **virtual circuit.**

- In packet switching methods, a message is broken into small parts, called packets.
- Each packet is tagged with appropriate source and destination addresses.
- Since packets have a strictly defined maximum length, they can be stored in main memory instead of disk, therefore access delay and cost are minimized.
- Also the transmission speeds, between nodes, are optimized.
- With current technology, packets are generally accepted onto the network on a first-come, first-served basis.



Packet Switching

Advantages:

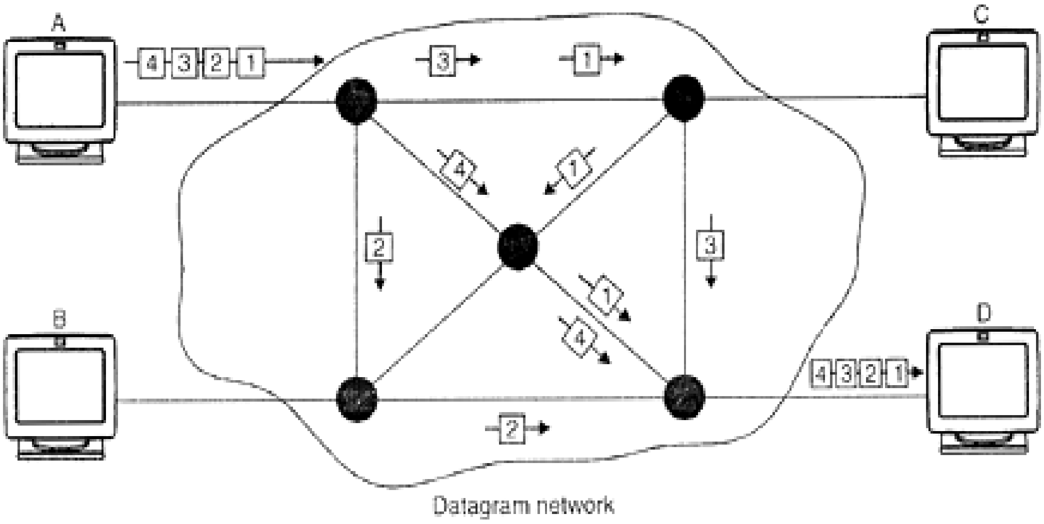
- Better utilization of the network segments in terms of the usage of the network path.
- If a certain link goes down during the transmission, the remaining packets can be sent through another route.
- Since many users can share transmission resources efficiently, the cost of intermittent data communication is reduced.

Disadvantages:

- Variable transmission delays caused by packet processing and packet queues at packet switches.
- Some packet-switching networks support variable packet sizes; this contributes to longer packet processing times at packet switches.
- Sometimes packet may not arrive at their destination in the order in which they were originally transmitted

Datagram packet Switching

- Each message is divided into a stream of packets. Each packet is separately addressed and treated as an independent unit with its own control instructions.
- The switching devices route each packet independently through the network, with each intermediate node determining the packet's next route segment.
- Before transmission starts, the sequence of packets and there are established by the exchange of control information between the sending terminal, the network and the receiving terminal.
- Resources are not allocated for any packet so there is no reserved bandwidth.
- The switches in datagram network are referred to as **routers**.
- No dedicated connection is established between the sender and the receiver, so this network is called as **connectionless** network.
- Datagram packet switching operates at network layer



Datagram packet switching

Advantages:

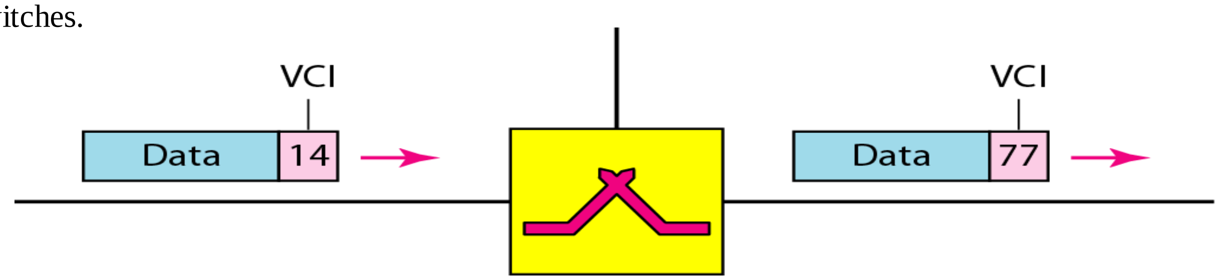
- No call setup phase required.
- More flexible because routing can be used to avoid congested port of the network.
- Cheaper in cost.

Disadvantages:

- Packets are forwarded slowly as compare to the Virtual circuit approach.

Virtual Circuit Packet Switching

- It establishes a logical connection between the sending and receiving devices called **Virtual circuit**.
- The sending device starts the conversation by communicating with the receiving device and agreeing as communication parameters, such as maximum message size and the path to be taken.
- Once this virtual circuit is established; the two devices use it for the rest of the conversation.
- All packets travel through the logical connection established between the sending device and the receiving device.
- Similar to circuit switched network, there are **setup** and **teardown** phases along with the **data transfer** phase.
- Virtual circuit is established in the data link layer.
- Virtual Circuit Identifier(VCI) is a small number which is used by a frame between two switches.



Three phases of communication

- A source and destination have to undergo three phases to communicate between each other, they are:

  1. Set up
  2. Data Transfer
  3. Teardown

- **Set up Phase:**
  - In the Set up phase a switch creates an entry for a virtual circuit by following two approaches-

    i) Permanent Virtual Circuit (PVC)
    ii) Switched Virtual Circuit (SVC)

i) Permanent Virtual Circuit (PVC) –

- The PVC is like a leased telephone line between two parties. One party can pick up the phone and talk to the other one without dialing.
- A source and destination choose to have a PVC between them.
- Then the corresponding table entries are recorded for all the switches.

ii) Switched Virtual Circuit (SVC) –

- In SVC a temporary connection is established between the source and destination.
- This connection exists only when the data is to be transferred.

- When source A wants to establish a virtual circuit with destination B then the following two steps are to be followed :
    1. Set up Request   2. Acknowledgement

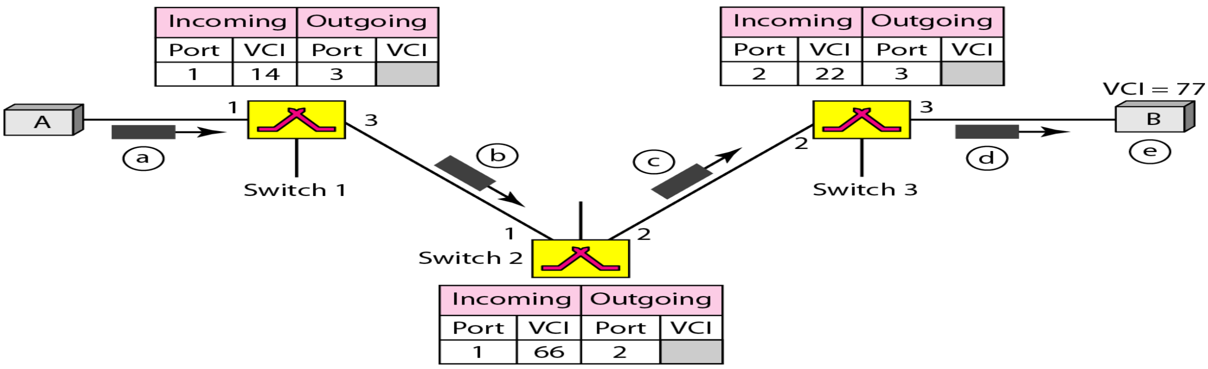


Figure: Setup request in a virtual-circuit network

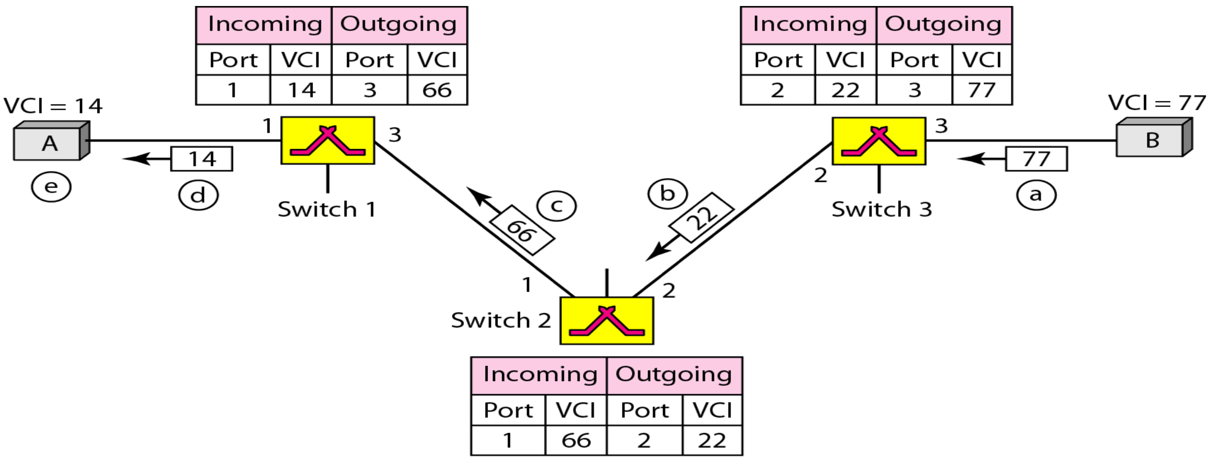


Figure: Setup acknowledgment in a virtual-circuit network

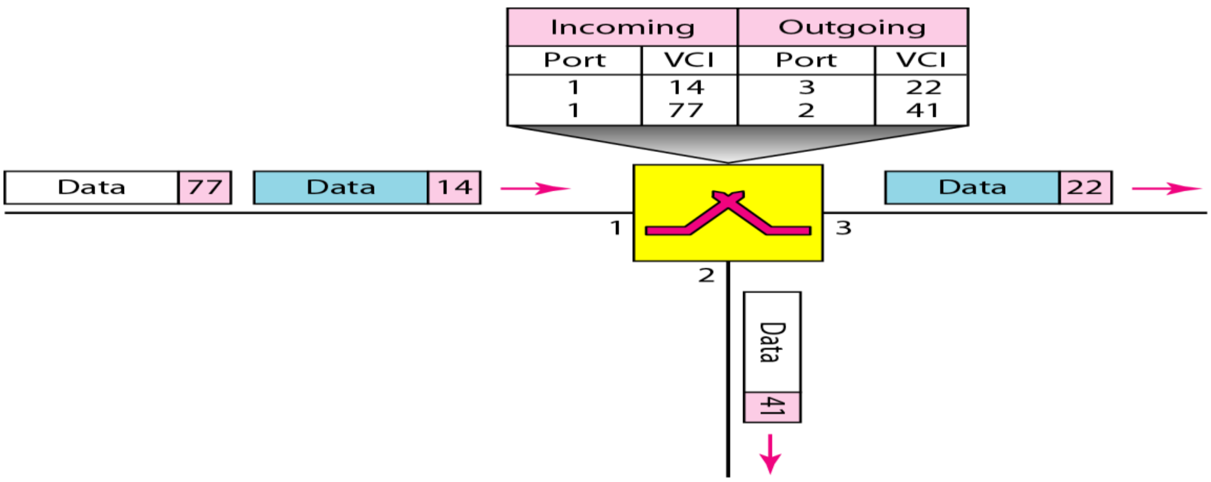


Figure: Switch and tables in a virtual-circuit network

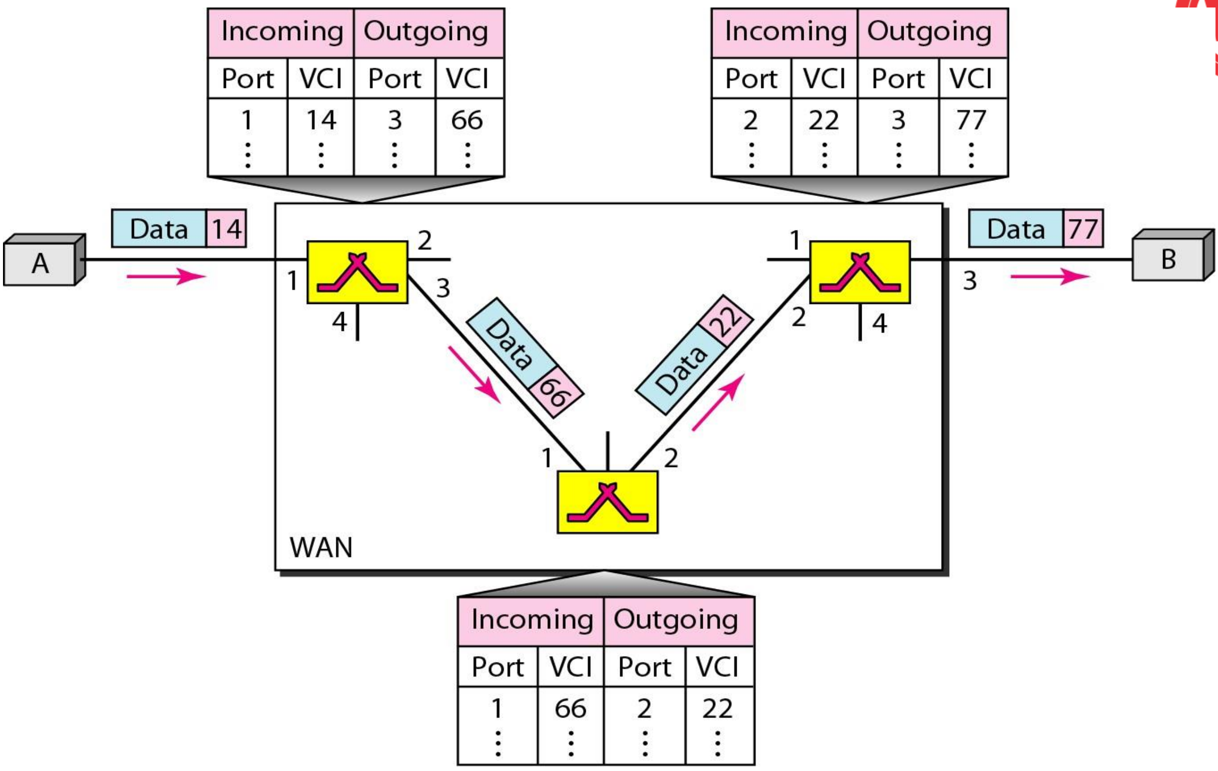


Figure: Source-to-destination data transfer in a virtual-circuit network

Advantages of Virtual circuit Switching:

- Virtual circuit provides packet sequencing and error control.
- Packet forwarding is fast and quick.
- Multiple packets send by the same source to same destination.

Disadvantages of Virtual circuit Switching:

- Loss of a node losses all circuits through that node so its less reliable.
- Less flexible than other approaches.
- Cost is high than Datagram approach

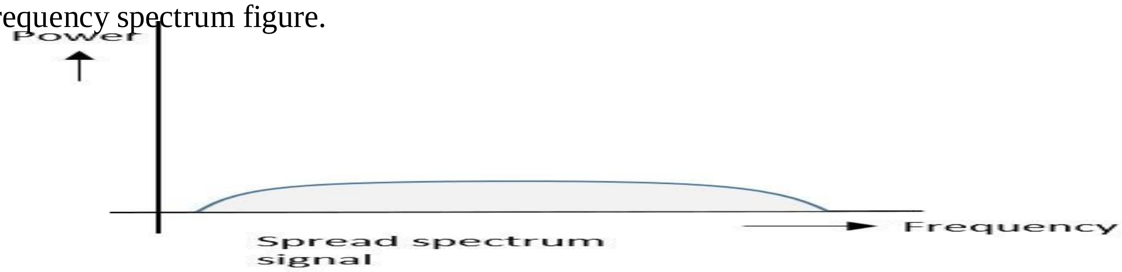
Comparison of Datagram approach and Virtual Circuit Packet Switching:

Sr. No.	Datagram approach	Virtual circuit packet switching
1.	In this approach each packet is considered as a totally independent packet from all others.	In this approach preplanned route established before any packet sent.
2.	More flexible because of routing can be used to avoid congested port of the network.	Less flexible.
3.	Slow in packet forwarding.	Packets are forwarded quickly.
4.	More Reliable	Less reliable because loss of node losses all circuit through that node.

Spread Spectrum

- Spread spectrum is an increasingly important form of encoding for wireless communications. It is used to transmit either analog or digital data, using an analog signal.
- The basic idea of spread spectrum is to modulate the signal so as to increase significantly the bandwidth (spread the spectrum) of the signal to be transmitted.
- It was initially developed for military and intelligence requirements. The use of spread spectrum makes jamming and interception more difficult and provides improved reception.
- The first type of spread spectrum developed is known as **frequency hopping**. A more recent type of spread spectrum is **direct sequence**. Both of these techniques are used in various wireless communications standards and products.

➢ The spread spectrum signals have the signal strength distributed as shown in the following frequency spectrum figure.



➢ Following are some of its features −
  - Band of signals occupy a wide range of frequencies.
  - Power density is very low.
  - Energy is wide spread.

➢ With these features, the spread spectrum signals are highly resistant to interference or jamming.

General Model of Spread Spectrum System

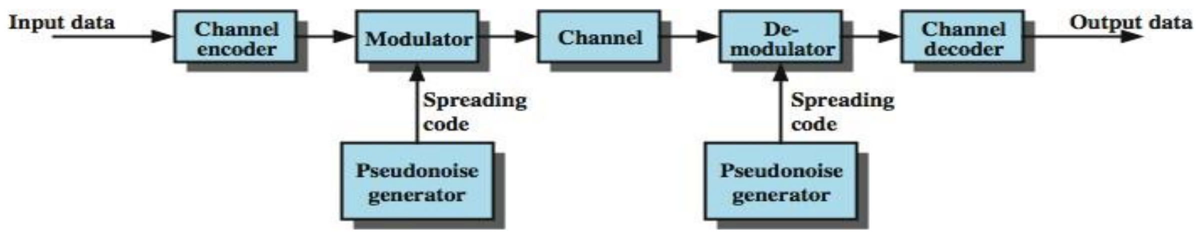


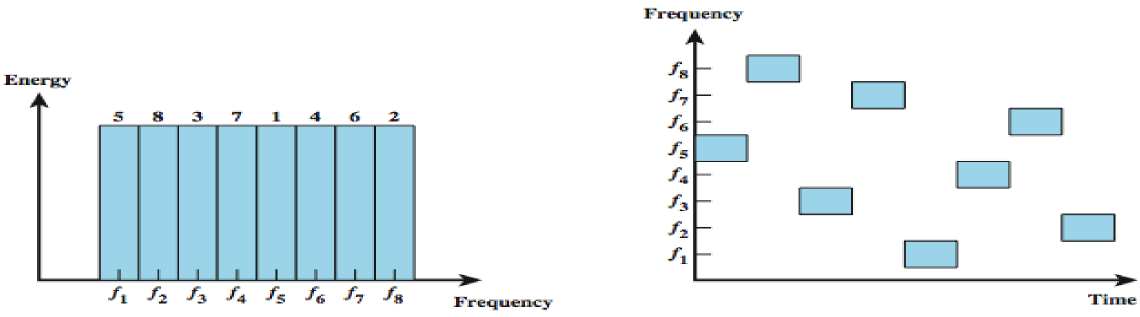
Figure highlights the key characteristics of any spread spectrum system. Input is fed into a channel encoder that produces an analog signal with a relatively narrow bandwidth around some center frequency. This signal is further modulated using a sequence of digits known as a spreading code or spreading sequence. Typically, but not always, the spreading code is generated by a pseudo noise, or pseudorandom number, generator. The effect of this modulation is to increase significantly the bandwidth (spread the spectrum) of the signal to be transmitted. On the receiving end, the same digit sequence is used to demodulate the spread spectrum signal. Finally, the signal is fed into a channel decoder to recover the data.

Frequency Hopped Spread Spectrum (FHSS)

- This is frequency hopping technique, where the users are made to change the frequencies of usage, from one to another in a specified time interval, hence called as frequency hopping.
- For example, a frequency was allotted to sender 1 for a particular period of time. Now, after a while, sender 1 hops to the other frequency and sender 2 uses the first frequency, which was previously used by sender 1. This is called as frequency reuse.
- The frequencies of the data are hopped from one to another in order to provide a secure transmission. The amount of time spent on each frequency hop is called as Dwell time.
- With frequency-hopping spread spectrum (FHSS), the signal is broadcast over a seemingly random series of radio frequencies, hopping from frequency to frequency at fixed intervals.
- A receiver, hopping between frequencies in synchronization with the transmitter, picks up the message.
- Would-be eavesdroppers hear only unintelligible blips. Attempts to jam the signal on one frequency succeed only at knocking out a few bits of it.

Example:

- Following Figure shows an example of a frequency-hopping signal.
- A number of channels are allocated for the FH signal. Typically, there are 2^k carrier frequencies forming 2^k channels.
- The spacing between carrier frequencies and hence the width of each channel usually corresponds to the bandwidth of the input signal.
- The transmitter operates in one channel at a time for a fixed interval; for example, the IEEE 802.11 standard uses a 300-ms interval. During that interval, some number of bits (possibly a fraction of a bit, as discussed subsequently) is transmitted using some encoding scheme.
- A spreading code dictates the sequence of channels used. Both transmitter and receiver use the same code to tune into a sequence of channels in synchronization.



