BHADRAK ENGINEERING SCHOOL & TECHNOLOGY (BEST), ASURALI, BHADRAK

# Analog & Digital Communication (Th- 03)

(As per the 2020-21 syllabus of the SCTE&VT, Bhubaneswar, Odisha)


## Fifth Semester

## Electronics & Tele-comm. Engg.

**Prepared By:** *Er K M Jena & Er D Mohapatra*

# ANALOG & DIGITAL COMMUNICATION

## CHAPTER-WISE DISTRIBUTION OF PERIODS & MARKS

| Sl. No. | Chapter/ Unit No. | Name of The Chapter/ Unit | Periods as per Syllabus | Expected Marks |
|---|---|---|---|---|
| 01 | 01 | Elements of Communication Systems. | 10 | 15 |
| 02 | 02 | Amplitude (Linear) Modulation System. | 15 | 20 |
| 03 | 03 | Angle Modulation Systems. | 10 | 15 |
| 04 | 04 | Am & Fm Transmitter & Receiver. | 8 | 15 |
| 05 | 05 | Analog To Digital Conversion & Pulse Modulation System. | 17 | 25 |
| 06 | 06 | Digital modulation Techniques. | 15 | 20 |
| **TOTAL** | | | **75** | **110** |

# UNIT-1

# ELEMENTS OF COMMUNICATION SYSTEMS

**LEARING OBJECTIVES:**

**1.1** *Communication Process- Concept of Elements of Communication System & its Block diagram*
**1.2** *Source of information & Communication Channels*
**1.3** *Classification of Communication systems (Line & Wireless or Radio)*
**1.4** *Modulation Process, need of modulation and classify modulation process*
**1.5** *Analog and Digital Signals & its conversion*
**1.6** *Basic concept of Signals & Signals classification (Analog and Digital)*
**1.7** *Bandwidth Limitation*

## 1.1 Communication Process- Concept of Elements of Communication System & its Block diagram

**Communication Process: -**

- Communication means exchanging information.
- The process of transmission and reception of information is called communication process.

**Types of Communication Systems: -**

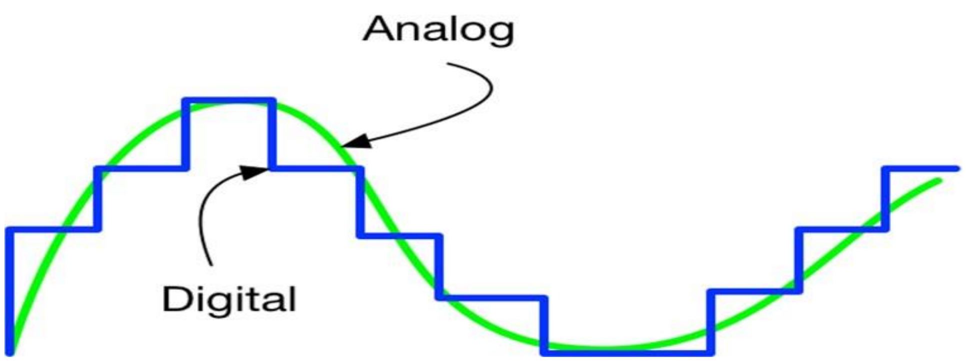
- Depending on Signal specification or technology, the communication system is classified as follows:

**(1) Analog:**

- Analog technology communicates data as electronic signals of varying frequency or amplitude. Broadcast and telephone transmission are common examples of Analog technology.

**(2) Digital:**

- In digital technology, the data are generated and processed in two states: High (represented as 1) and Low (represented as 0). Digital technology stores and transmits data in the form of 1s and 0s.