(a) Channel assignment (b) Channel use

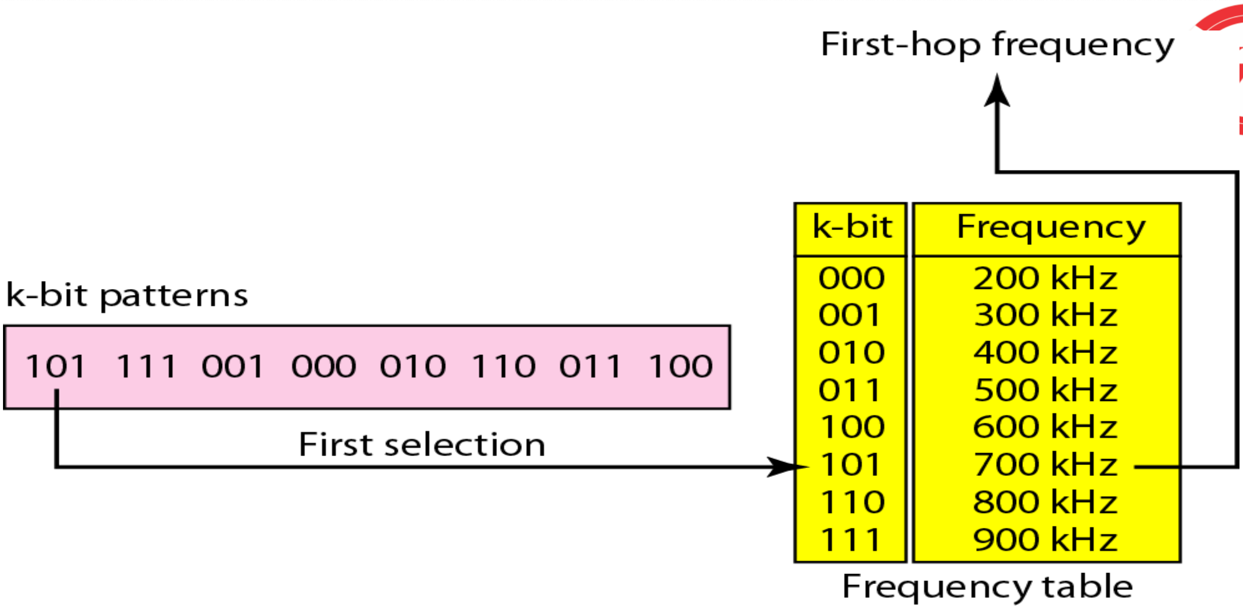


Figure: Frequency Selection

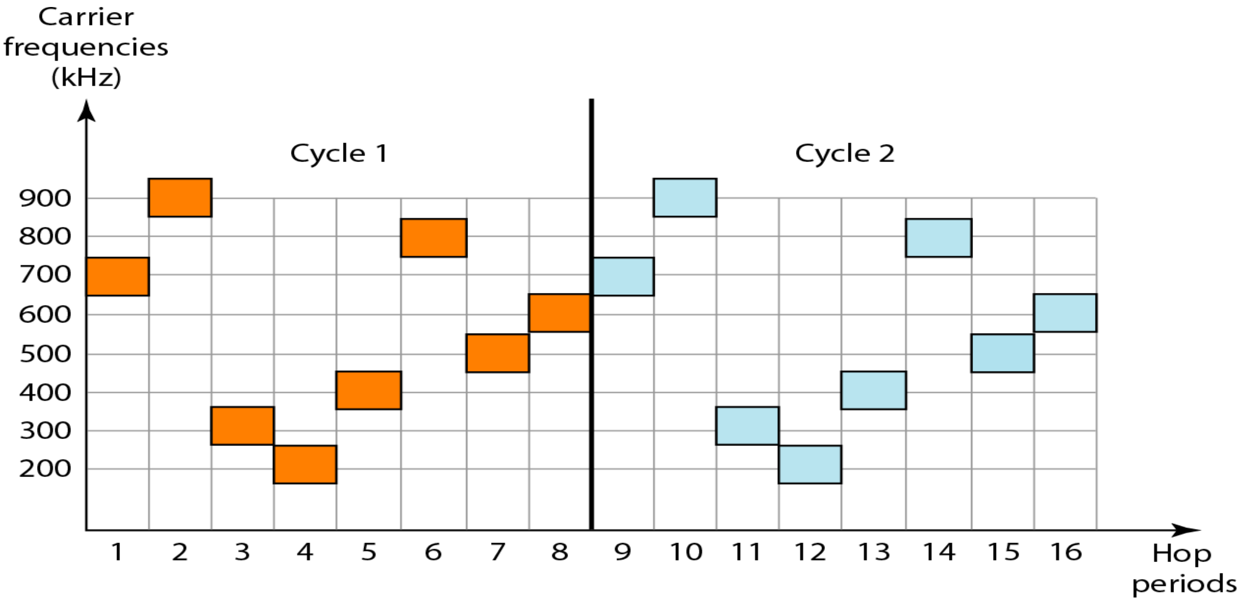


Figure: FHSS Cycle

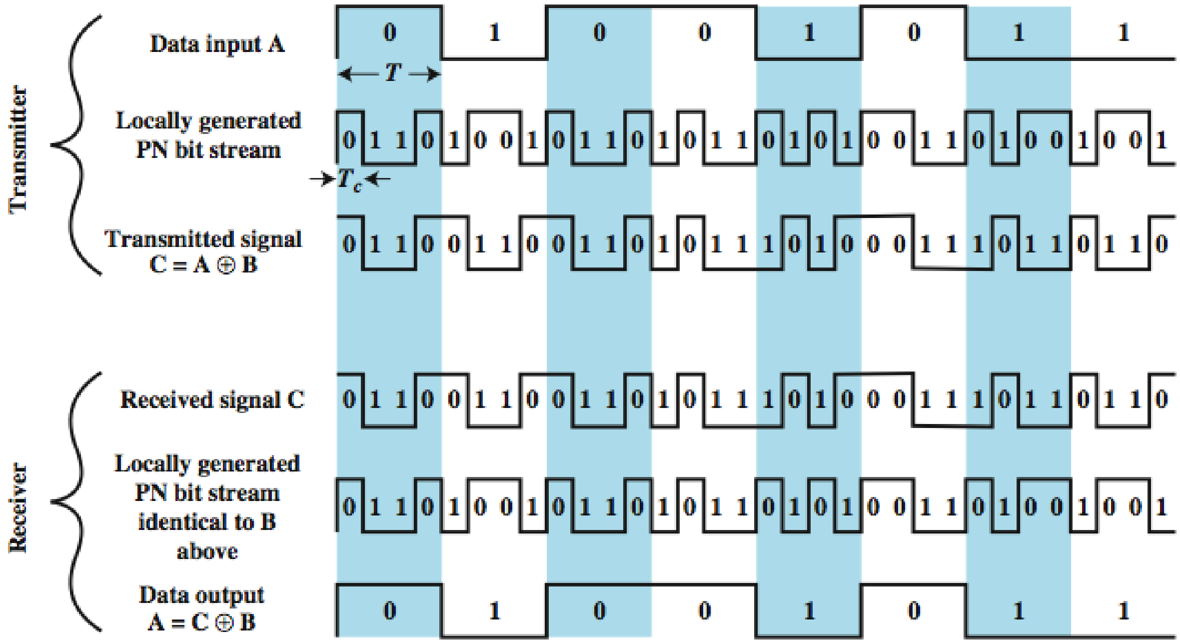
Direct Sequence Spread Spectrum (DSSS)

- Whenever a user wants to send data using this DSSS technique, each and every bit of the user data is multiplied by a secret code, called as **chipping code**.
- This chipping code is nothing but the spreading code which is multiplied with the original message and transmitted. The receiver uses the same code to retrieve the original message.
- With direct sequence spread spectrum (DSSS), each bit in the original signal is represented by multiple bits in the transmitted signal, using a spreading code. The spreading code spreads the signal across a wider frequency band in direct proportion to the number of bits used. Therefore, a 10-bit spreading code spreads the signal across a frequency band that is 10 times greater than a 1-bit spreading code.

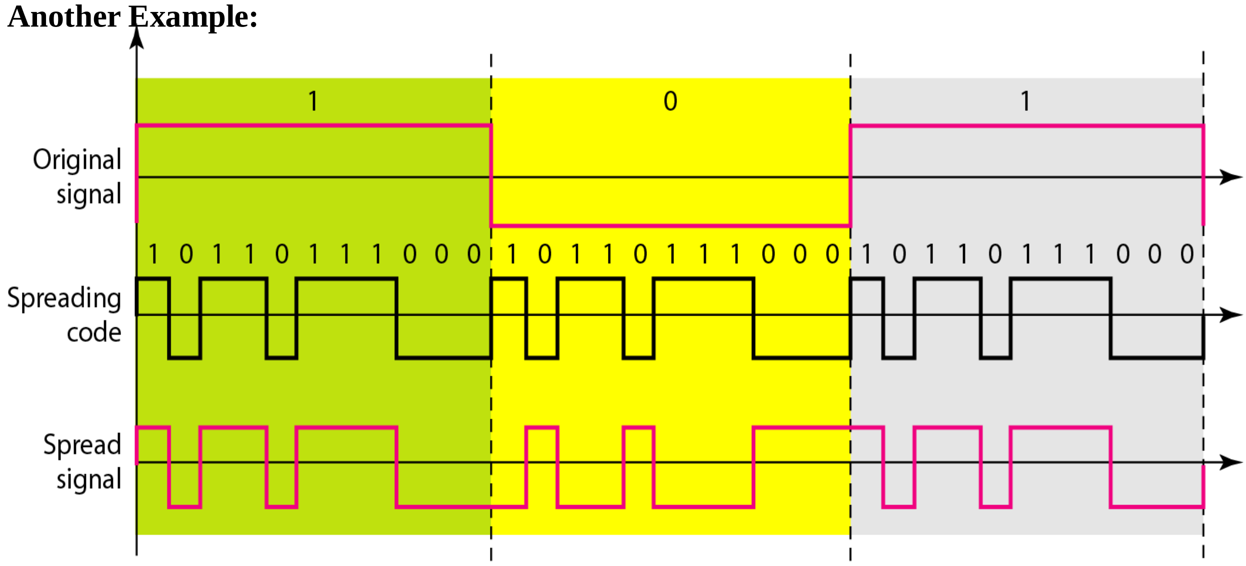
Direct Sequence Spread Spectrum Example

One technique with direct sequence spread spectrum is to combine the digital information stream with the spreading code bit stream using an exclusive-OR (XOR).

Figure shows an example. Note that an information bit of one inverts the spreading code bits in the combination, while an information bit of zero causes the spreading code bits to be transmitted without inversion. The combination bit stream has the data rate of the original spreading code sequence, so it has a wider bandwidth than the information stream. In this example, the spreading code bit stream is clocked at four times the information rate.



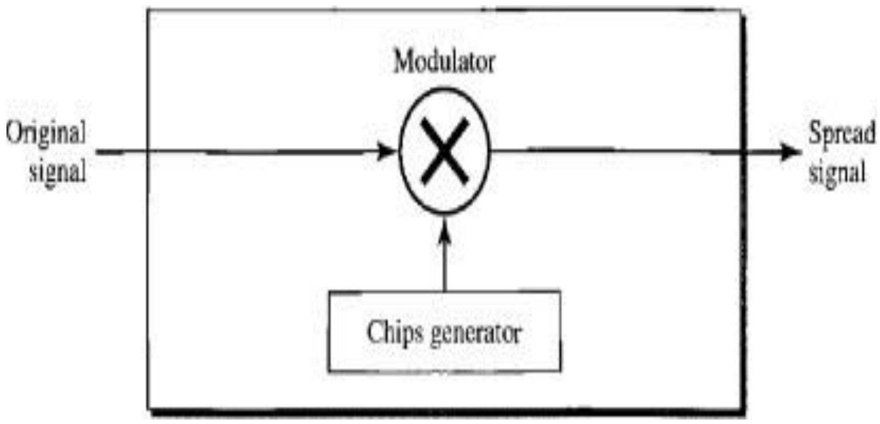
Another Example:



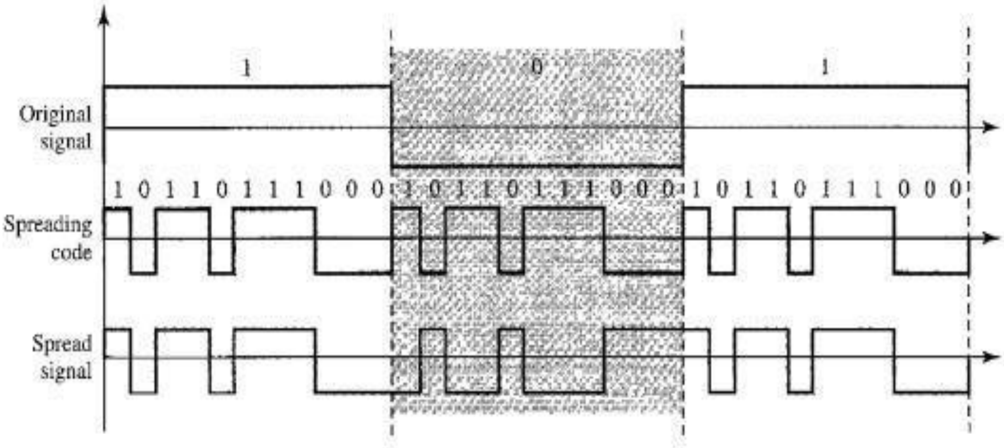
FHSS	DSSS / CDMA
Multiple frequencies are used	Single frequency is used
Hard to find the user's frequency at any instant of time	User frequency, once allotted is always the same
Frequency reuse is allowed	Frequency reuse is not allowed
Sender need not wait	Sender has to wait if the spectrum is busy
Power strength of the signal is high	Power strength of the signal is low
Stronger and penetrates through the obstacles	It is weaker compared to FHSS
It is never affected by interference	It can be affected by interference
It is cheaper	It is expensive
This is the commonly used technique	This technique is not frequently used

Question: Explain DSSS mechanism with neat diagram.

- The direct sequence spread spectrum (DSSS) technique also expands the bandwidth of the original signal, but the process is different.
- In DSSS, we replace each data bit with n bits using a spreading code. In other words, each bit is assigned a code of n bits, called chips, where the chip rate is n times that of the data bit.



- As an example, let us consider the sequence used in a wireless LAN,the famous Barker sequence where n is 11. We assume that the original signal and the chips in the chip generator use polar NRZ encoding. Figure shows the chips and the result of multiplying the original data by the chips to get spread signal.



Question: Compare DSSS with FHSS.

Compare	DSSS	FHSS
Definition	PN sequence of large bandwidth is multiplied with narrow band data signal.	Data bits are transmitted in different frequency slots which are changed by PN sequence.
Modulation method	M-ary FSK	BPSK
Acquisition time	Short	Long
Effect of distance	More	Less

Chapter No: 4

ISO OSI Reference Model

Who made:

- International Standards Organization (ISO)
- A Model of How Protocols and Networking Components Could be Made
- "*Open*" means the concepts are nonproprietary; can be used by anyone.
- OSI is not a protocol. It is a model for understanding and designing a network architecture that is flexible and robust

Open Systems Interconnect (OSI) Model: -

- The OSI model describes how data flows from one computer, through a network to another computer
- The OSI model divides the tasks involved with moving information between networked computers into 7 smaller, more manageable sub-task.
- A task is then assigned to each of the seven OSI layers.
- Each layer is reasonably self-contained so that the tasks assigned to each layer can be implemented independently.

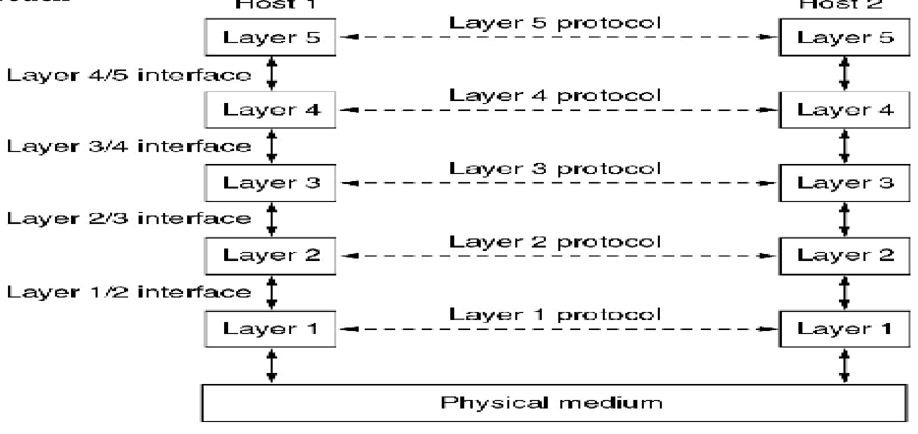
Network Architecture

- A set of layers and protocols is called a network architecture
- It refers to the physical and logical design of a network

Why Layered Architecture?

- Layer architecture simplifies the network design.
- It is easy to debug network applications in a layered architecture network.
- The network management is easier due to the layered architecture.
- Network layers follow a set of rules, called protocol.
- The protocol defines the format of the data being exchanged, and the control and timing for the handshake between layers.

Layered Approach



The entities comprising the corresponding layers on different machines are called peers

- It is the peers that communicate by using the protocols
- Actually, data is not transferred from layer *n* on one machine to layer *n* on another machine
- Each layer passes data and control information to the layer immediately below it, until the lowest layer is reached
- Actual data communication takes place through the lowest layer – the physical layer

Design Issues for the Layers

- Addressing
- Error control
- Order of messages must be preserved
- Flow control – fast sender and slow receiver!
- Disassembling, transmitting, and reassembling large messages
- Multiplexing / de-multiplexing
- Routing

Concept of Services and Protocols:-

- A service is a set of operations that a layer provides to the layer above it
- Service defines what operations the layer is prepared to perform
- A service relates to the interface between two layers – the lower layer is service provider and the upper layer is service user

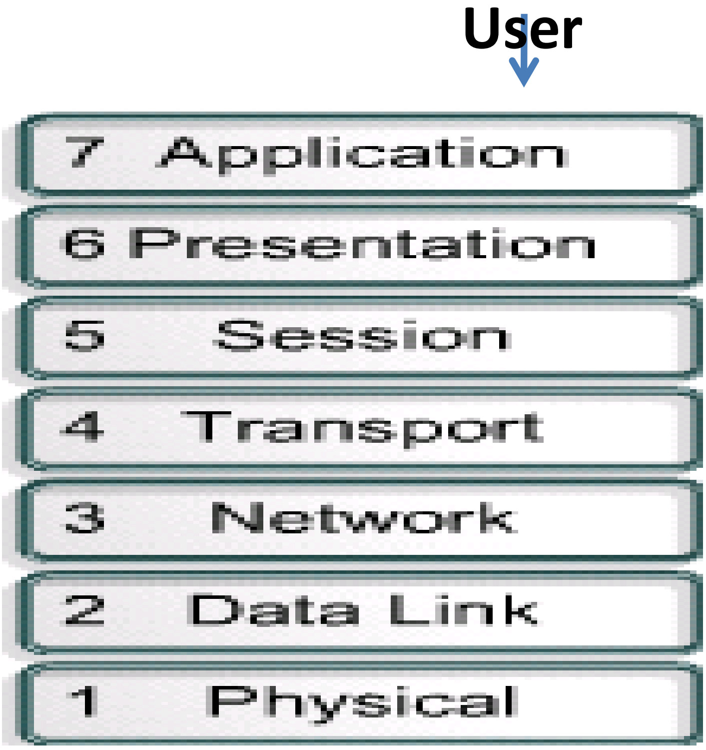
Concept of Services and Protocols

- A protocol is a set of rules governing the format and meaning of the packets
- Protocols relate to packets sent between peer entities on different machines
- Entities use protocols
- Protocols can be changed provided the services visible to the user do not change. Thus services and protocols are completely decoupled

Services and Protocols

- Analogy with programming languages
- A service is like an object in an object oriented language
- What operations can be performed on this object is defined
- How these operations are to be performed is not defined
- Protocol relates to the *implementation* of the service – how it is done

The Layers of the OSI Model

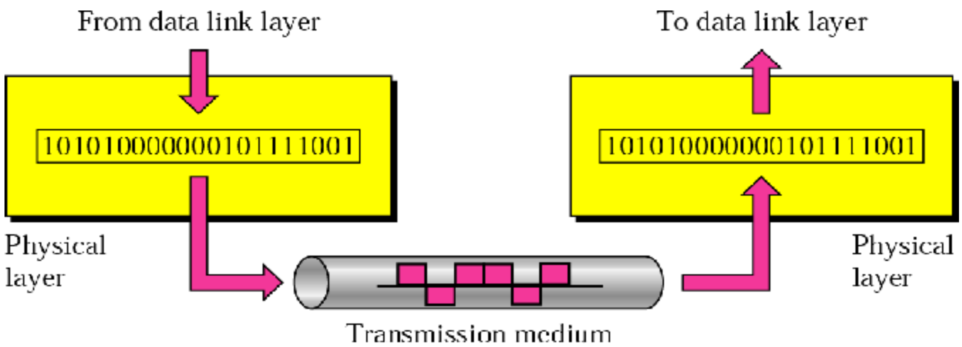


Physical layer:-

- Specifications for the physical components of the network.

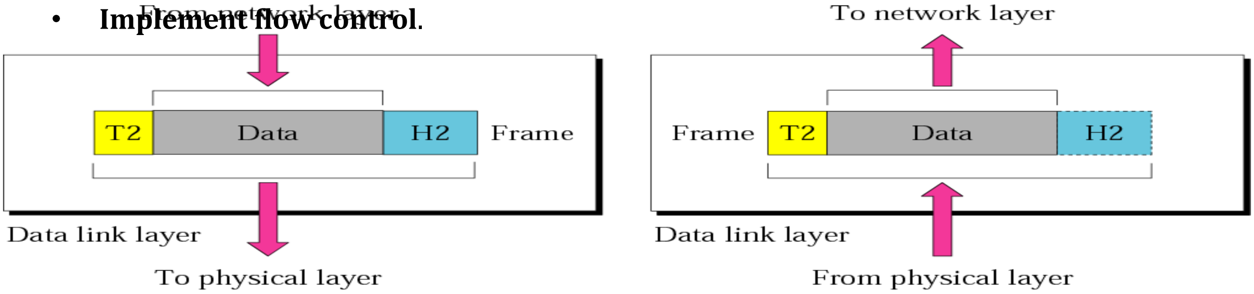
Functions of Physical Layer:

- **Bit representation** – encode bits into electrical or optical signals
- **Transmission rate** – The number of bits sent each second
- Physical characteristics of transmission media
- **Synchronizing** the sender and receiver clocks
- **Transmission mode** – simplex, half-duplex, full duplex
- **Physical Topology** – how devices are connected – ring, star, mesh, bus topology



Data Link Layer

- Data link layer attempts to provide reliable communication over the physical layer interface.
- **Breaks the outgoing data into frames** and **re-assemble the received frames**.
- Create and detect frame boundaries.
- **Handle errors** by implementing an acknowledgement and retransmission scheme.
- **Implement flow control**.



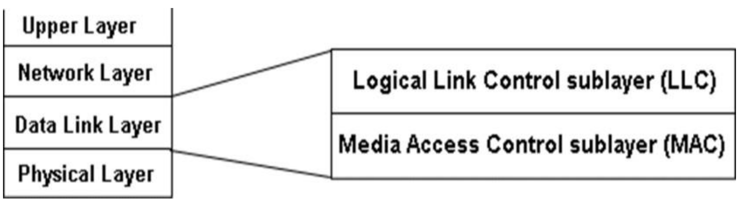
The data link layer is responsible for moving frames from one hop (node) to the next.

Functions of Data Link Layer

- **Framing-**
  - Divides the stream of bits into manageable data units called frames.
- **Physical addressing-**
  - Adds a header to the frame to define the sender and/or receiver of the frame.
- **Flow control-**
  - Imposes a flow control mechanism to avoid overwhelming the receiver. Synchronization between fast sender and slow receiver.
- **Error control-**
  - Adds mechanisms to detect and retransmit damaged or lost frames (CRC).
- **Access control-**
  - Determine which device has control over the link at any given time.
- **Link establishment and termination**:
  - Establishes and terminates the logical link between two nodes.
- **Frame sequencing**:
  - Transmits/receives frames sequentially.
- **Frame acknowledgment**:
  - Provides/expects frame acknowledgments.

DLL is divided into two Sub-Layers

- **LLC Sub Layer**
- **MAC Sub Layer**



Logical Link Control Sub Layer

- It is upper portion of the Data Link layer.
- Performs **Flow control** and **management of connection errors**.
- LLC supports three types of connections:

1. **Unacknowledged connectionless service**:
    - does not perform reliability checks or maintain a connection, very fast, most commonly used
2. **Connection oriented service**:
    - once the connection is established, blocks of data can be transferred between nodes until one of the node terminates the connection.
3. **Acknowledged connectionless service**:
    - provides a mechanism through which individual frames can be acknowledged.

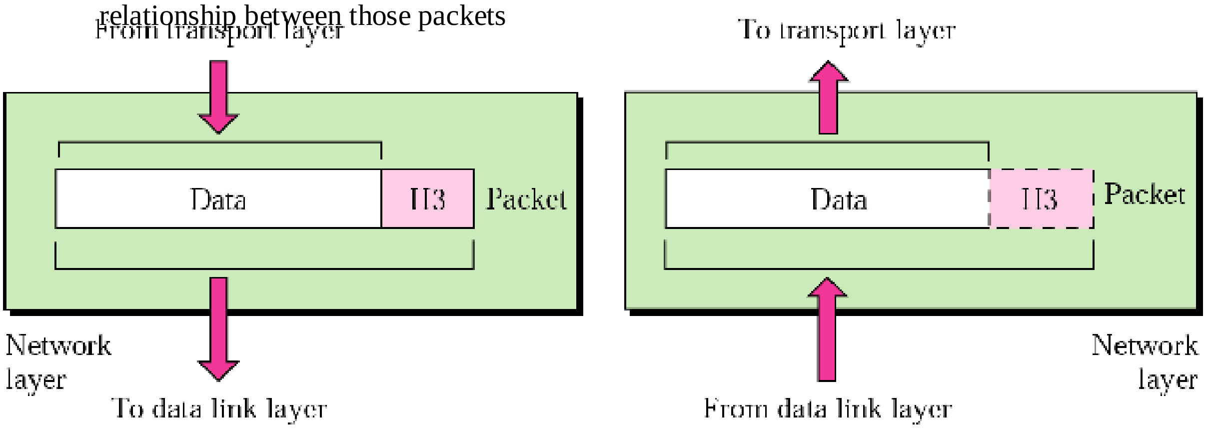
Media Access Control Sub Layer

- This sub layer contains methods to **regulate the timing** of data signals and **eliminate collisions**.
- The MAC sub layer determines where one frame of data ends and the next one starts - **frame synchronization.**
- There are four means of frame synchronization:
    - Time based,
    - Character counting,
    - Byte stuffing and
    - Bit stuffing.

Network Layer:-

Main functions of this layer are:

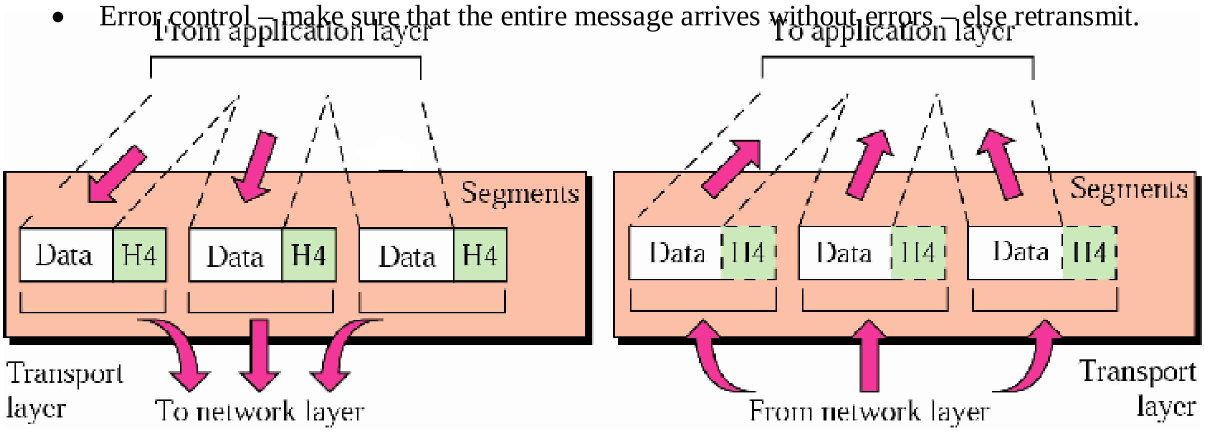
- Responsible for delivery of packets across multiple networks
- Routing – Provide mechanisms to transmit data over independent networks that are linked together.
- Network layer is responsible only for delivery of individual packets and it does not recognize any relationship between those packets



Transport Layer:-

Main functions of this layer are:

- Responsible for source-to destination delivery of the entire message
- Segmentation and reassembly – divide message into smaller segments, number them and transmit. Reassemble these messages at the receiving end.
- Error control – make sure that the entire message arrives without errors – else retransmit.

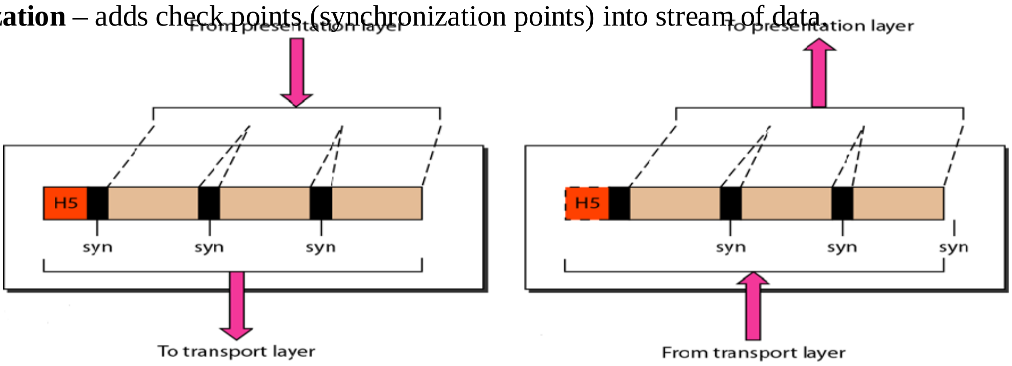


Session Layer:-

Main functions of this layer are:

• **Dialog control** – allows two systems to enter into a dialog, keep a track of whose turn it is to transmit

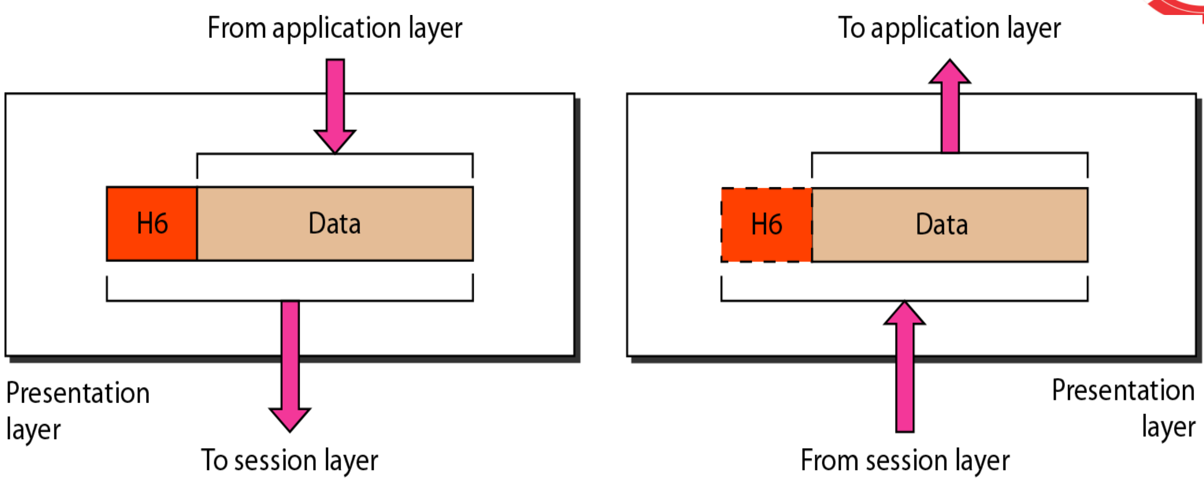
• **Synchronization** – adds check points (synchronization points) into stream of data.



Presentation Layer:-

Responsibilities of this layer are:

- **Translation**
  - Different computers use different encoding systems (bit order translation)
  - Convert data into a common format before transmitting.
  - Syntax represents info such as character codes - how many bits to represent data – 8 or 7 bits
- **Compression** – reduce number of bits to be transmitted Encryption – transform data into an unintelligible format at the sending end for data security
- **Decryption** – at the receiving end

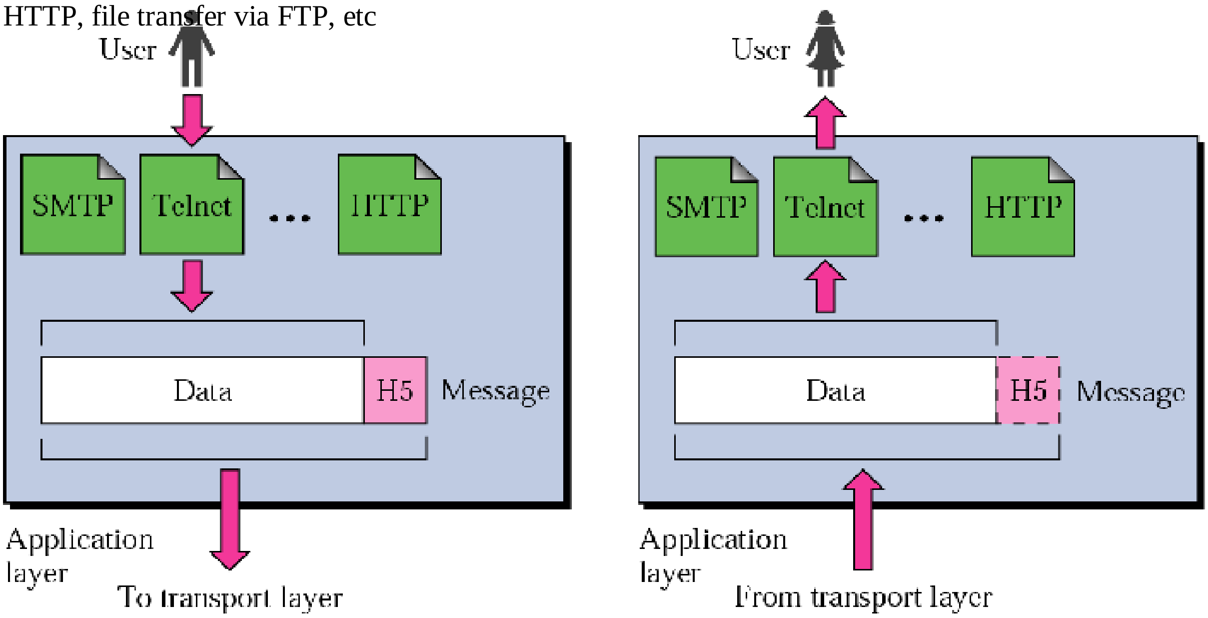


Application Layer:-

•Contains protocols that allow the users to access the network (FTP, HTTP, SMTP, etc)

• Does not include application programs such as email, browsers, word processing applications, etc.

• Protocols contain utilities and network-based services that support email via SMTP, Internet access via HTTP, file transfer via FTP, etc



Data Encapsulation

- The outgoing information will travel down through the layers to the lowest layer.
- While moving down on the source machine, it acquires all the control information which is required to reach the destination machine.
- The control information is in the form of Headers and Trailer which surrounds the data received from the layer above.
- This process of adding headers and trailers to the data is called as **data encapsulation.**

- The information added by each layer is in the form of **headers or trailers**.
- At layer 1 the entire package is converted to a form that can be transferred to the receiving machine.
- At the **receiving machine**, **the message is unwrapped layer by layer**, with each process receiving and removing the data meant for it.
- For example, layer 2 removes the data meant for it, then passes the rest to layer 3.
- Layer 3 then removes the data meant for it and passes the rest to layer 4, and so on.
- The headers and trailers contain control information. The headers and trailers form **the envelope** which carries the message to the desired destination.

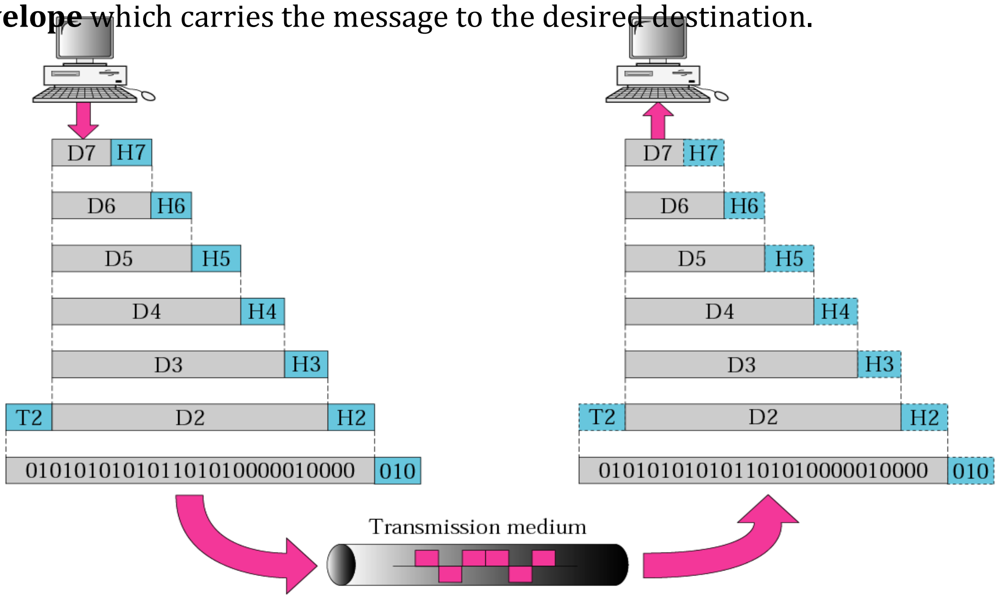


Figure: Data Encapsulation

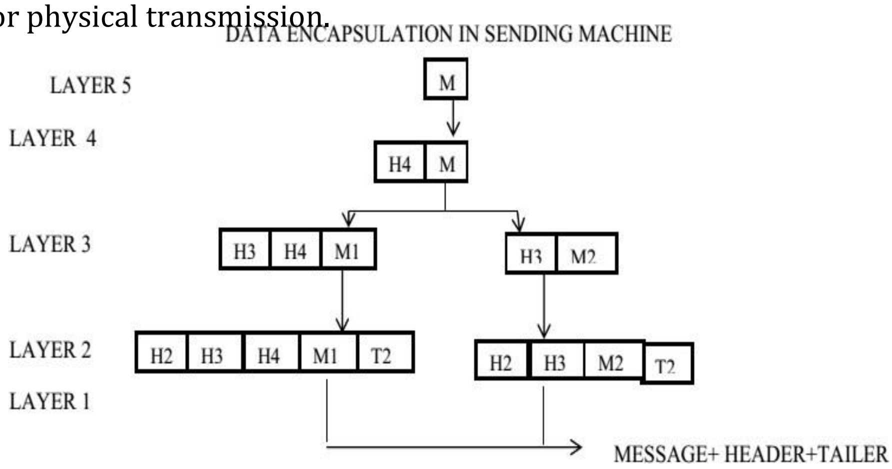
- D7 means the data unit at layer 7, D6 means the data unit at layer 6, and so on.
- The process starts at layer 7 (the application layer), then moves from layer to layer in descending, sequential order.
- At each layer, a **header,** or possibly a **trailer,** can be added to the data unit.
- Commonly, the trailer is added only at layer 2.
- When the formatted data unit passes through the physical layer (layer 1), it is changed into an electromagnetic signal and transported along a physical link.

Example of Data Encapsulation

The figure shows the example of five layer stack for data encapsulation.

- The fifth layer of sending machine wants to send a message M to the fifth layer of destination machine.
- The message M is produced by layer 5 of machine 1 and given to layer 4 for transmission. Layer 4 adds header H4 in front of the message and pass it to layer 3.

- Layer 3 breaks up the incoming message into small units as M1 and M2 and pass these packets to layer 2.
- Layer 2 adds the header as well as footer to each packet obtained from layer 3 and pass it to layer 1 for physical transmission.



Horizontal communication

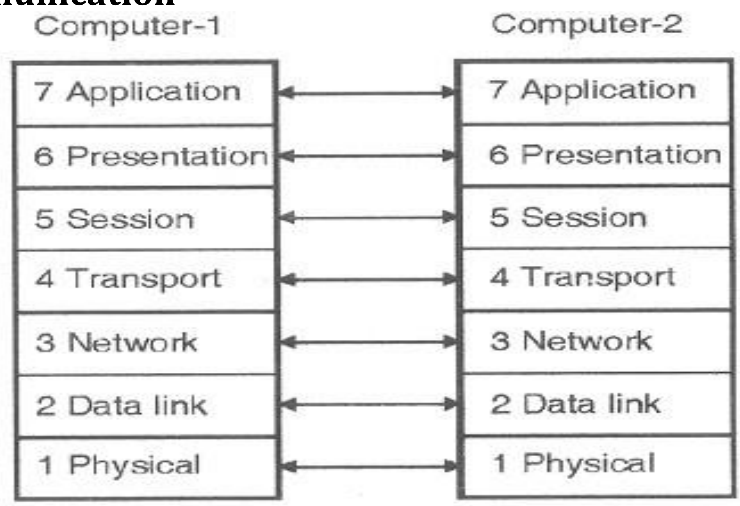


Fig: Horizontal Communication in OSI Model.

1. The horizontal communication is the logical connection between the layers, there is no direct communication between them.
2. Information included in each protocol header by the transmitting system is a message that will be carried to the same protocol in the destination system.
3. For two computers to communicate over a n/w, the protocol used at each layer of the OSI model in the transmitting system must be duplicated at the receiving system.
4. The packet travels up through the protocol stack and each successive header is stripped of by the appropriate protocol & processed.
5. When the packet arrived at its destination, the process by which the headers are applied at the source is repeated in server.

Vertical communication:

1. In addition to communicating horizontally with the same protocol in the other system, the header information also enables each layer to communicate with the layer above & below it.
   Eg. The n/w layer will communicate with the data link layer & transport layer.
2. This interlayer communication is called communication vertical.
3. When a system receives a packet & passes it up through various layers the data link layer protocol header includes a field which specifies the name of n/w layer protocol to be used to process the packet.
4. The n/w layer protocol header will specify the name of transport layer protocol to be used to process the packet.
5. Due to vertical communication, it becomes protocol at each layer simultaneously.

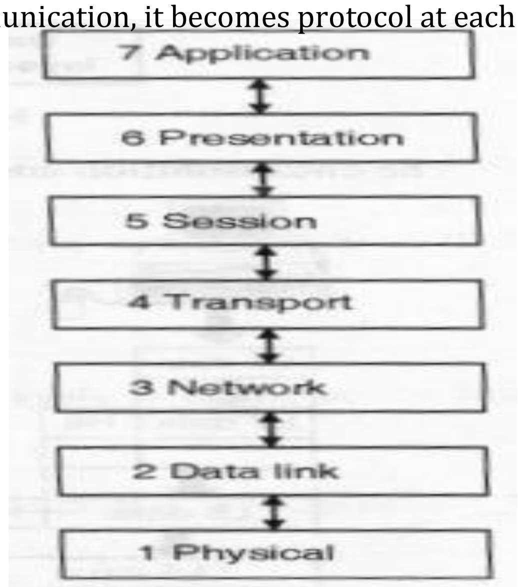


Fig: Vertical communication

Error Detection and Correction

★ Networks must be able to transfer data from one device to another with complete accuracy.

★ Data can be corrupted during transmission.

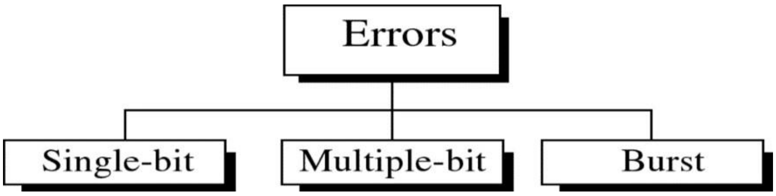
★ For reliable communication, errors must be detected and corrected.

★ Error detection and correction are implemented either at the data link layer or the transport layer of the OSI model.

Definition of Error

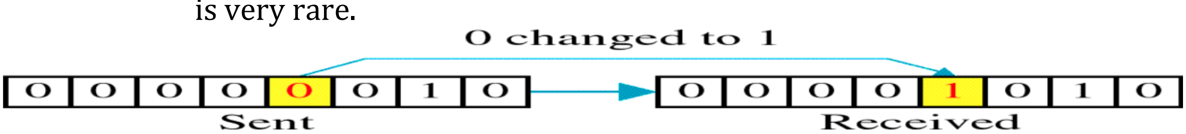
Networks must be able to transform data from once device to another with complete accuracy. While the transmission data can be corrupted, for reliable communication errors must be detected and corrected.