## Examples of Communication Systems: -

- The following are a few examples of communication systems:

  1. Internet
  2. Public Switched Telephone network
  3. Television

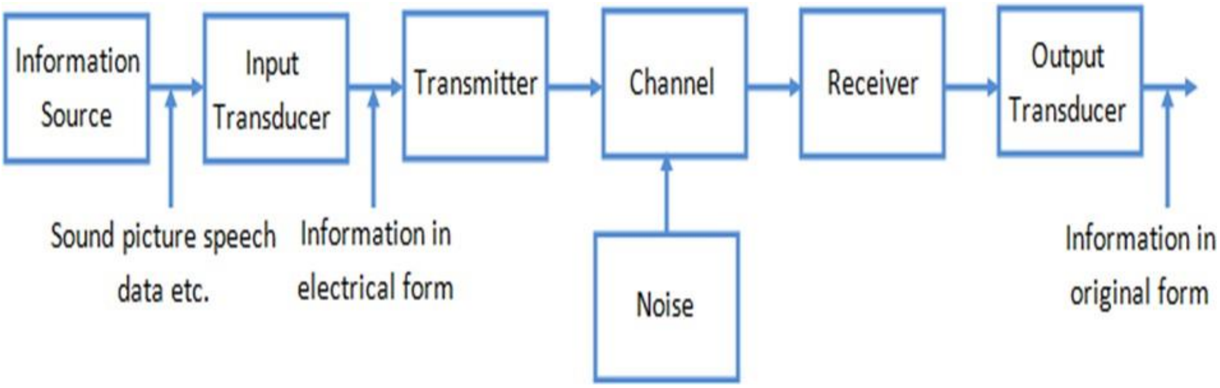
## Elements of Communication System: -

- The communication system is a system which describes the information exchange between two points. The major elements of communication are:

1. Transmitter.
2. Channel or medium.
3. Receiver.

## Block Diagram of Communication System: -

Below Figure shows the block diagram of a general communication system, in which the different functional elements are represented by blocks.




## (I) Information Source: -

• The function of information source is to produce required message which has to be transmitted. In general, there can be various messages in the form of words, group of words, code, symbols, sound signal etc.

## (ii) Input Transducer: -

• A transducer is a device which converts one form of energy into another form. In this case the input signal may it be sound, pressure, light, video signal etc. are converted into electrical signal.

## (iii) Transmitter: -

• The function of the transmitter is to transmit the modulated signal with high intensity in electrical form.

## (iv) The Channel and the Noise: -

- The term channel means the medium through which the message travels from the transmitter to the receiver. In other words, we can say that the function of the channel is to provide a physical connection between the transmitter and the receiver.

- During the process of transmission and reception the signal gets distorted due to noise introduced in the system. Noise is an unwanted signal which may interfere with signal at any point in a communication system.

**(v) Receiver: -**

- The main function of the receiver is to receive the message signal in electrical form and reproduction of the original signal is accomplished by a process known as demodulation or detection. Demodulation is the reverse process of modulation carried out in transmitter.

**(vi) Output Transducer: -**

- The function of output transducer is to convert the electrical signal into original form(information signal).

**(vii) Destination: -**

- Destination is the final point where the message signal intended to reach.

## 1.2 Source of information & Communication Channels.

➢ **Source of information & Communication Channels: -**

**Source of information: -**

- A source of information is one of the basic concepts of communication.

- Sources are objects which encode message data and transmit the information through channel to receivers.

**Communication Channels: -**

- A communication channel refers either to a physical transmission medium such as a wire, or to a logical connection such as air between the transmitter & receiver.

## 1.3 Classification of Communication systems (Line & Wireless or Radio):

**Classification of Communication systems (Line & Wireless or Radio): -**

- The communication system basically classified in to two types:

  1. **Line communication system.**
  2. **Wireless or Radio communication system.**

**1. Line/Wired communication system: -**

- Line/wired communication systems include all communications systems for which data is sent through a wire.

## ➢ Types of Line/ Wired Connections: -

1. **Twisted pair:** Consists of a pair of wires that are twisted together. The twisting reduces noise on the wires.