Types of Errors



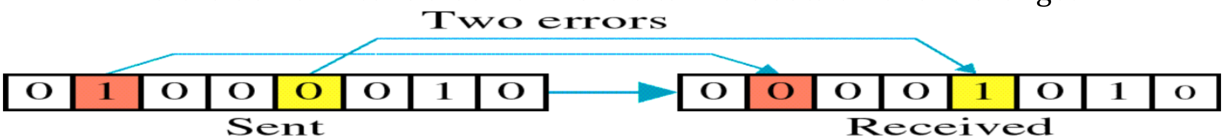
Single-bit errors

- In a single-bit error, only 1 bit in the data unit has changed from either 0 to 1 or 1 to 0.
- Single bit errors are the least likely type of errors in serial data transmission because the noise must have a very short duration which is very rare. However this kind of errors can happen in parallel transmission.
- *Example:*
    - If data is sent at 1Mbps then each bit lasts only 1/1,000,000 sec. or 1 μs.
    - For a single-bit error to occur, the noise must have a duration of only 1 μs, which is very rare.



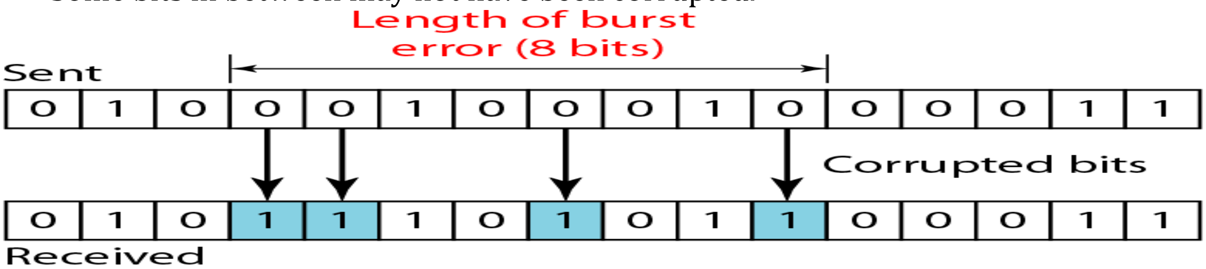
Multiple-bit errors

- A multi bit error means that 2 or more bits in the data unit have changed



Burst errors

- A burst error means that 2 or more **consecutive** bits in the data unit have changed.
- The term **burst error** means that two or more bits in the data unit have changed from 1 to 0 or from 0 to 1.
- **Burst errors 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.



★ Burst error is most likely to happen in serial transmission since the duration of noise is normally longer than the duration of a bit.

★ The number of bits affected depends on the data rate and duration of noise.

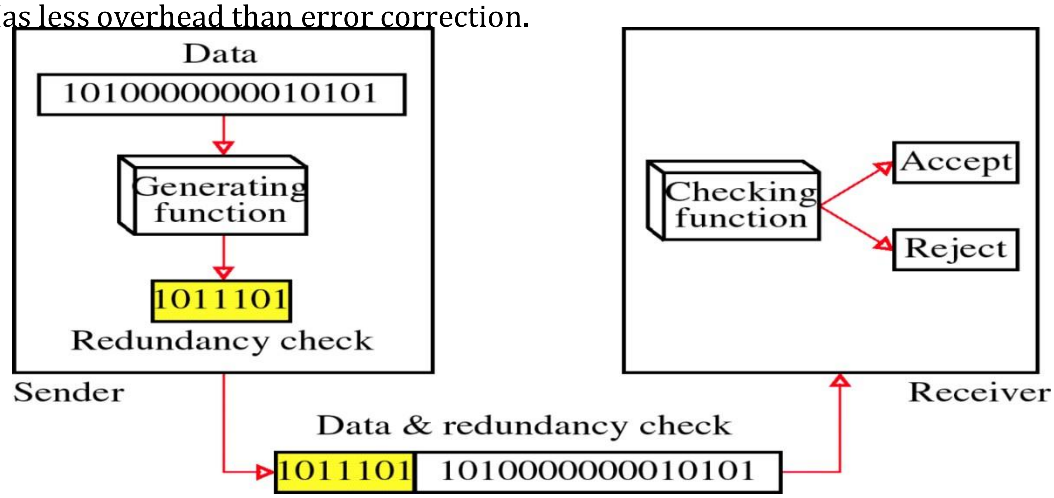
Example:

 If data is sent at rate = 1Kbps then a noise of 1/100 sec can affect 10 bits.(1/100*1000)

 If same data is sent at rate = 1Mbps then a noise of 1/100 sec can affect 10,000 bits.($1/100*10^6$)

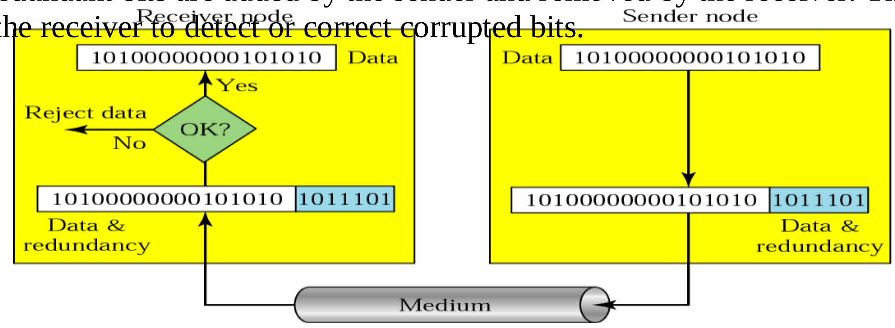
Error detection

- Error detection means to decide whether the received data is correct or not without having a copy of the original message.
- Error detection uses the concept of **redundancy**, which means adding extra bits for detecting errors at the destination.
- Enough redundancy is added to detect an error.
- The receiver knows an error occurred but does not know which bit(s) is(are) in error.
- Has less overhead than error correction.



Redundancy:

- 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:

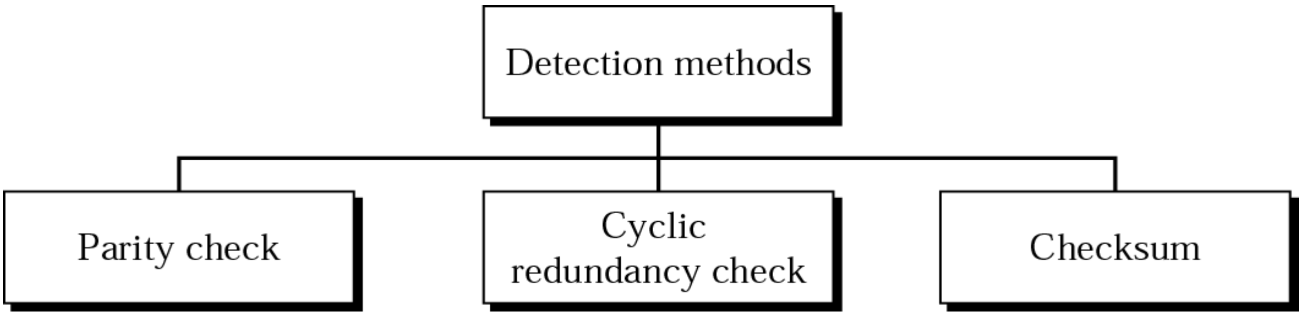
- The correction of errors is more difficult than detection.
- In error detection, we are looking only to see if any error has occurred. The answer is a simple yes or no.
- In error correction, we need to know the exact number of bits that are corrupted and more importantly, their location in the message. The number of errors and the size of message are important.

Forward Error Correction Versus Retransmission:
There are **two** main methods of **error correction**.

- **Forward error correction** is the process in which the receiver tries to guess the message by using redundant bits. This is possible if the number of errors is small.
- **Correction by retransmission** is a technique in which the receiver detects the occurrence of an error and asks the sender to resend the message. Resending is repeated until a message arrives that the receiver believes to be error-free.

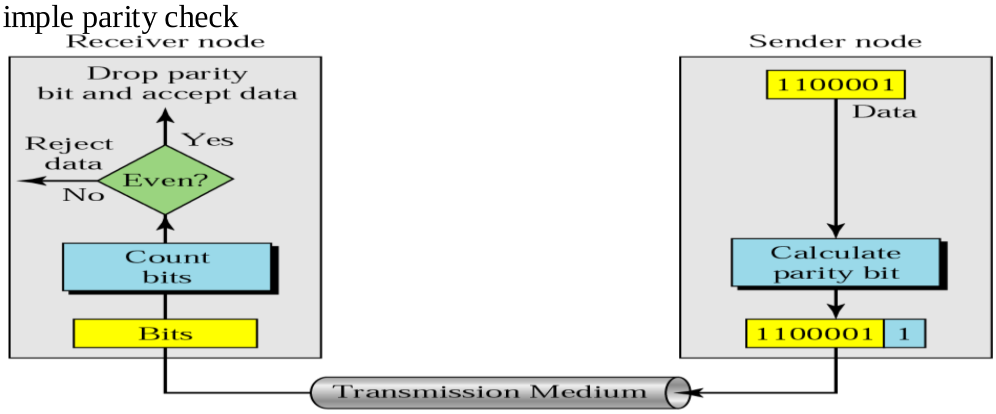
Detection Techniques

- Parity Checks
- Checksum methods
- Cyclic redundancy checks (CRC)



- # Parity Check

  - A parity bit is added to every data unit so that the total number of 1s(including the parity bit) becomes even for even-parity check or odd for odd-parity check
  - Fig: Simple parity check



Example-1: Suppose the sender wants to send the word *world*. In ASCII the five characters are coded as

1110111 1101111 1110010 1101100 1100100

The following shows the actual bits sent

11101110 11011110 11100100 11011000 11001001

Example-2: Now suppose the word world in Example 1 is received by the receiver without being corrupted in transmission.

11101110 11011110 11100100 11011000 11001001

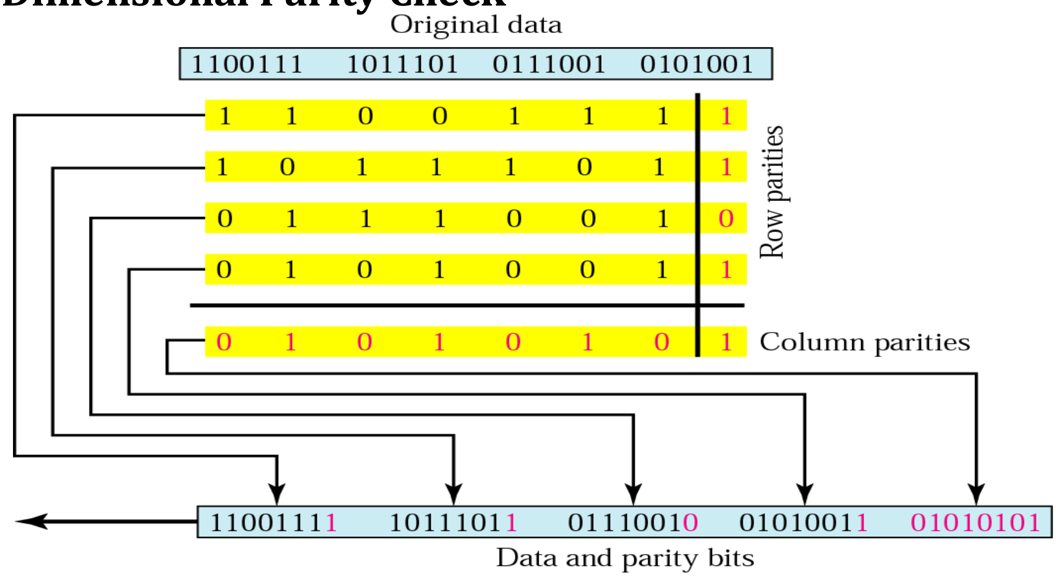
The receiver counts the 1s in each character and comes up with even numbers (6, 6, 4, 4, 4). The data are accepted.

Example-3: Now suppose the word world in Example 1 is corrupted during transmission.

11111110 11011110 11101100 11011000 11001001

The receiver counts the 1s in each character and comes up with even and odd numbers (7, 6, 5, 4, 4). **The receiver knows that the data are corrupted, discards them, and asks for retransmission.**

Two –Dimensional Parity Check



Example: Suppose the following block is sent:

10101001 00111001 11011101 11100111 10101010

However, it is hit by a burst noise of length 8, and some bits are corrupted.

10100011 10001001 11011101 11100111 10101010

When the receiver checks the parity bits, some of the bits do not follow the even-parity rule and the whole block is discarded.

10100011 10001001 11011101 11100111 10101010

Question: State any two drawbacks of parity checking for error detection.

Answer: Drawbacks of parity checking for error detection:

1. Can be used to detect single bit errors
2. Cannot detect location of errors.
3. Overheads are more.

Question: Assuming even parity technique find the parity bit for following frames:

i) 0000010 ii) 1111000 iii) 1010101 iv) 1011011

Answer:

Sr. No	Data	Parity bit
1	0000010	1
2	1111000	0
3	1010101	0
4	1011011	1

Question: Assuming odd parity, find the parity bit for each of the following data unit:

(i) 1011010 (ii) 0010110 (iii) 1001111 (iv) 1100000

Answer: Odd parity refers to number of „1‟ present in a byte to be transmitted should be odd.

(i) 1011010:

Step 1: Count the number of "1"s in the byte

Answer: 4

Step 2: compute the parity bit

Answer: 1011010 1

Since the total number of 1's is 4, the odd parity will have a value of "1".

(ii) 0010110:

Step 1: Count the number of "1"s in the byte

Answer: 3

Step 2: compute the parity bit

Answer: 0010110 0

Since the total number of 1's is 3, the odd parity will have a value of "0".

(iii) 1001111:

Step 1: Count the number of 1's in the byte

Answer: 5

Step 2: compute the parity bit

Answer: 1001111 0

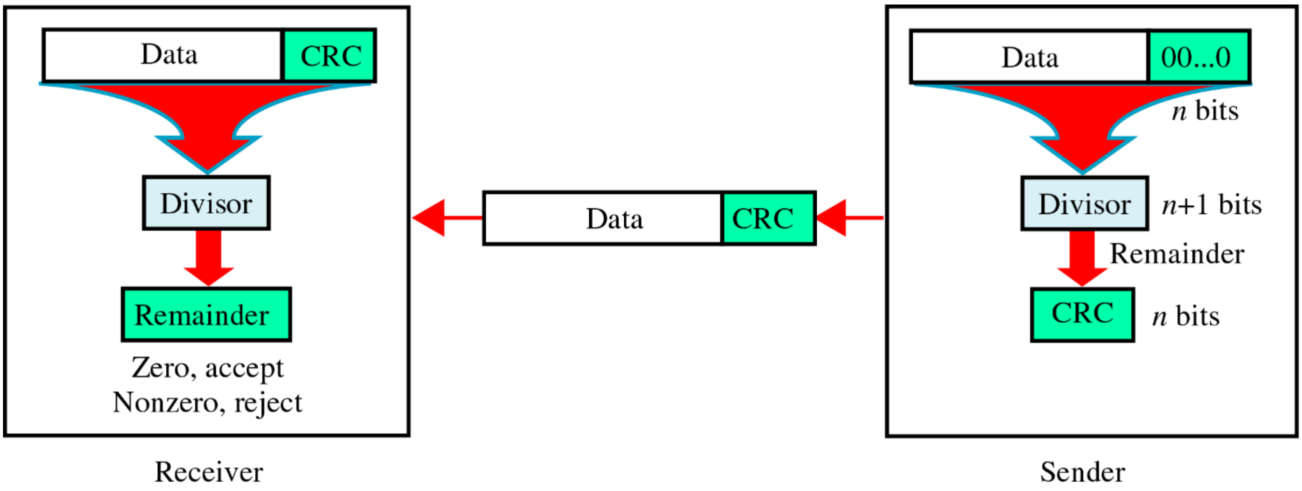
Since the total number of 1's is 5, the odd parity will have a value of "0".
(iv) 1100000:
Step 1: Count the number of 1's in the byte
Answer: 2
Step 2: compute the parity bit
Answer: 1100000 1
Since the total number of 1's is 2, the odd parity will have a value of "1".

- **CRC(Cyclic Redundancy Check)**

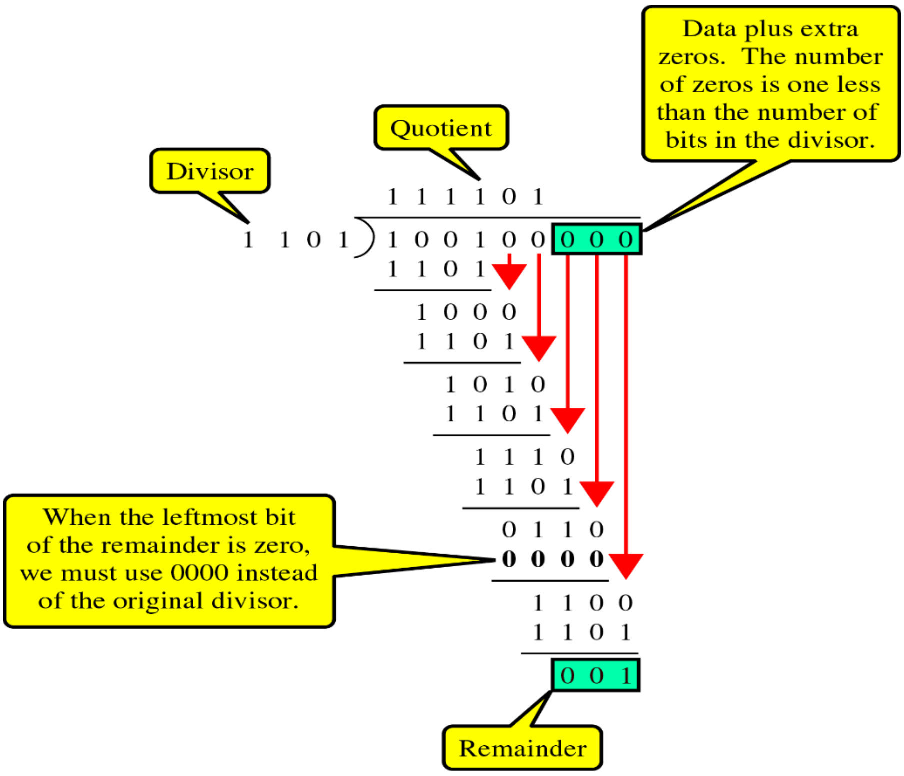
- **CRC Encoder:**
  - In the encoder, the dataword has k bits; the codeword has n bits.
  - The size of the dataword is augmented by adding $n-k$ 0s to the right-hand side of the word. The n-bit result is fed into the generator.
  - The generator uses a divisor of size $n - k + 1$, predefined and agreed upon. The generator divides the augmented dataword by the divisor (modulo-2 division).
  - The quotient of the division is discarded; the remainder $r2$ $r1$ $r0$ is appended to the dataword to create the codeword.

- **CRC Decoder:**
  - The codeword 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 dataword is separated from the received codeword and accepted.
  - Otherwise, everything is discarded.

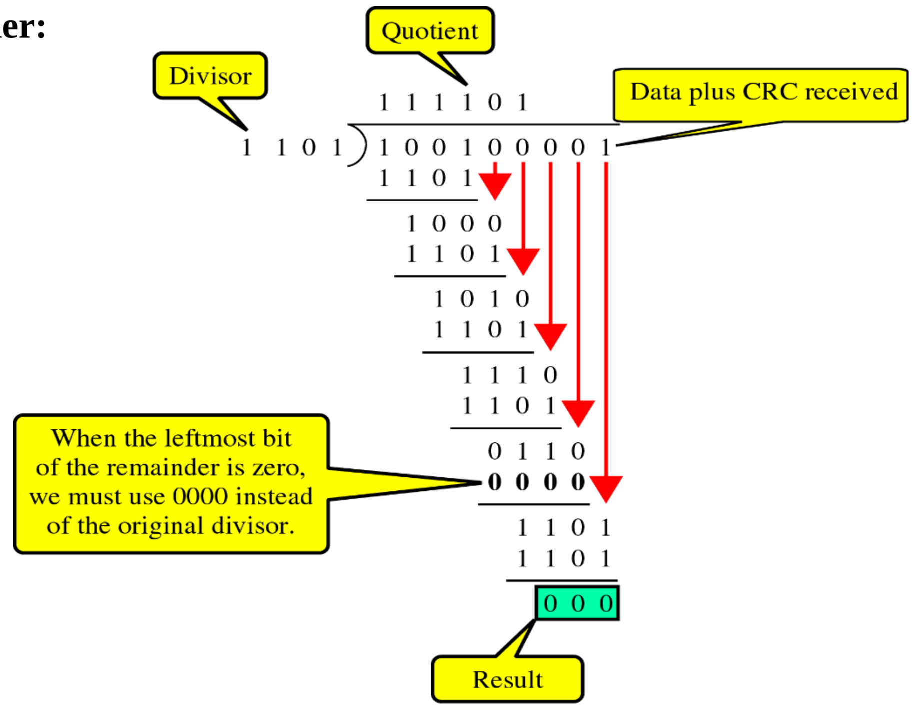
- **CRC(Cyclic Redundancy Check) is based on binary division.**



❑ **CRC generator uses modular-2 division.**

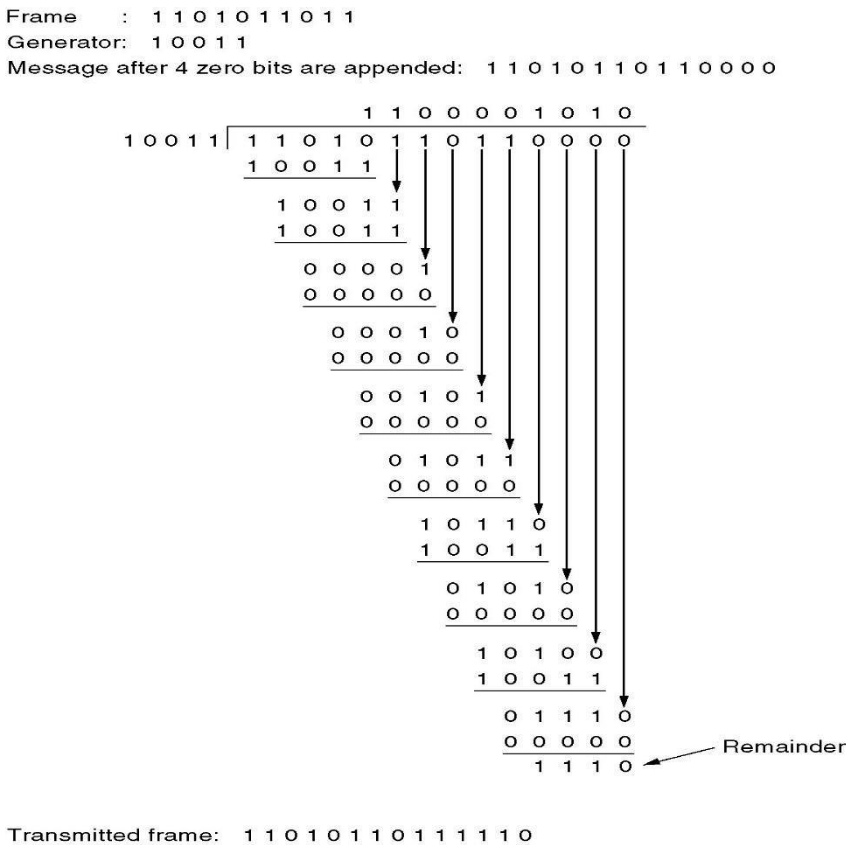


CRC Decoder:



Another Example: **Original Message:1 1 0 1 0 1 1 0 1 1**

- ❑ Devisor: 1 0 0 1 1
- ❑ Remainder:1 1 1 0
- ❑ Transmitted msg: 1 1 0 1 0 1 1 0 1 1 1 1 1 0



Question: **Explain the process of CRC with respect to following example. If G (X) = 110010 and M (X) = 101 then calculate CRC for above stream.**

Answer:

Procedure:- data bits= G(X)=110010 divisor=M(X)=101

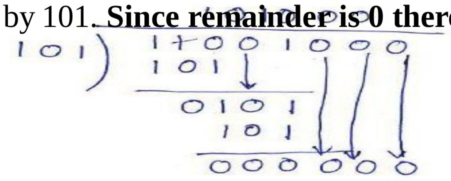
Here divisor is 3 bits so we need to append 2 zeroes (2 bit) to the data bits for division. Division carried is the normal binary division.