2. **Coaxial Cable:** Coaxial cables consist of a cylindrical wire running down the middle of an insulating sheath. Surrounding the insulating sheath is a conductive sheath. Coaxial cables are highly resistant to noise.

3. **Fiber Optic Cable:** A fiber optic cable consists of a very long thin fiber of glass down which light pulses can be sent. The data rates supported by fiber optic networks are incredibly faster.

## ➢ Advantages: -

- High immunity to outside interference and noise
- Allocation of frequencies is determined by the owner(s) of the wire, not by regulatory authorities.

## 2. Wireless or Radio communication system: -

- In this system, the information can be transmitted through the air without requiring any cable or wires or other electronic conductors,or by using electromagnetic waves like IR, RF, satellite, etc.

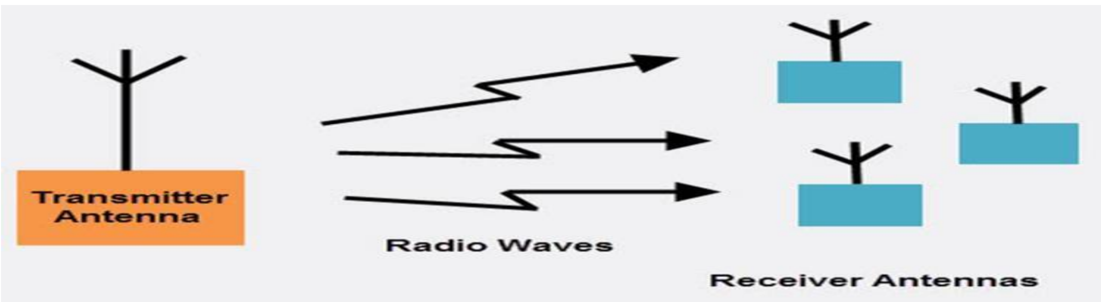
## ➢ Types of Wireless Communication: -

1. **Satellite Communication:** Satellite communication is widely spread all over the world to allow users to stay connected almost anywhere on the earth.

2. **Infrared Communication:** Infrared wireless communication communicates information in a device or systems through IR radiation.

3. **Broadcast Radio:**

- The first wireless communication technology is the open radio communication.
- Radio uses a transmitter which is used to transmit the data in the form of radio waves to a receiving antenna. A broadcast sends information over long distances at up to two megabits/Sec (AM/FM Radio).




- **Advantages of Wireless Communication: -**
  - Any data or information can be transmitted faster with a high speed.
  - Maintenance and installation are less costly for these networks.
  - The internet can be accessed from anywhere.
- **Disadvantages of Wireless Communication: -**
  - An unauthorized person can easily capture the wireless signals which spread through the air.
  - It is very important to secure the wireless network so that the information cannot be misused by unauthorized users
- **Applications of Wireless Communication: -**
  - Applications of wireless communication involve
    1) security systems
    2) Television remote control
    3) Wi-Fi, Cell phones
    4) Wireless power transfer
    5) Computer interface devices and various wireless communication-based projects.

## 1.4 Modulation Process, need of modulation and classify modulation process:

### Modulation: -

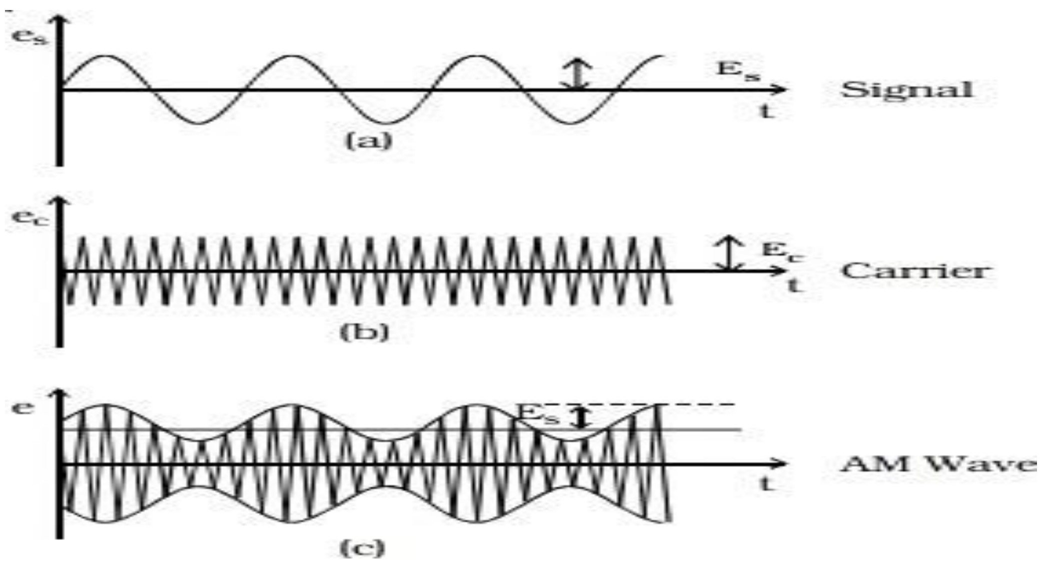
- The process of changing some characteristics (i.e. amplitude, phase, frequency) of carrier wave in accordance with the intensity of the information signal is known as modulation.
- And the resultant wave is a modulated wave which is to be transmitted.

### Types Of Modulation: -

- There are 3 types of Modulation: -

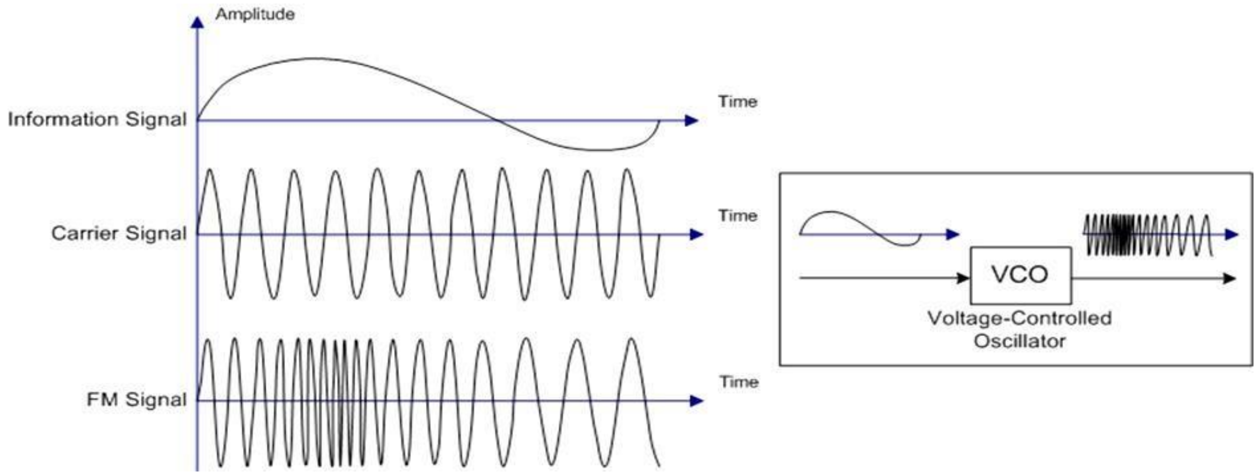
### I. Amplitude Modulation (AM)

- Amplitude of carrier wave varies in accordance with the intensity of the information signal is called amplitude modulation.