Result is calculated by the following condition:

1. If the remainder after division process is zero, it indicates that the data bits has no errors and the data bit is acceptable
2. If the remainder after division is non-zero , it indicates that the data bits has errors and we have to append the remainder bits to the original data bits and then send the data again. This remainder bits are called as the CRC. So the data bits transmitted will be DATA + CRC.

Consider the given example, lets perform division process for CRC.

Here the divisor is 3 bits hence we append 2 zeroes to the data bits, so the data bits will be 11001000 this will be divided by 101. **Since remainder is 0 there is no error in the data.**

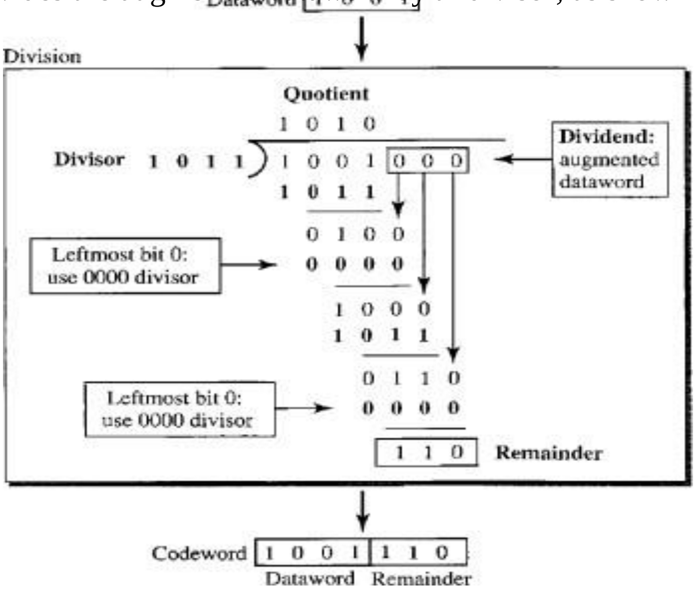


Reminder is 00 = CRC

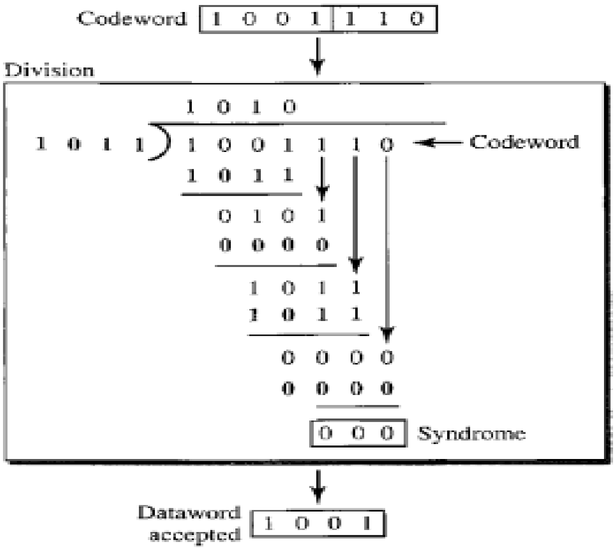
∴ Data + CRC = 110010 + 00

= 11001000

Question: Explain process of CRC (Cyclic Redundancy Check) with example.
Answer:
CRC Encoder:
In the encoder, the dataword has k bits (4 here); the codeword has n bits (7 here). The size of the dataword 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 $n - k + 1$ (4 here), predefined and agreed upon. The generator divides the augmented dataword by the divisor (modulo-2 division). The quotient of the division is discarded; the remainder $r_2 r_1 r_0$ is appended to the dataword to create the codeword.
Example:
Let us take a closer look at the encoder. The encoder takes the dataword and augments it with $n - k$ number of 0s. It then divides the augmented dataword by th divisor, as shown in Figure.



CRC Decoder:
The codeword 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 dataword is separated from the received codeword and accepted. Otherwise, everything is discarded.



Checksum:

Checksum is an error detection method. It can be done by following steps

Step-01:

- ❑ At sender side, If m bit checksum is used, the data unit to be transmitted is divided into segments of m bits.
- ❑ All the m bit segments are added.
- ❑ The result of the sum is then complemented using 1"s complement arithmetic.
- ❑ The value so obtained is called as checksum**.**

Step-02:

- ❑ The data along with the checksum value is transmitted to the receiver.

Step-03:

- ❑ All the m bit segments are added along with the checksum value. The value so obtained is complemented and the result is checked.
- ❑ Case-01: Result = 0 If the result is zero
  - ❖ Receiver assumes that no error occurred in the data during the transmission.
  - ❖ Receiver accepts the data.
- ❑ Case-02: Result ≠ 0 If the result is non-zero,
  - ❖ Receiver assumes that error occurred in the data during the transmission.
  - ❖ Receiver discards the data and asks the sender for retransmission.

Example:

- ❑ At the sender Original data : 10101001 00111001

```
10101001
00111001-

11100010     Sum
00011101     Checksum(1's complement)
```

Data to be transmitted:10101001 00111001 00011101

- ❑ At the receiver

Received data:10101001 00111001 00011101

```
10101001
00111001
00011101--

11111111 ← Sum
00000000 ← Complement is zero so the data is accepted.
```

Example-2: Consider the data unit to be transmitted is- 10011001111000100010010010000100

Consider 8 bit checksum is used.

- ❑ Step-01:
  - ❖ At sender side,
  - ❖ The given data unit is divided into segments of 8 bits as-
- ❑ Now, all the segments are added and the result is obtained as- 10011001 + 11100010 + 00100100 + 10000100 = 1000100011
  - ❖ Since the result consists of 10 bits, so extra 2 bits are wrapped around. 00100011 + 10 = 00100101 (8 bits)
  - ❖ Now, 1"s complement is taken which is 11011010.
  - ❖ Thus, checksum value = 11011010
- ❑ Step-02:
  - ❖ The data along with the checksum value is transmitted to the receiver.

- ❑ Step-03:
  - ❖ At receiver side, The received data unit is divided into segments of 8 bits.
  - ❖ All the segments along with the checksum value are added.
  - ❖ Sum of all segments + Checksum value =
  - ❖ 00100101 + 11011010= 11111111
  - ❖ Complemented value = 00000000
  - ❖ Since the result is 0, receiver assumes no error occurred in the data and therefore accepts it.

Error Correction: It can be handled in two ways

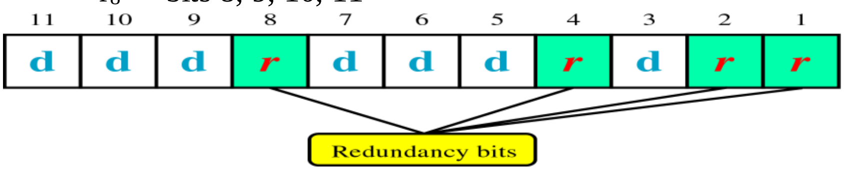
- When an error is discovered, the receiver can have the sender retransmit the entire data unit.
  - Automatic Repeat Request (ARQ)
- A receiver can use an error-correcting code, which automatically corrects certain errors.
  - Hamming Code

Hamming Code: **developed by R.W.Hamming**

- ❑ Positions of redundancy bits in 7 bit Hamming code is shown in fig.

Each r bit is the VRC bit for one combination of data bits, ie even parity bit is generated using the following bits and code is generated.

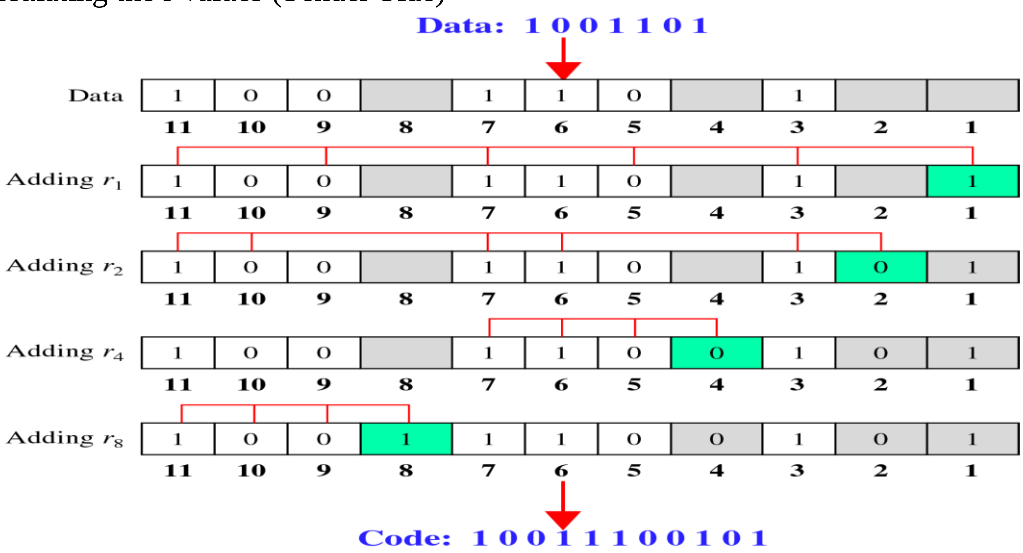
- r_1 = bits 1, 3, 5, 7, 9, 11
- r_2 = bits 2, 3, 6, 7, 10, 11
- r_4 = bits 4, 5, 6, 7
- r_8 = bits 8, 9, 10, 11



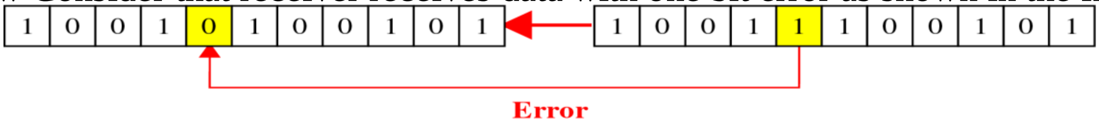
- ❑ The redundant bits are inserted at each 2^n bit where n=0,1,2,3,……

Example:

- ❑ Calculating the r values (Sender Side)

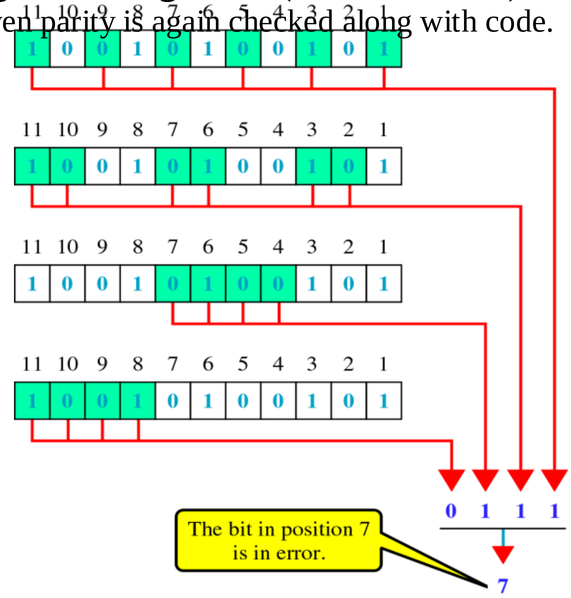


1. Now Consider that receiver receives data with one bit error as shown in the fig.



2. Error detection using Hamming Code (Receiver Side)

3. At the receiver side the even parity is again checked along with code.



4. The parity generated shows the bit position in error.

Framing:

- ❖ The data link layer needs to pack bits into frames, so that each frame is distinguishable from another.
- ❖ *Our postal system practices a type of framing. The simple act of inserting a letter into an envelope separates one piece of information from another; the envelope serves as the delimiter.*
- ❖ Framing in the data link layer separates a message from one source to a destination, or from other messages to other destinations, by adding addresses.
- ❖ The destination address defines where the packet is to go; the sender address helps the recipient acknowledge the receipt.
- ❖ Frames can be of fixed or variable size.

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

- ❖ Variable-size framing is used in local- area networks.
- ❖ In variable-size framing, we need a way to define the end of the frame and the beginning of the next.
- ❖ Historically, two approaches were used for this purpose:
  - ❖ a character-oriented approach and
  - ❖ a bit-oriented approach.

Question: Two channels one with a bit rate of 100 Kbps and another with bit rate of 200 Kbps are to be multiplexed.

Answer the following questions:

i) Calculate size of frames in bits

ii) Calculate the frame rate

iii) Calculate the duration of frame

Answer:

Channel 1 has a bit rate of 100Kbps. Channel 2 has a bit rate of 200Kbps Hence channel 2 is demultiplexed into 2 channels of 100Kbps each. Hence 3 channels of 100 Kbps are multiplexed effectively.

Let us consider that one slot of the channel 1 is allocated and two slots of the channel 2 is allocated in the frame .

i) Calculate size of frames in bits: Thus **e**ach frame carries 3 bits.

ii) Calculate the frame rate: The total bit rate of the multiplexed link is 300kbps. Each frame has 3 bits. The frame rate is100,000 frames per second (Any other assumption may also be considered).

iii) Calculate the duration of frame: Thus the frame duration is

- 1/100,000s or 1μs.

Flow and Error Control

- Data communication requires at least two devices working together, one to send and the other to receive.
- Even such a basic arrangement requires a great deal of coordination for an intelligible exchange to occur.
- The most important responsibilities of the data link layer are flow control and error control.
- Collectively, these functions are known as **data link control**.

Flow Control:

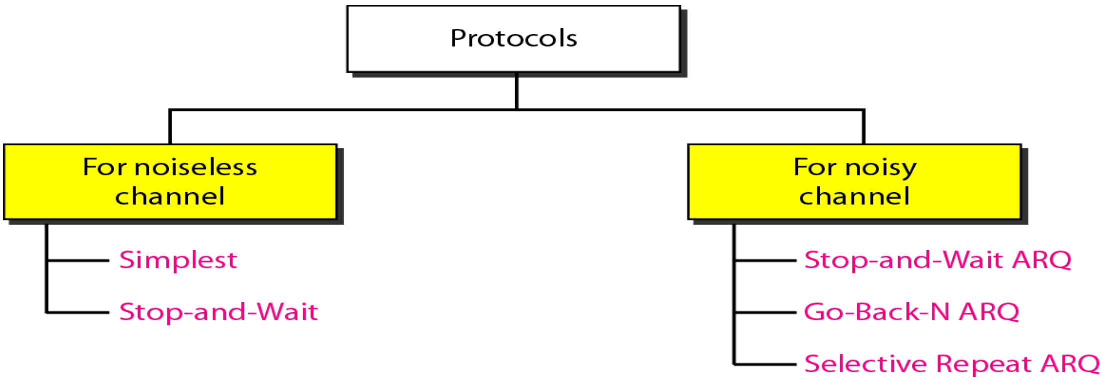
- Flow control coordinates the amount of data that can be sent before receiving an acknowledgment and is one of the most important duties of the data link layer.
- The flow of data must not be allowed to overwhelm the receiver.
- Any receiving device has a limited speed at which it can process incoming 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).

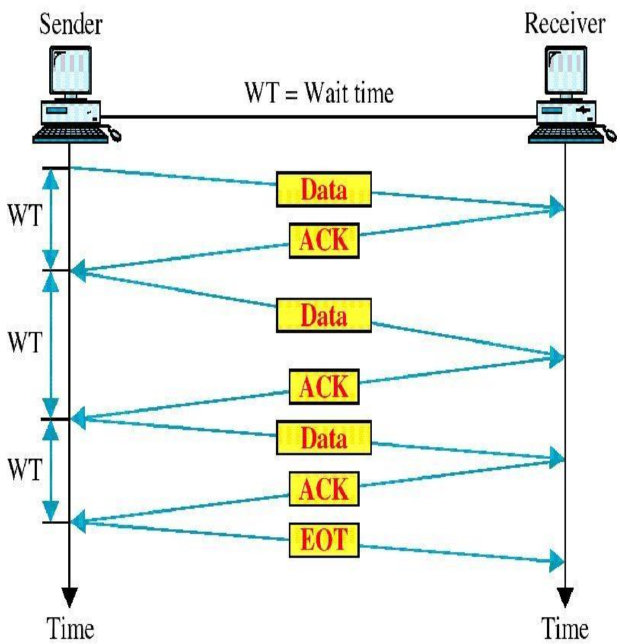
Flow Control and Error Control Technics



Question: Explain stop and wait ARQ with example.

Stop and Wait:

This is a very simple method where in the sender sends one frame of data and necessarily waits for an acknowledgement (ACK) from the receiver before sending the next frame. Only after the sender receives and acknowledgement for a frame does it send the next frame. Thus, the transmission always takes the form Data-ACK-Data-ACK….etc, where the Data frames are sent by the sender, and the ACK frames are sent by the receiver back to the sender. This is shown in figure. The stop-and wait- approach is pretty simple to implement. Every frame must be individually acknowledged before the next frame can be transmitted. However, therein also lies its drawback. Since the sender must receive each acknowledgement before it can transmit the next frame, it makes the transmission very slow.

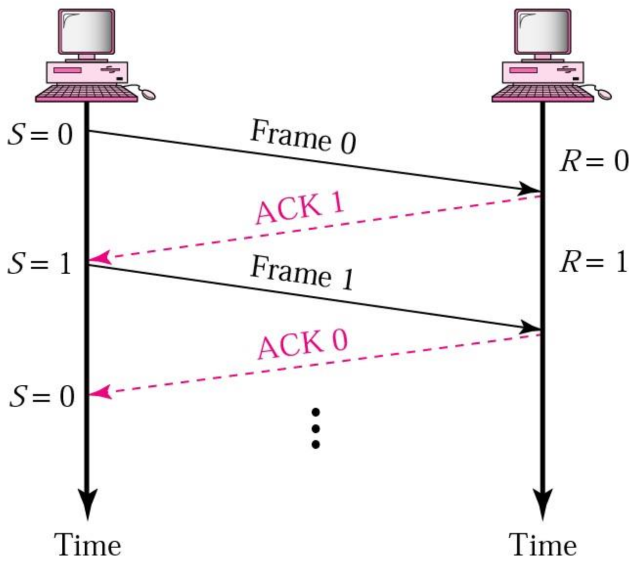


5. ***Stop-and-Wait ARQ*** *has the following features:*
   - ✓ *The sending device keeps a copy of the sent frame transmitted until it receives an acknowledgment(ACK)*
   - ✓ *The sender starts a timer when it sends a frame. If an ACK is not received within an allocated time period, the sender resends it*
   - ✓ *Both frames and acknowledgment (ACK) are numbered alternately 0 and 1(two sequence number only)*
   - ✓ *This numbering allows for identification of frames in case of duplicate transmission.*
   - ✓ *The acknowledgment number defines the number of next expected frame. (frame 0 received ACK 1 is sent)*
   - ✓ *A damage or lost frame treated by the same manner by the receiver*
   - ✓ *If the receiver detects an error in the received frame, or receives a frame out of order it simply discards the frame*
   - ✓ *The receiver send only positive ACK for frames received safe; it is silent about the frames damage or lost.*
   - ✓ *The sender has a control variable Sthat holds the number of most recently sent frame (0 or 1). The receiver has control variable R, that holds the number of the next frame expected (0,or 1)*

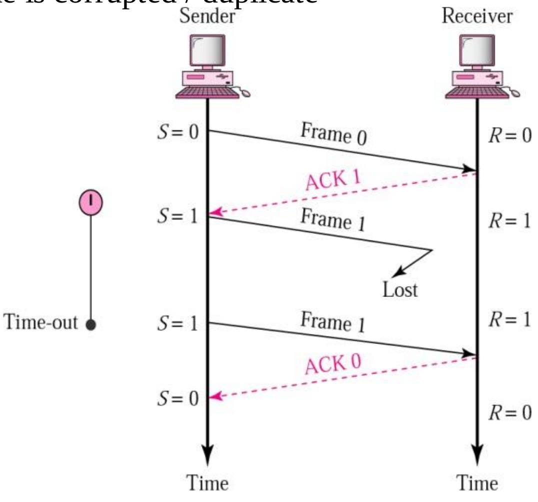
   Cases of Operations:
   1. *Normal operation*
   2. *The frame is lost*
   3. *The Acknowledgment (ACK) is lost*
   4. *The Ack is delayed*

1. Normal Operation:

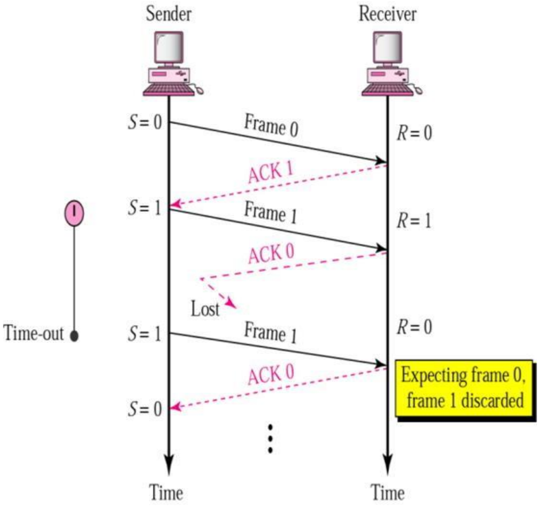
- ➢ The sender will not send the next frame until it is sure that the current one is correctly receive
- ➢ sequence number is necessary to check for duplicated frames



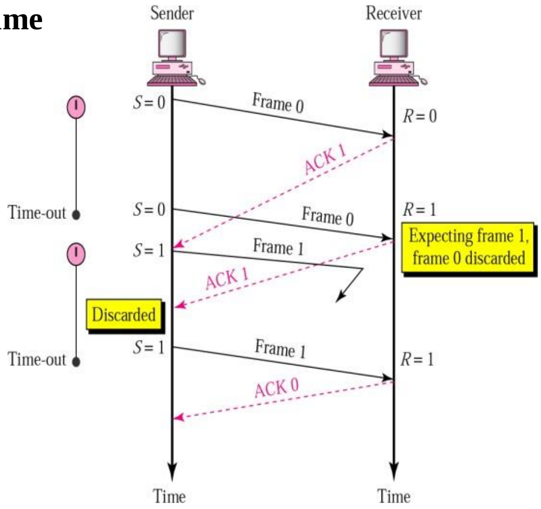
2. **Lost or damaged frame**
   - A damage or lost frame treated by the same manner by the receiver.
   - No NACK when frame is corrupted / duplicate



3. **Lost ACK frame**



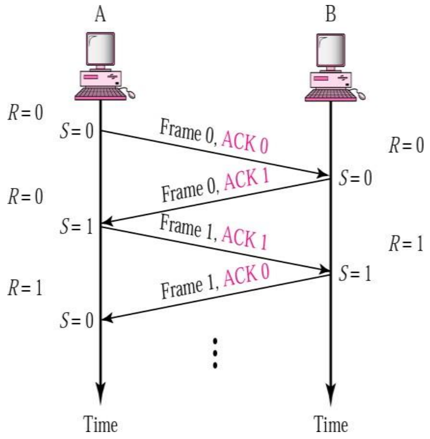
4. **Delayed ACK and lost frame**



5.