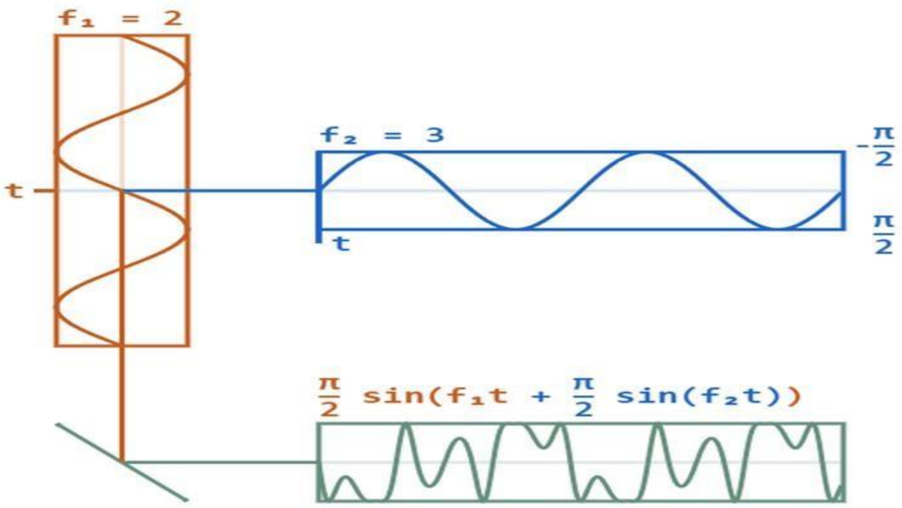
## ii. Frequency Modulation (FM)

- Frequency of carrier wave varies in accordance with the intensity of the information signal is called Frequency modulation.




## iii. Phase Modulation (PM)

- Phase of carrier wave varies in accordance with the intensity of the information signal is called Phase modulation.




- **Need of modulation: -**
  - Antenna length: If without modulation means less frequency original signal is transmitted, very high lengthy antenna is required.
  - Multiplexing: Transmission of multiple of message signals over a channel is known as multiplexing.
  - Narrow banding: If the baseband signal with frequency range 50hz to 10 khz is radiated directly in a broad cast system,a wideband antenna is required.

# 1.5 Analog and Digital Signals & its conversion.

## Basic concepts of signals: -

**Signal: -** A signal is a function that conveys information about a phenomenon. It refers to **any time varying voltage, current, or electromagnetic wave** that carries information.

## Classification: -

➢ Two main types of signals are:

1. Analog signals.
2. Digital signals.

### 1. Analog signals: -

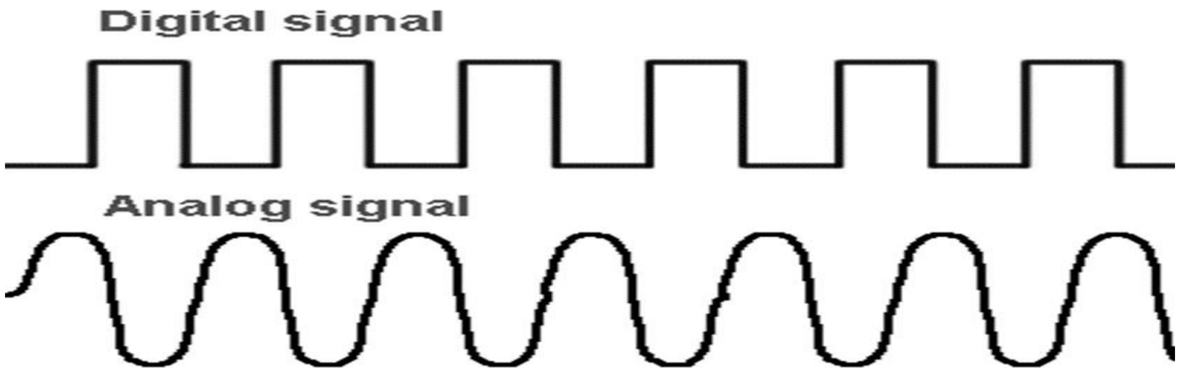
- An analog signal is any continuous signal for which the time-varying feature (variable) of the signal is a representation of some other time-varying quantity.

- For example, in an analog audio signal, the instantaneous voltage of the signal varies continuously with the pressure of the sound waves.

### 2. Digital signals: -

- A digital signal is a continuous-time physical signal, alternating between a discrete number of waveforms.

- Digital signals often arise via sampling of analog signals, for example, a continually fluctuating voltage on a line that can be digitized by an analog-to-digital converter circuit.




# 1.6 Basic concept of Signals & Signals classification (Analog and Digital):

## Signal conversion: -

➢ The conversion must be done if the measurement or the output signal does not meet input specification of the receiving device.

➢ **It is basically 2 types:**

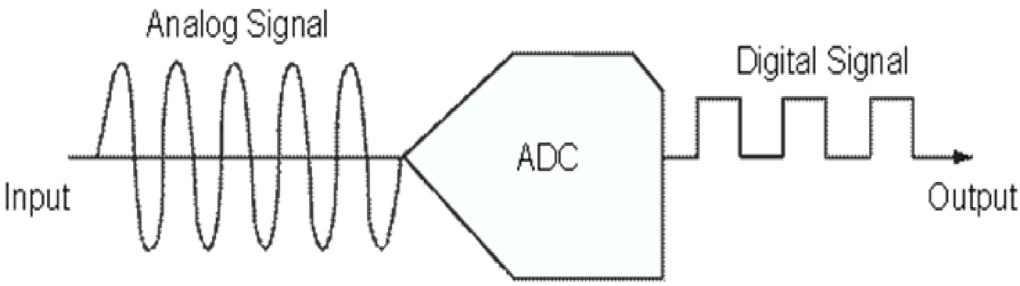
1. Analog to digital conversion. (A/D C)
2. Digital to analog conversion. (D/A C)

### 1. Analog to digital conversion (A/D C): -

- Analog-to-Digital converters (ADC) translate analog signals, like temperature, pressure, voltage, current, distance, or light intensity, into a digital representation of that signal. This digital representation can then be processed, manipulated, computed, transmitted or stored.

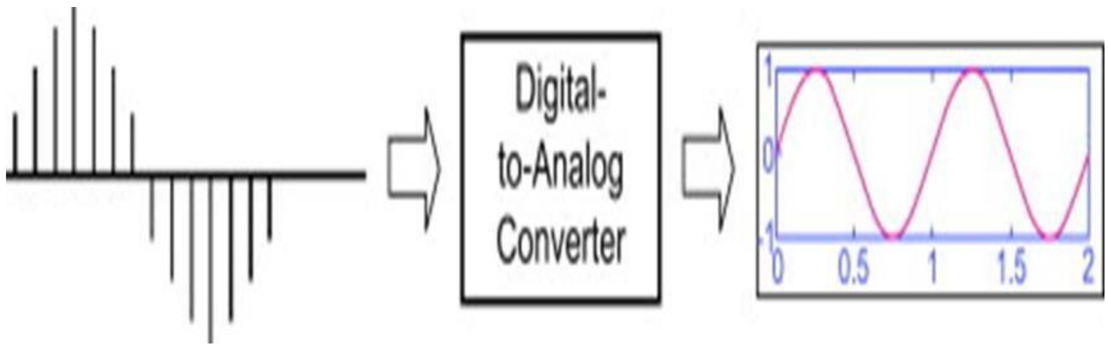
- The input to an analog-to-digital converter (ADC) consists of a voltage that varies among a theoretically infinite number of values. Examples are sine waves.

- The output of the ADC has defined levels or states. The number of states is almost always a power of two -- that is, 2, 4, 8, 16, etc.
- The simplest digital signals have only two states, and are called binary i.e. 0 or 1.