Piggybacking (Bidirectional transmission)

- Is a method to combine a data frame with an acknowledgment.
- It can save bandwidth because data frame and an ACK frame can combined into just one frame.



Sliding window protocol

Sliding window protocols apply Pipelining:A task is begun before the previous task has ended.

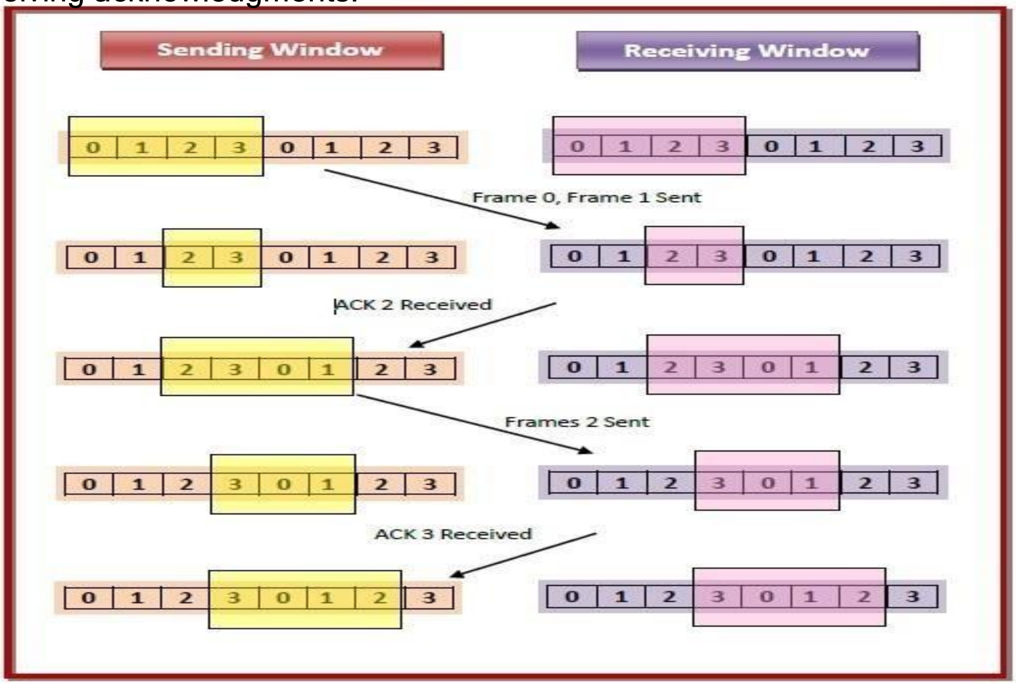
- ➢ The Stop and Wait ARQ offers error and flow control, but may cause big performance issues as sender always waits for acknowledgement even if it has next packet ready to send. Consider a situation where you have a high bandwidth connection and propagation delay is also high (you are connected to some server in some other country though a high speed connection), you can't use this full speed due to limitations of stop and wait.
- ➢ Sliding Window protocol handles this efficiency issue by sending more than one packet at a time with a larger sequence numbers. The idea is same as pipelining in architectures.

Working Principle

- ➢ In these protocols, the sender has a buffer called the sending window and the receiver has buffer called the receiving window.
- ➢ The size of the sending window determines the sequence number of the outbound frames. If the sequence number of the frames is an n-bit field, then the range of sequence numbers that can be assigned is 0 to 2^n-1. Consequently, the size of the sending window is 2^n-1. Thus in order to accommodate a sending window size of 2^n-1, a n-bit sequence number is chosen.
- ➢ The sequence numbers are numbered as modulo-n. For example, if the sending window size is 4, then the sequence numbers will be 0, 1, 2, 3, 0, 1, 2, 3, 0, 1, and so on. The number of bits in the sequence number is 2 to generate the binary sequence 00, 01, 10, 11.
- ➢ The size of the receiving window is the maximum number of frames that the receiver can accept at a time. It determines the maximum number of frames that the sender can send before receiving acknowledgment.

Example

- Suppose that we have sender window and receiver window each of size 4. So the sequence numbering of both the windows will be 0,1,2,3,0,1,2 and so on. The following diagram shows the positions of the windows after sending the frames and receiving acknowledgments.

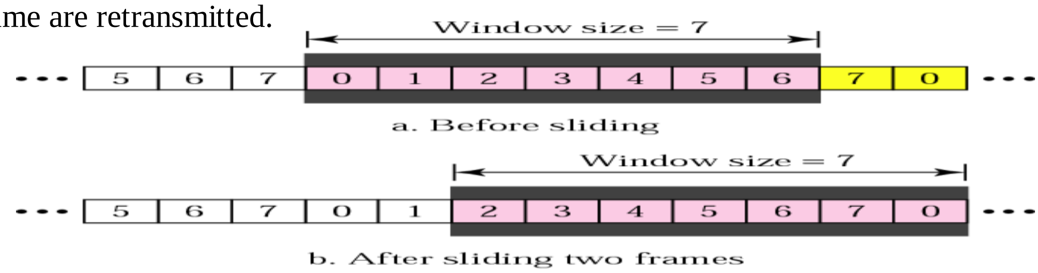


Types of Sliding Window Protocols

The Sliding Window ARQ (Automatic Repeat reQuest) protocols are of two categories -

- **Go – Back – *N* ARQ**

Go – Back – *N* ARQ provides for sending multiple frames before receiving the acknowledgment for the first frame. It uses the concept of sliding window, and so is also called sliding window protocol. The frames are sequentially numbered and a finite number of frames are sent. If the acknowledgment of a frame is not received within the time period, all frames starting from that frame are retransmitted.



Working Principle

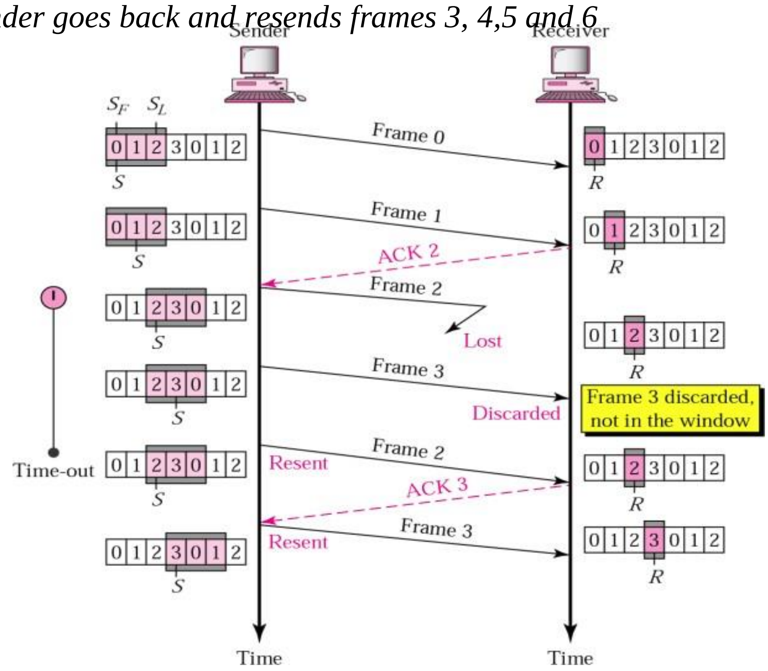
Go – Back – N ARQ provides for sending multiple frames before receiving the acknowledgment for the first frame. The frames are sequentially numbered and a finite number of frames. The maximum number of frames that can be sent depends upon the size of the sending window. If the acknowledgment of a frame is not received within an agreed upon time period, all frames starting from that frame are retransmitted.

The size of the sending window determines the sequence number of the outbound frames. If the sequence number of the frames is an n-bit field, then the range of sequence numbers that can be assigned is 0 to 2^n-1. Consequently, the size of the sending window is 2^n-1. Thus in order to accommodate a sending window size of 2^n-1, a n-bit sequence number is chosen.

The sequence numbers are numbered as modulo-n. For example, if the sending window size is 4, then the sequence numbers will be 0, 1, 2, 3, 0, 1, 2, 3, 0, 1, and so on. The number of bits in the sequence number is 2 to generate the binary sequence 00, 01, 10, 11.

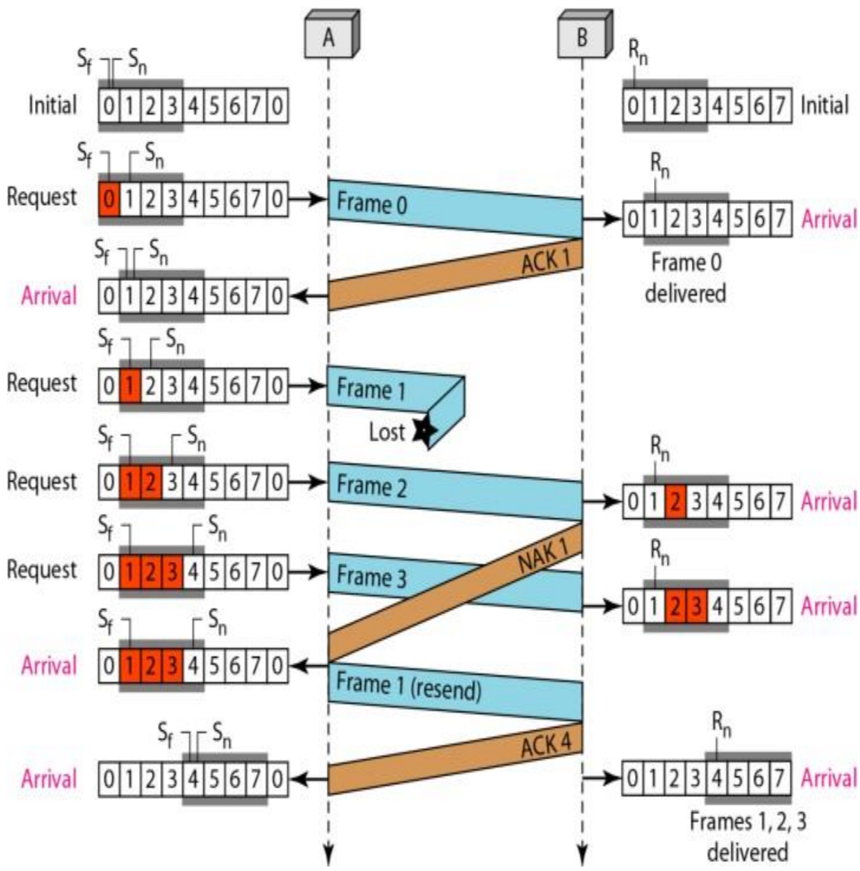
The size of the receiving window is 1.

Example: The sender has sent frame 6, and timer expires for frame 3(frame 3 has not been acknowledge); the sender goes back and resends frames 3, 4,5 and 6.



- **Selective Repeat ARQ**

This protocol also provides for sending multiple frames before receiving the acknowledgment for the first frame. However, here only the erroneous or lost frames are retransmitted, while the good frames are received and buffered.



Question: Explain the following flow and error control techniques: Go back N ARQ.
Answer: Go-Back-N ARQ:
In Go-Back-N ARQ method, both sender and receiver maintain a window.

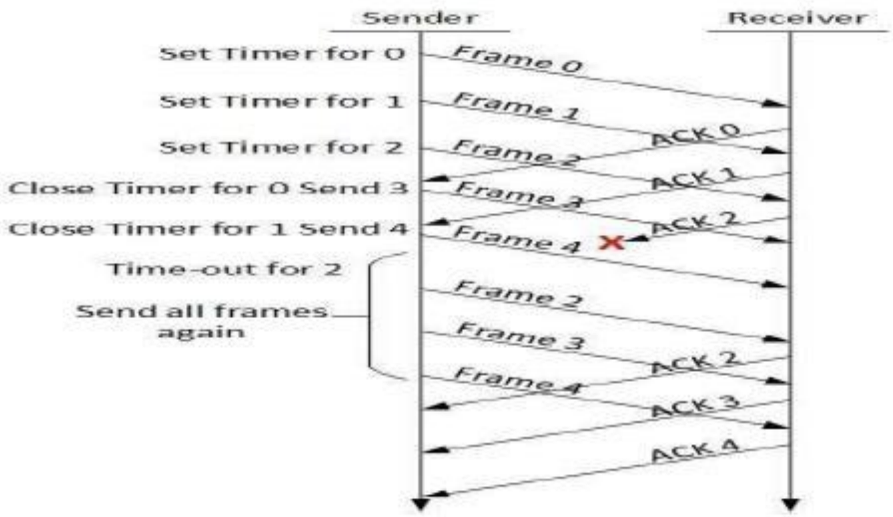


Fig: Go-Back-N ARQ

- The sending-window size enables the sender to send multiple frames without receiving the acknowledgement of the previous ones.
- The receiving-window enables the receiver to receive multiple frames and acknowledge them. The receiver keeps track of incoming frame's sequence number.
- When the sender sends all the frames in window, it checks up to what sequence number it has received positive acknowledgement.

- If all frames are positively acknowledged, the sender sends next set of frames.
- If sender finds that it has received NACK (negative acknowledgement)or has not receive any ACK for a particular frame, it retransmits all the frames after which it does not receive any positive ACK.

Chapter No: 5

WIRELESS COMMUNICATION

IEEE Wireless Standards

- Wireless networks are standardized by IEEE.
- Under 802 LAN MAN standards committee.

The 802.11 standard is defined through several specifications of WLANs. It defines an over-the-air interface between a wireless client and a base station or between two wireless clients.

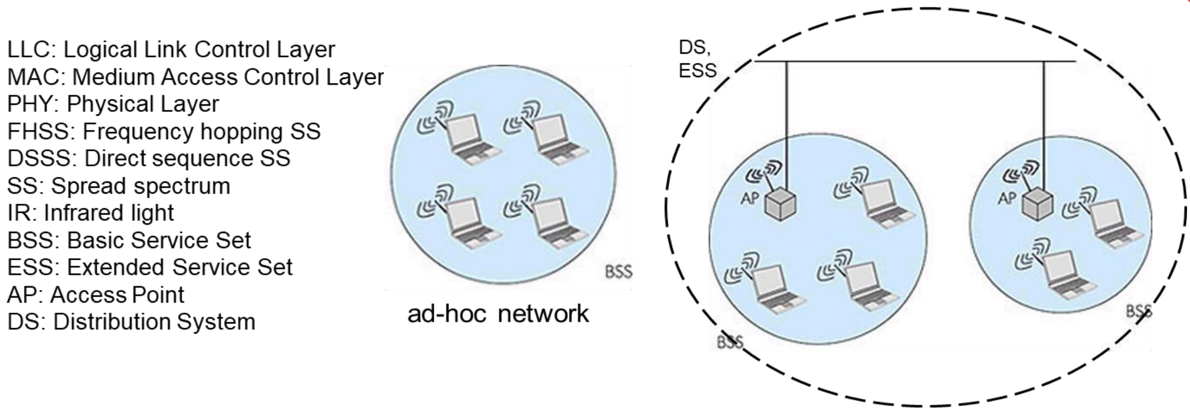
There are several specifications in the 802.11 family −

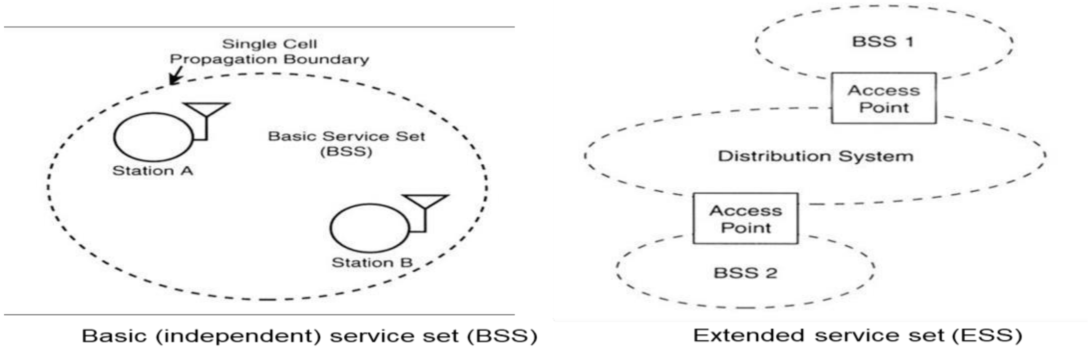
- **802.11** − This pertains to wireless LANs and provides 1 - or 2-Mbps transmission in the 2.4-GHz band using either frequency-hopping spread spectrum (FHSS) or direct-sequence spread spectrum (DSSS).
- **802.11a** − This is an extension to 802.11 that pertains to wireless LANs and goes as fast as 54 Mbps in the 5-GHz band. 802.11a employs the orthogonal frequency division multiplexing (OFDM) encoding scheme as opposed to either FHSS or DSSS.
- **802.11b** − The 802.11 high rate WiFi is an extension to 802.11 that pertains to wireless LANs and yields a connection as fast as 11 Mbps transmission (with a fallback to 5.5, 2, and 1 Mbps depending on strength of signal) in the 2.4-GHz band. The 802.11b specification uses only DSSS. Note that 802.11b was actually an amendment to the original 802.11 standard added in 1999 to permit wireless functionality to be analogous to hard-wired Ethernet connections.
- **802.11g** − This pertains to wireless LANs and provides 20+ Mbps in the 2.4-GHz band.
- **802.11n** — 802.11n builds upon previous 802.11 standards by adding ***m**ultiple* *input **m**ultiple-**o**utput* (MIMO). The additional transmitter and receiver antennas allow for increased data throughput through spatial multiplexing and increased range by exploiting the spatial diversity through coding schemes like Alamouti coding. The real speed would be 100 Mbit/s (even 250 Mbit/s in PHY level), and so up to 4-5 times faster than 802.11g.
- **802.11ac** — 802.11ac builds upon previous 802.11 standards, particularly the 802.11n standard, to deliver data rates of 433Mbps per spatial stream, or 1.3Gbps in a three-antenna (three stream) design. The 802.11ac specification operates only in the 5 GHz frequency range and features support for wider channels (80MHz and 160MHz) and beam forming capabilities by default to help achieve its higher wireless speeds.

Wireless LAN and IEEE 802.11

Wireless LANs are those Local Area Networks that use high frequency radio waves instead of cables for connecting the devices in LAN. Users connected by WLANs can move around within the area of network coverage. Most WLANs are based upon the standard IEEE 802.11 or WiFi.

IEEE 802.11 Architecture





The components of an IEEE 802.11 architecture are as follows

1) Stations (STA): Stations comprise all devices and equipment's that are connected to the wireless LAN. A station can be of two types:

1. **Wireless Access Points (WAP):** WAPs or simply access points (AP) are generally wireless routers that form the base stations or access.
2. **Client. :** Clients are workstations, computers, laptops, printers, smartphones, etc.

Each station has a wireless network interface controller.

2) Basic Service Set (BSS): A basic service set is a group of stations communicating at physical layer level. BSS can be of two categories depending upon mode of operation:

1. **Infrastructure BSS:** Here, the devices communicate with other devices through access points.
2. **Independent BSS:** Here, the devices communicate in peer-to-peer basis in an ad hoc manner.

3) Extended Service Set (ESS): It is a set of all connected BSS.

4) Distribution System (DS): It connects access points in ESS.

Advantages of WLANs

1. They provide clutter free homes, offices and other networked places.
2. The LANs are scalable in nature, i.e. devices may be added or removed from the network at a greater ease than wired LANs.
3. The system is portable within the network coverage and access to the network is not bounded by the length of the cables.
4. Installation and setup is much easier than wired counterparts.
5. The equipment and setup costs are reduced.

Disadvantages of WLANs

1. Since radio waves are used for communications, the signals are noisier with more interference from nearby systems.
2. Greater care is needed for encrypting information. Also, they are more prone to errors. So, they require greater bandwidth than the wired LANs.
3. WLANs are slower than wired LANs.