**2. <u>Digital to analog conversion (D/A C):-</u>**

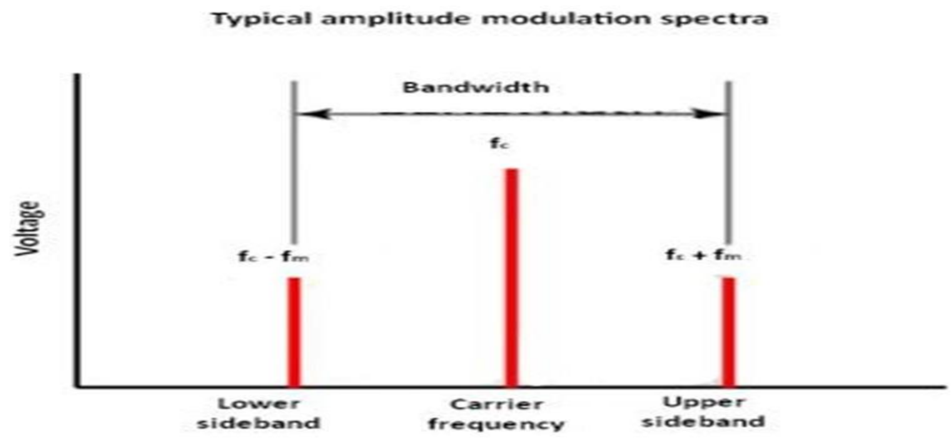
- Digital-to-analog conversion (DAC), Process by which digital signals (which have a binary state) are converted to analog signals (which theoretically have an infinite number of states).
- For example, a modem converts computer digital data to analog audio-frequency signals that can be transmitted over telephone lines.
- Figure shows a digital signal passing through a digital-to-analog (D/A or DAC) converter which transforms the digital signal into an analog signal and outputs of that signal to the environment.




## 1.7 Bandwidth limitation

**<u>Bandwidth: -</u>**

- Bandwidth is the difference between the upper and lower cut off frequencies in a continuous band of frequencies. It is typically measured in hertz.
- It is of two types. One is pass band bandwidth & other is baseband bandwidth.
- Passband bandwidth is the difference between the upper and lower cut off frequencies. For example- a band-pass filter, a communication channel, or a signal spectrum.
- Baseband bandwidth applies to a low-pass filter or baseband signal; the bandwidth is equal to its upper cut off frequency. For example- a base band signal.




(Where, fc=carrier frequency, fm=modulating frequency)

## Bandwidth limitation: -

- The information theory states that the greater is the transmission bandwidth of a communication system, the more is the information that can be transmitted.
- However, in AM radio the maximum modulating frequency is restricted up to 5 kHz and hence the maximum bandwidth of AM transmission is 10 kHz.

# POSSIBLE SHORT TYPE QUESTIONS WITH ANSWERS

**1. Define communication process.**

**Ans:** Communication means exchanging information.

•The process of transmission and reception of information is called communication process.

**2. Define Communication System? Write the elements of Communication System.**

**Ans:** The communication system is a system which describes the information exchange between two points. The major elements of communication are:

1. Transmitter.
2. Channel or medium.
3. Receiver.

**3. Define Bandwidth.**

**Ans:** Bandwidth is the difference between the upper and lower frequencies in a continuous band of frequencies. It is typically measured in hertz.

**4. Define Bandwidth limitation.**

**Ans:** The information theory states that the greater is the transmission bandwidth of a communication system, the more is the information that can be transmitted.

- However, in AM radio the maximum modulating frequency is restricted up to 5 kHz and hence the maximum bandwidth of AM transmission is 10 kHz.

**5. Define Digital to analog conversion (D/A C).**

**Ans:** Digital-to-analog conversion (DAC), Process by which digital signals (which have a binary state) are converted to analog signals (which theoretically have an infinite number of states).

**6. Define Analog to digital conversion (A/D C).**

**Ans:** Analog-to-Digital converters (ADC) translate analog signals, like temperature, pressure, voltage, current, distance, or light intensity, into a digital representation of that signal. This digital representation can then be processed, manipulated, computed, transmitted or stored.

- The simplest digital signals have only two states, and are called binary i.e. 0 or 1.

**7. Define modulation.**

**Ans:** The process of changing some characteristics (i.e. amplitude, phase, frequency) of carrier wave in accordance with the intensity of the information signal is known as modulation.