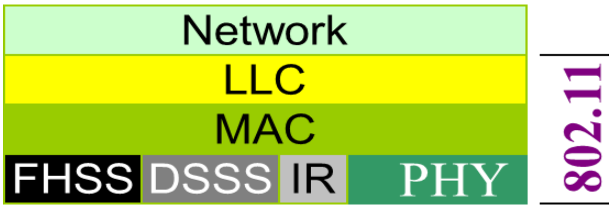


Fig: LAN Architecture

MAC sublayer Functions

802.11 support two different modes of operations. These are:

1. Distributed Coordination Function (DCF)

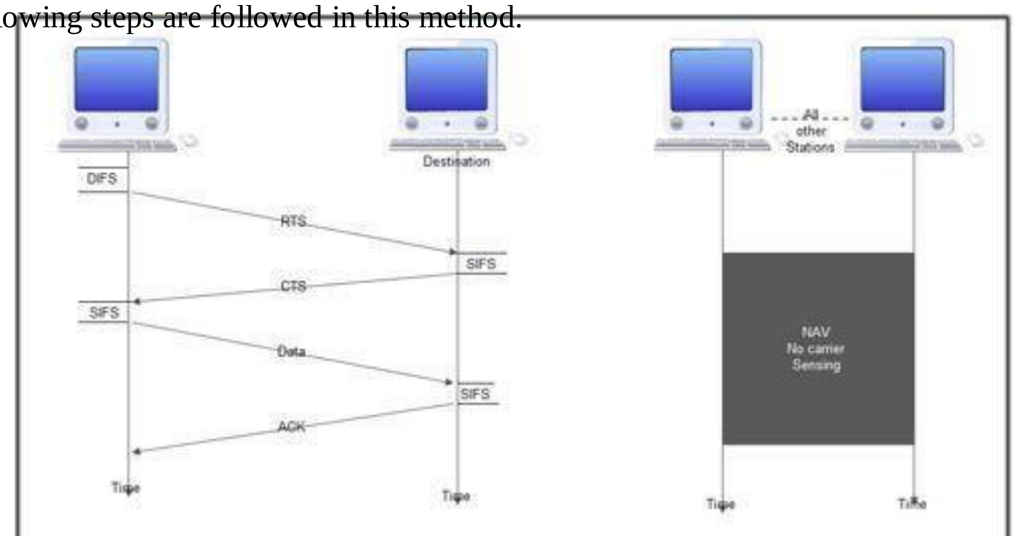
2. Point Coordination Function (PCF)

1. Distributed Coordination Function

• The DCF is used in BSS having no access point.

• DCF uses CSMA/CA protocol for transmission.

• The following steps are followed in this method.



CSMA/CA and NAV forIEEE 802.11

1. When a station wants to transmit, it senses the channel to see whether it is free or not.

2. If the channel is not free the station waits for back off time.

3. If the station finds a channel to be idle, the station waits for a period of time called distributed interframe space (DIFS).

4. The station then sends control frame called request to send (RTS) as shown in figure.

5. The destination station receives the frame and waits for a short period of time called short interframe space (SIFS).

6. The destination station then sends a control frame called clear to send (CTS) to the source station. This frame indicates that the destination station is ready to receive data.

7. The sender then waits for SIFS time and sends data.

8. The destination waits for SIFS time and sends acknowledgement for the received frame.

2. Point Coordination Function

• PCF method is used in infrastructure network. In this Access point is used to control the network activity.

• It is implemented on top of the DCF and IS used for time sensitive transmissions.

• PCF uses centralized, contention free polling access method.

• The AP performs polling for stations that wants to transmit data. The various stations are polled one after the other.

• To give priority to PCF over DCF, another interframe space called PIFS is defined. PIFS (PCF IFS) is shorter than DIFS.

• If at the same time, a station is using DCF and AP is using PCF, then AP is given priority over the station.

• Due to this priority of PCF over DCF, stations that only use DCF may not gain access to the channel.

• To overcome this problem, a repetition interval is defined that is repeated continuously. This repetition interval starts with a special control frame called beacon frame.

• When a station hears beacon frame, it start their NAV for the duration of the period of the repetition interval.

802.11 Addressing

• There are four different addressing cases depending upon the value of *To DS And from* DS subfields of FC field.

• Each flag can be 0 or 1, resulting in 4 different situations.

1. If *To* DS = 0 and *From* DS = 0, it indicates that frame is not going to distribution system and is not coming from a distribution system. The frame is going from one station in a BSS to another.

2. If *To* DS = 0 and *From* DS = 1, it indicates that the frame is coming from a distribution system. The frame is coming from an AP and is going to a station. The address 3 contains original sender of the frame (in another BSS).

3. If *To* DS = 1 and *From* DS = 0, it indicates that the frame is going to a distribution system. The frame is going from a station to an AP. The address 3 field contains the final destination of the frame.

4. If *To* DS = 1 and *From* DS = 1,it indicates that frame is going from one AP to another AP in a wireless distributed system.

The table below specifies the addresses of all four cases.

TO DS	From DS	Address 1	Address 2	Address3	Addres 4
0	0	Destination	Source	BSS ID	N/A
0	1	Destination	Sending AP	Source	N/A
1	0	Receiving AP	Source	Destination	N/A
1	1	Receiving AP	Sending AP	Destination	Source

Question: Explain the concept of piconet and scatter net of Bluetooth.

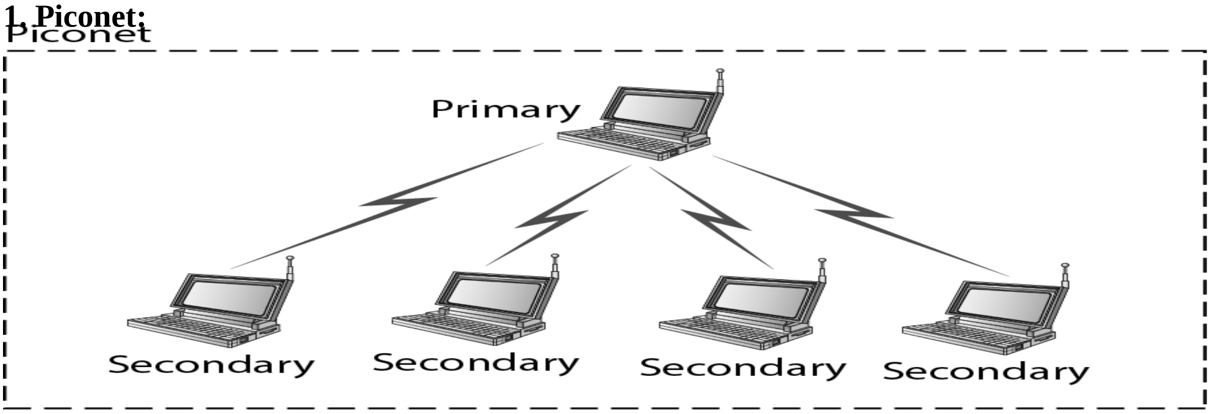
Bluetooth:

Bluetooth is an open specification for short range wireless transmission of voice and data. It provides a simple, Low cost seamless wireless connectivity between personal digital assistants, cellular phone laptops, and other portable handheld devices.

Bluetooth is low power consuming technology with transmission distances of up to 30 feet and a throughput of about 1 Mbps.

- Bluetooth defines two types of networks: **1. Piconet** **2. Scatternet**

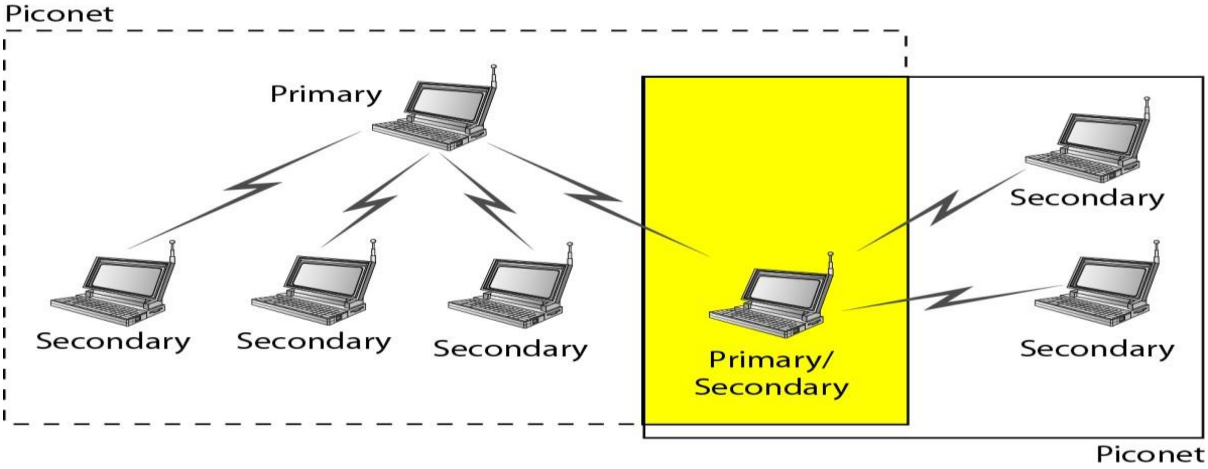
1. Piconet:



- **Piconet** is a Bluetooth network that consists of one **primary** (**master**) node and seven active **secondary** (**slave**) nodes.
- All Slave stations are **synchronized** with Master.
- Thus, piconet can have up to **eight** active nodes (1 master and 7 slaves) or stations within the distance of 10 meters.
- There can be only one primary or master station in each piconet.
- The communication between the primary and the secondary can be one-to-one or one-to-many.
- All communication is between master and a slave. **Salve-slave communication is not possible**.

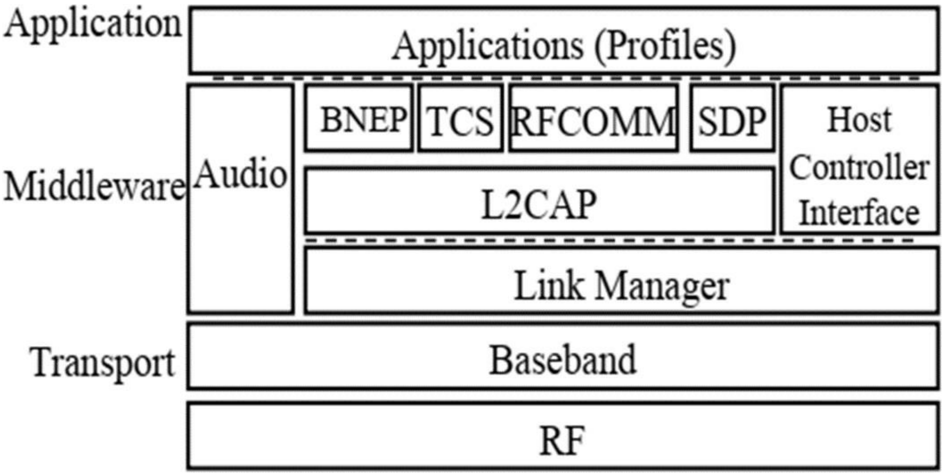
- In addition to seven active slave stations, a piconet can have up to **255 parked nodes**. These parked nodes are secondary or slave stations and cannot take part in communication until it is moved from parked state to active state.

2. Scatternet :



- **Scatternet** is formed by combining various piconets.
- A **slave** in one piconet can act as a **master** or **primary** in other piconet.
- Such a station or node can receive messages from the master in the first piconet and deliver the message to its slaves in other piconet where it is acting as master. This node is also called bridge slave.
- Thus a station can be a member of two piconets.
- A station cannot be a master in two piconets.

Bluetooth Protocol Stack



The protocols in this group are designed to

- Allow devices to locate and connect
- Carry audio and data traffic where audio traffic has higher priority.

- Support synchronous and asynchronous transmission for telephony grade voice communication
- Manage physical and logical links between devices so that layers above and applications can pass data through connections.
- The following protocols are in this group:

i. Logical link control and adaptation protocol layer (L2CAP)

- All data traffic is routed through this layer.
- This layer shields higher layers from details of lower layers.
- It segments larger packets from higher layers into smaller packets that can be easily handled by lower layers.
- It facilitates maintenance of desired grade of service in two peer devices.

ii. Link manager layer (LML)

- It negotiates properties of Bluetooth air interface between communicating devices.
- These properties may be bandwidth allocation, support services of particular type, etc.
- This layer also supervises devices pairing.
- Device pairing generates and stores authentication key specific to a device
- It is also responsible for power control and may request adjustments in power levels.

iii. Baseband and radio layers

- The baseband layer is responsible for searching other devices, assigning master and slave roles.
- This layer also controls Bluetooth unit's synchronization and transmission frequency hopping sequence. It manages link between devices and determines packet types supported for synchronous and asynchronous traffic.

iv. Host Controller Interface (HCI)

- The HCI allows higher layers of stack, including applications, to access the baseband, link manager, etc., through a single standard interface.
- It serves the purpose of interoperability between host devices and Bluetooth modules.
- HCI commands, module may enter certain modes of operation. Higher layers are informed about certain events through HCI.

2. Middleware protocol group

- The protocols in this group are needed for existing applications to operate over Bluetooth links.
- These protocols may be third party protocols (Industry standard) or developed by 'simple interest group (SIG)' specifically for Bluetooth.
- Some of the protocols in this group:

i. RFCOMM layer

- It provides a virtual serial port for applications needed for scenarios like dial-up networking, etc.
- This eliminates the use of cables.

ii. Service Discovery protocol layer (SDP)

- The SDP is a standard method for Bluetooth devices to discover and learn about the services offered by other device once a connection is established with it.

iii. Infrared data association(IrDA) interoperability protocols

- The SIG has adopted some IrDA protocols to ensure interoperability between applications to exchange a wide variety of data.

iv. Object exchange protocol (OBEX)

- It is developed by IrDA to exchange objects simple and spontaneous manner.
- It uses client-server model.
- It is independent of transport mechanism and transport ‘Application programming Interface (API)’, provided it realizes a reliable transport base.
- It defines a folder-listing object, which is used to browse contents of folders on a remote device.

v. Networking layers

- Bluetooth wireless technology uses peer-to-peer network topology.
- Dial-up networking uses AT commands.
- In most cases, network accessed is IP network with use of standard protocols like TCP, UDP, HTTP
- A device can connect to IP network using network access point. The internet PPP is used to connect to access point.

vi. Telephone control specifications layer (TCS) and audio

- This layer is designed to set up voice calls. It supports functions like call control and group management.
- TCS can also be used to set up data calls.
- TCS protocols are compatible with ITU Specifications.
- Bluetooth audio communication takes place at rate of 64Kbps using one of two encoding schemes: 8-bit logarithmic PCM or continuous variable slope delta modulation.

3. Application group

- This group consists of actual applications that make use of Bluetooth links and refers to software that exists above protocol stack.
- The Bluetooth-SIG does not define any application protocols nor does it specify any API. Bluetooth profiles are developed to establish a base point for use of a protocol stack to accomplish a given usage case.

Bluetooth Devices:

- Every Bluetooth device consists of a built in short range **radio transmitter**. The current data rate is1 Mbps.
- So an interface between the IEEE 802.11 wireless LAN and Bluetooth LAN is possible.
- Bluetooth specification standard defines a short-range(10 meter) radio link.
- The devices carrying Bluetooth-enabled chips can easily transfer data through walls, clothing and luggage bags.
- The interaction between devices occurs by itself without direct human intervention whenever they are within each other’s range.

- Each Bluetooth-enabled device contains a 1.5 inch square transceiver chip operating in the ISM band of 2.40 GHz to 2.48 GHz.
- The ISM band is divided into 79 channels with each carrying a bandwidth of 1 MHz.

Bluetooth Applications:

1. It is used for providing communication between peripheral devices like wireless mouse or keyboard with the computer.
2. It is used by modern healthcare devices to send signals to monitors.
3. It is used by modern communicating devices like mobile phone, PDAs, palmtops etc to transfer data rapidly.
4. It is used for dial up networking. Thus allowing a notebook computer to call via a mobile phone.
5. It is used for cordless telephoning to connect a handset and its local base station.
6. It also allows hands-free voice communication with headset.
7. It also enables a mobile computer to connect to a fixed LAN.
8. It can also be used for file transfer operations from one mobile phone to another.

Advantages of Bluetooth:

- Bluetooth is inexpensive
- Bluetooth provides low interference
- It require low energy consumption
- It allows sharing of data
- It is cheaper in cost
- Easy to use
- Disadvantage of Bluetooth
- It only allows short range (30 feet) communication between devices.
- Bluetooth only offers 1 mbps data transfer rate.

Question: In Bluetooth communication calculate the length of frame for following scenarios: (i) Three slot (ii) Five slot

Answer: Assume data rate = 1 mbps

In Bluetooth communication, when the link speed or data rate is

1Mbps each slot length is 625µs or 1600 hops/sec Packets can be of 1, 3, 5 slots.

i) Since each slot length is 625µs, Total length of the frame containing three slots is

625*3=1875µs, Or 1600*3=4800 hops/sec

ii) Since each slot length is 625µs, Total length of the frame containing five slots is

625*5=3125µs, Or 1600*=8000 hops/sec.

Smart Bluetooth

- In 2010, Sony started the development of a Bluetooth version called smart Bluetooth, a smaller low powered version of Bluetooth, which targeted the market of fitness and healthcare.
- Smart Bluetooth needed to be small and power efficient.
- However small and power efficiency is loved by all so a part of the Sony development became a part of the Bluetooth version 4.0 standard.
- This standard is also known as BLE which stands for Bluetooth low energy.
- Bluetooth low energy is a wireless personal area network technology designed and marketed by the Bluetooth special interest group.
- Mobile operating systems including iOS, Android, windows phone, blackberry, as well as macOS, Linux, windows 8 and windows 10 natively support Bluetooth low energy.
- There are two trademarks from the Bluetooth SIG as bellow

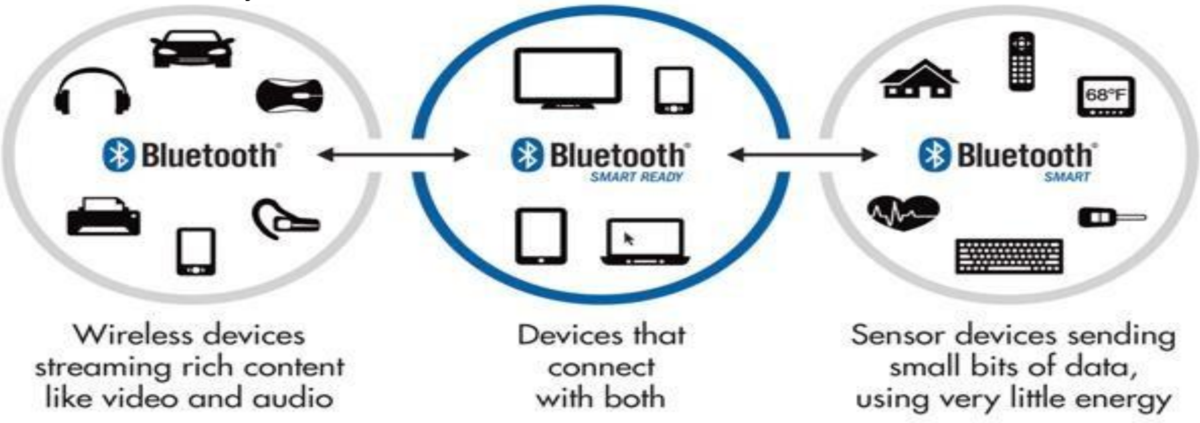
1. Bluetooth Smart Ready (HUBS):

- Bluetooth Smart Ready devices are the devices that receive data sent from the classic Bluetooth and Bluetooth smart devices and give it to applications that make use of that data
- The applications could be running on these devices themselves or could be running anywhere else on the internet
- E.g. phones, tablets PCs etc

2. Bluetooth smart (sensor type devices):-

- Bluetooth smart devices are sensor type devices that are used to collect a specific piece of information.
- After collecting this information these devices send it to the Bluetooth smart ready devices.
- Eg heart rate monitor, thermometers, sports equipment etc.

These devices collect a specific piece of information like heart rate or temperature and then relay it to the Bluetooth smart ready devices



- As shown in fig the mobile phone is the smart ready devices and it is communicating to two devices at the same time.

1. Streaming audio data to a Bluetooth headset.

2. Collecting temperature information from a Bluetooth smart thermometer and acting as a hub to relay that information to a server located in the hospital. The server can then take the appropriate action like informing the doctor or pharmacist.

2. Near Field Communication (NFC)

- NFC is a currently emerging and yet promising area which will have an enormous impact on the mobile technology throughout the world within just a few years

- NFC refers to wireless communication technology over short distance (less than 10 cm)
- NFC is a simple but profound technology that is fast evolving along with other mobile technologies in the market.
- This technology enables interaction between the virtual mobile world and the physical world.
- NFC is a wireless communication technology that potentially facilitates mobile phone usage of billions of people throughout the world offers an enormous number of use cases including credit cards, debit cards, loyalty cards, car keys, access keys for hotels, offices and houses, e-payments, e-ticketing, smart advertising, data money transfer and social services, eventually integrating all such materials into single mobile phone.
- NFC is a short range, high frequency, low bandwidth and wireless communication technology between two NFC enabled devices.
- Communication between NFC devices occurs at 13.56MHz high frequency which was originally used by radio frequency identification (RFID)

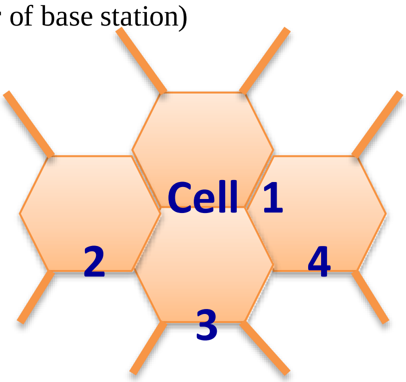
There are two different roles that a device can play in NFC which can be illustrated as a "request and reply" concept as shown in fig.

- The initiator sends a request message to a target and the target replies by sending a message back to the initiator.
- In this case the role of the initiator is to start the communication.
- The role of the target is to respond to the requests coming from the initiator.
- An active device can act as both an initiator and a target. However, a passive device cannot be an initiator.

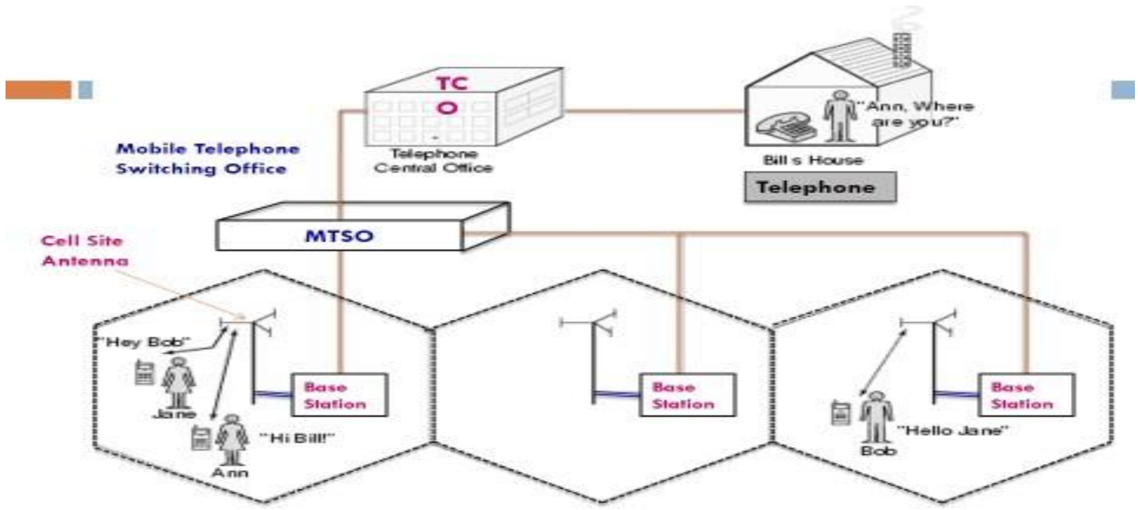
Mobile Telephone System

Basic Concept

- **Mobile or Cellular phone** is wireless communication just like cordless phone.
- In cell phone distance is not restricted to within home but one can travel in the city or even outside the city without interruption in communication.
- The demand for cellular phone is increasing at alarming level and is likely that wired communication will be replaced by wireless technology.
- In the cellular system city is divided into small areas called '**Cells'**. Each cell is around 10 square kilometer. (Depends upon power of base station)



- The cells are normally thought of hexagons. Because cell phones and base stations use low power transmitters, the same frequencies can be reused in non-adjacent cell.



- The Cellular network is as shown in figure.
- A cellular network system is formed by connecting the following five components

1. Mobile Station (MS):-

- MS are usually a mobile phone. Each mobile phone contains a transceiver (transmitter and receiver) an antenna and control circuitry.
- Antenna converts the transmitted RF signal into an EM wave and the received EM waves into an RF.

- The same antenna is used for both transmission and reception so there is a duplex switch to multiplex the same antenna.

2. Base Station:-

- At the cell site **base station** is equipped to transmit, receive, and switch calls to and from any mobile unit within the cell to the MTSO.
- The cell just covers only few square kilometer areas, thus reducing the power requirement necessary to communicate with cellular phones

3. MTSO:-

- Each cell is linked to central location called the **Mobile Telephone Switching Office (MTSO)**.
- MTSO coordinates all mobile calls between an area comprised of several cell sites and the central office. Time and billing information for each mobile unit is accounted for by MTSO.

4. Base station controller:-

- A number of BS are connected to a BSC.
- An important function of BSC is that it manages the "handoff" from one BS to another as a subscriber moves from cell to cell
- The BSC contains logic to control each of the BSs

5. PSTN (Public switched Telephone Network):-

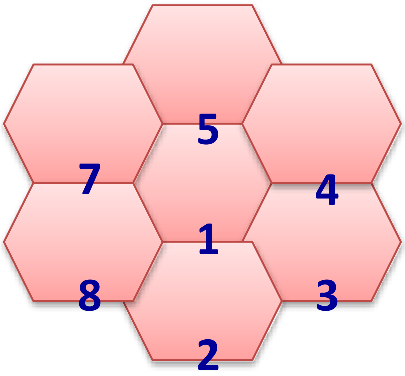
- It is a cellular network that can be viewed as an interface between mobile units and a telecommunication infrastructure. Therefore the PSTN network is nothing but the land based section of the network.
- It is necessary that the BSs are to be connected to a switching network and that network is to be connected to other networks such as the PSTN so that calls can be made to and from mobile subscribers.

6. Cell:

- The basic geographic unit of a cellular communication system is called as a **Cell**.
- Its shape is hexagonal.
- The size of cell is not fixed.
- Practically the shape of the cell may not be a perfect hexagon.

7. Cluster:

- A group of cells is called as a **Cluster**.
- The cluster size(n) is not fixed.
- It depends on the requirement of a particular area.
- Fig. shows the cluster of SEVEN cells or a SEVEN cell cluster (n=7).

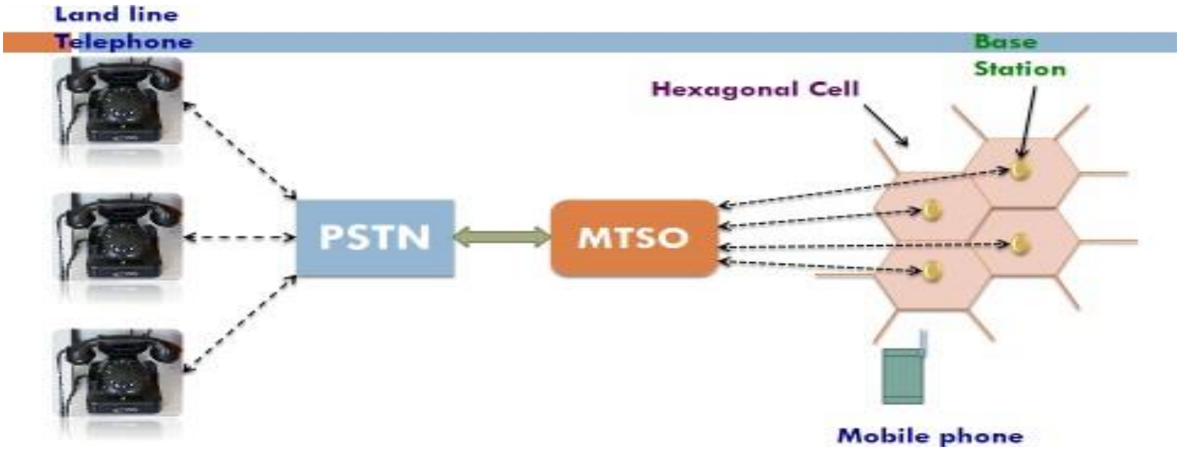


Bands in Cellular Telephony

- Classically analog communication is used for the cellular telephony. Frequency Modulation (FM) is used for communication between the mobile phone and the cell office.
- Generally two frequency bands are allocated for this purpose. One for the communication initiated by the cell phone and for the land phone.
- For cellular communication, the FCC has appointed 40 MHz of the frequency spectrum from 825 to 845 MHz and 870 to 890 MHz . Full-duplex operation is possible by separating transmit and receive signals into separate frequency bands. Cellular phone units transmit in the lower band of frequencies, 825 to 845 MHz, and receive in the higher band, 870 to 890 MHz.
- The opposite frequency bands are used by the base units at the cell sites. Within these two bands, 666 separate channels (333 channels per band) have been assigned for voice and control. Each channel occupies a bandwidth of 30 KHz.

Basic Structure of the Mobile Phone System

- In the **mobile communication** system either the transmitter or the receiver or both are going to be movable. As the points between which the communication takes place are movable, the communication channel is essentially air, which means it is a wireless communication.
- The structure of the mobile phone network along with the PSTN is shown in figure.



- Each cell has a Base station situated at the center.
- The task of the Base stations is to act as an interface between the mobile phone and the cellular radio system.
- The Base station of all the cells is connected to the MTSO.
- The interface is a bi-directional [i.e. Exchange of information between MTSO and Base station is a two way].
- The MTSO acts as the interface between the PSTN and the Base station. PSTN performs the supervision and control operations in the mobile communication system.
- The communication can take place between two Mobile subscribers or between a mobile subscriber and a Landline Telephone.
- If a mobile subscriber travels from one cell area to the other then it automatically gets connected to the Base station of that cell. Thus the service provided to a mobile subscriber is continuous without any break.

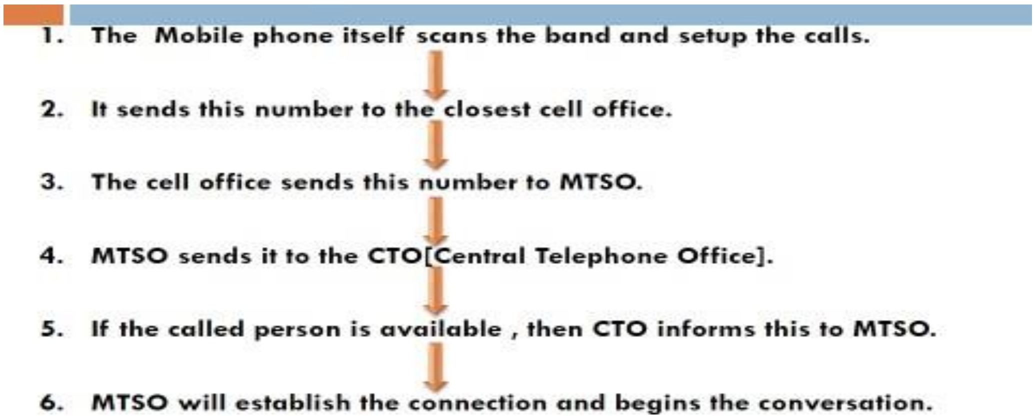
Function of MTSO:

- The MTSO control all the cells and provides the interface between each cell and the main telephone office.
- As the vehicle moves from one cell to the next the system automatically switches from one cell to the next.
- The MTSO switches from the vehicle to the stronger cell within a very short time.

Calls using Mobile Phones

Case 1 : Call initiated by a mobile phone :

- When we make a call from the mobile by entering the required 10 digit phone number the sequence of events takes place as follows:



Transmitting/Receiving/Handoff Operations

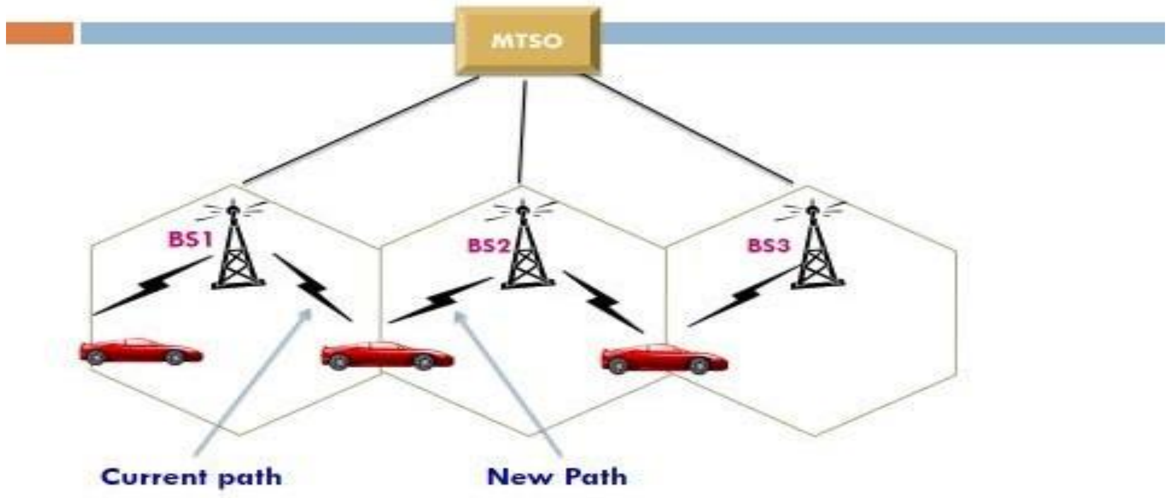
Hand off Procedure:

- During the conversation, if a Mobile phone crosses the exiting cell, the signal became weak.
- The MTSO is checking the signal level continuously, so if it finds signal level low then it immediately switch the call which can improve the signal strength.
- The MTSO will then change the cell carrying channel very smoothly without interrupting the call or changing user.
- This process of handling the signal of Old channel to the new channel is called as **Hand-off / Handover** Procedure.
- The user can continue talking without even noticing that the Hand-off Procedure has taken place.

Different types of Hand Offs:

Following are various types of handoffs supported by a Mobile Station:

1. Hard Hand off
2. Soft Hand off
3. Delayed Hand off
4. Forced Hand off
5. Queued Hand off



1. Hard Hand Off:

- The hand off is known as Hard Handoff if a mobile station transmits between two base stations having different frequency assignments.

2. Soft Hand Off:

- The hand off is known as soft handoff if the mobile station starts communication with a new base station without stopping the communication with the older base station.

3. Delayed Hand Off:

- In many situations, instead of one level, a two level handoff procedure is used, in order to provide a high opportunity for a successful handoff.
- A hand off can be delayed if no available cell could take the call.
- If due to some reason the mobile unit is in a hole (Place in a cell with low signal level) or neighboring cell is busy then the handoff is requested after every 5 seconds. But if the signal strength becomes lower and reaches the second handoff level then the handoff will take place without any condition, immediately this process is called **Delayed Handoff.**

4. Forced Hand Off:

- A forced hand off is defined as the hand off which could normally occur but is prevented from happening or a hand off that should not occur but is forced to happen.

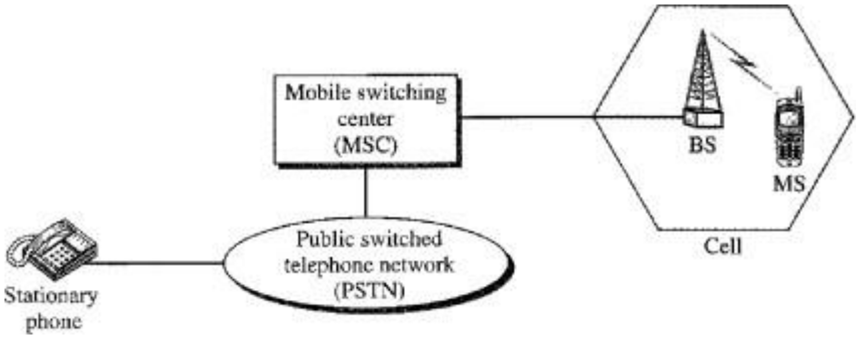
5. Queued Hand Off:

- In the queued hand off process, the MTSO arranges the hand off requests in a queue instead of rejecting them, if the new cell sites are busy.
- These hand off requests are then entertained in a sequential manner. Queuing of hand offs is more effective than the two threshold hand off. Also, a queuing scheme is effective only when the hand off requests arrive at the MTSO in batches or bundles.

Question: Draw and explain Mobile Telephone System Architecture.
Answer:

Cellular telephony is designed to provide communications between two moving units, called mobile stations (MSs), or between one mobile unit and one stationary unit, often called a land unit. A service provider must be able to locate and track a caller, assign a channel to the call, and transfer the channel from base station to base station as the caller moves out of range.

To make this tracking possible, each cellular service area is divided into small regions called cells. Each cell contains an antenna and is controlled by a solar or AC powered network station, called the base station (BS). Each base station, in turn, is controlled by a switching office, called a mobile switching center (MSC). The MSC coordinates communication between all the base stations and the telephone central office. It is a computerized center that is responsible of connecting calls, recording call information, and billing.



Cell size is not fixed and can be increased or decreased on the population of the area. The typical radius of a cell is 1 to 12mi. High density areas require more, geographically smaller cells to meet traffic demands than do low-density areas. Once determined, cell size to optimize to prevent the interference of adjacent cell signals. The transmission power of each cell is kept low to prevent its signal from interfering with those of other cells.

Generations of Mobile Telephone System

In the past few decades, mobile wireless technologies have experience 4 or 5 generations of technology revolution and evolution, namely from 0G to 4G. Current research in mobile wireless technology concentrates on advance implementation of 4G technology and 5G technology. Currently 5G termis not officially used.

0G Wireless technology
0G refers to pre-cell phone mobile telephony technology, such as radio telephones that some had in cars before the advent of cell phones. Mobile radio telephone systems preceded modern cellular mobile telephony technology. Since they were the predecessors of the first generation of cellular telephones, these systems are called 0G (zero generation) systems.

1G: Analog Cellular Networks
The main technological development that distinguished the First Generation mobile phones from the previous generation was the use of multiple cell sites, and the ability to transfer calls from one site to the next as the user travelled between cells during a conversation. The first commercially automated cellular network (the 1G generations) was launched in Japan by NTT in 1979.

In 1984, Bell Labs developed modern commercial cellular technology, which employed multiple, centrally controlled base stations (cell sites), each providing service to a small area (a cell). The cell sites would be set up such that cells partially overlapped. In a cellular system, a signal between a base station (cell site) and a terminal (phone) only need be strong enough to reach between the two, so the same channel can be used simultaneously for separate conversations in different cells.

As the system expanded and neared capacity, the ability to reduce transmission power allowed new cells to be added, resulting in more, smaller cells and thus more capacity.

2G: Digital Networks
In the 1990s, the 'second generation' (2G) mobile phone systems emerged, primarily using the GSM standard. These 2G phone systems differed from the previous generation in their use of digital transmission instead of analog transmission, and also by the introduction of advanced and fast phone-to-network signaling. The rise in mobile phone usage as a result of 2G was explosive and this era also saw the advent of prepaid mobile phones.

The second generation introduced a new variant to communication, as SMS text messaging became possible, initially on GSM networks and eventually on all digital networks. Soon SMS became the communication method of preference for the youth. Today in many advanced markets the general public prefers sending text messages to placing voice calls.

Some benefits of 2G were Digital signals require consume less battery power, so it helps mobile batteries to last long. Digital coding improves the voice clarity and reduces noise in the line. Digital signals are considered environment friendly. Digital encryption has provided secrecy and safety to the data and voice calls. The use of 2G technology requires strong digital signals to help mobile phones work properly.

"2.5G" using GPRS (General Packet Radio Service) technology is a cellular wireless technology developed in between its predecessor, 2G, and its successor, 3G. GPRS could provide data rates from 56 kbit/s up to 115 kbit/s. It can be used for services such as Wireless Application Protocol (WAP) access, Multimedia Messaging Service (MMS), and for Internet communication services such as email and World Wide Web access.

2.75 – EDGE is an abbreviation for Enhanced Data rates for GSM Evolution. EDGE technology is an extended version of GSM. It allows the clear and fast transmission of data and information up to 384kbit/s speed.

3G : High speed IP data networks

As the use of 2G phones became more widespread and people began to use mobile phones in their daily lives, it became clear that demand for data services (such as access to the internet) was growing. Furthermore, if the experience from fixed broadband services was anything to go by, there would also be a demand for ever greater data speeds. The 2G technology was nowhere near up to the job, so the industry began to work on the next generation of technology known as 3G. The main technological difference that distinguishes 3G technology from 2G technology is the use of packet switching rather than circuit switching for data transmission.

The high connection speeds of 3G technology enabled a transformation in the industry: for the first time, media streaming of radio and even television content to 3G handsets became possible. In the mid 2000s an evolution of 3G technology begun to be implemented, namely High-Speed

Downlink Packet Access (HSDPA). It is an enhanced 3Gmobile telephony communications protocol in the High-Speed Packet Access (HSPA) family, also coined 3.5G, 3G+ or turbo 3G, which allows networks based on Universal Mobile Telecommunications System (UMTS) to have higher data transfer speeds and capacity. Current HSDPA deployments support down-link speeds of 1.8, 3.6, 7.2 and 14.0 Mbit/s. Further speed increases are available with HSPA+, which provides speeds of up to 42 Mbit/s downlink and 84 Mbit/s with Release 9 of the 3GPP standards.

4G: Growth of mobile broadband

Consequently, the industry began looking to data-optimized 4th-generation technologies, with the promise of speed improvements up to 10-fold over existing 3G technologies. It is basically the extension in the 3G technology with more bandwidth and services offers in the 3G. The expectation for the 4G technology is basically the high quality audio/video streaming over end to end Internet Protocol. The first two commercially available technologies billed as 4G were the WiMAX standard and the LTE standard.
One of the main ways in which 4G differed technologically from 3G was in its elimination of circuit switching, instead employing an all-IP network. Thus, 4G ushered in a treatment of voice calls just like any other type of streaming audio media, utilizing packet switching over internet, LAN or WAN networks via VoIP.

4G LTE data transfer speed can reach peak download 100 Mbit/s, peak upload 50 Mbit/s, WiMAX offers peak data rates of 128 Mbit/s downlink and 56 Mbit/s uplink.

What is VoLTE?

VoLTE stands for voice over Long Term Evolution. Utilising IMS technology, it is a digital packet voice service that is delivered over IP via an LTE access network.
Voice calls over LTE are recognised as the industry-agreed progression of voice services across mobile networks, deploying LTE radio access technology.

What are the benefits of VoLTE?

The implementation of VoLTE offers many benefits, both in terms of cost and operation. VoLTE:

- Provides a more efficient use of spectrum than traditional voice;
- Meets the rising demand for richer, more reliable services;
- Eliminates the need to have voice on one network and data on another;
- Unlocks new revenue potential, utilising IMS as the common service platform;
- Can be deployed in parallel with video calls over LTE and RCS multimedia services, including video share, multimedia messaging, chat and file transfer;
- Ensures that video services are fully interoperable across the operator community, just as voice services are, as demand for video calls grows;
- Increases handset battery life by 40 per cent (compared with VoIP);
- Delivers an unusually clear calling experience; and
- Provides rapid call establishment time.

Question: Enlist generations of mobile telephone system.

Answer: Generations of mobile telephone system:

- First Generation
- Second Generation:2.5G, 2.75G
- Third Generation:3.5, 3.75G
- Fourth Generation
- Fifth Generation

Question: Compare first, second, third and fourth generation mobile telephone systems (any 3 points).

Technology	1G	2G/2.5G	3G	4G
Bandwidth	2Kbps	14-64kbps	2Mbps	200Mbps
Technology	Analog cellular	Digital cellular	Broadband width/CDMA/IP Technology	Unified IP and seamless combo of LAN/WAN/WLAN
Service	Mobile telephony	Digital voice, Short messaging	Integrated high quality audio, video and data	Dynamic information access, variable devices.
Multiplexing	FDMA	TDMA/CDMA	CDMA	CDMA
Switching	Circuit	Circuit/circuit for access network and air	Packet except for air interface	All packet