**8.How many types of communication systems are there and what are they?**

**Ans:** The communication system basically classified in to two types:

1. Line communication system.
2. Wireless or Radio communication system.

**9. State the need of modulation.**

**Ans:** In a communication system, the baseband signal of a low-frequency spectrum is translated to a high frequency spectrum. This is achieved through modulation.

- Two signals are involved in the modulation process. The baseband signal and the carrier signal.

- The baseband transmission has many limitations which can be overcome using modulation. In the process of modulation, the baseband signal is transmitted i.e., shifted from low frequency to high frequency.

**10. Define Source of information.**

**Ans:** A source of information is one of the basic concepts of communication.

- Sources are objects which encode message data and transmit the information, via a channel, to receivers.

**11. Define Communication Channels.**

**Ans:** A communication channel refers either to a physical transmission medium such as a wire, or to a logical connection such as air between the transmitter & receiver.

**12. Define Analog and Digital Signal. [W-20]**

**Ans:** An analog signal is any continuous signal for which the time-varying feature (variable) of the signal is a representation of some other time-varying quantity.

A digital signal is a continuous-time physical signal, alternating between a discrete number of waveforms.

## LONG QUESTIONS

1. Explain the Block Diagram of Communication System.
2. Explain the classification of communication system.
3. Define modulation & classify it with fig.
4. Define analog & digital Signal. Explain its conversion.
5. Define signal conversion. Explain analog to digital signal (A/DC) conversion.
6. Explain Wireless or Radio communication system with its advantages & disadvantages.
7. Define Modulation. Explain the need of modulation in detail. [W-20]

# UNIT- 2.0

# AMPLITUDE (LINEAR) MODULATION SYSTEM

## Learning Objectives:

2.1 Amplitude modulation & derive the expression for amplitude modulation signal, power relation in AM wave & find Modulation Index

2.2 Generation of Amplitude Modulation (AM)- Linear level AM modulation only.

2.3 Demodulation of AM waves (liner diode detector, square law detector & PLL).

2.4 Explain SSB signal and DSBSC signa.l

2.5 Methods of generating & detection SSB-SC signal (Indirect method only):

2.6 Methods of generation DSB-SC signal (Ring Modulator) and detection of DSB-SC signal.

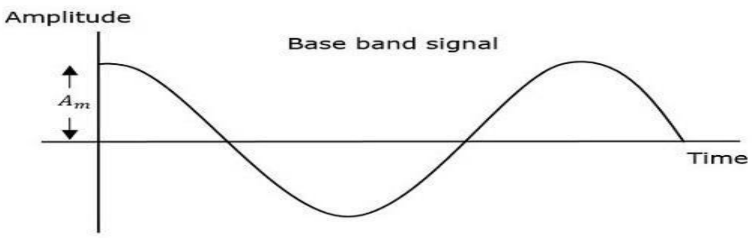
2.7 Concept of Balanced modulators.

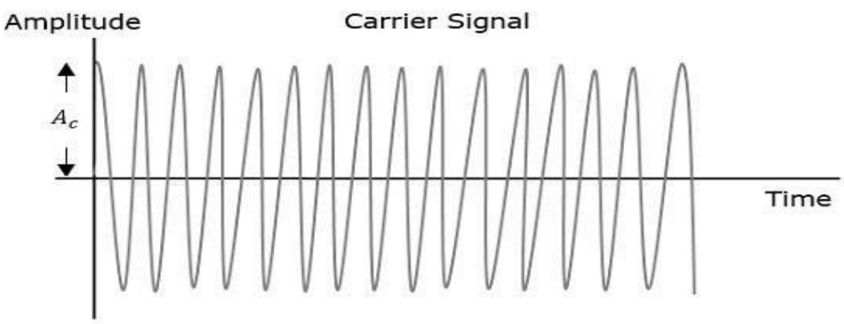
2.8 Vestigial Side Band Modulation.

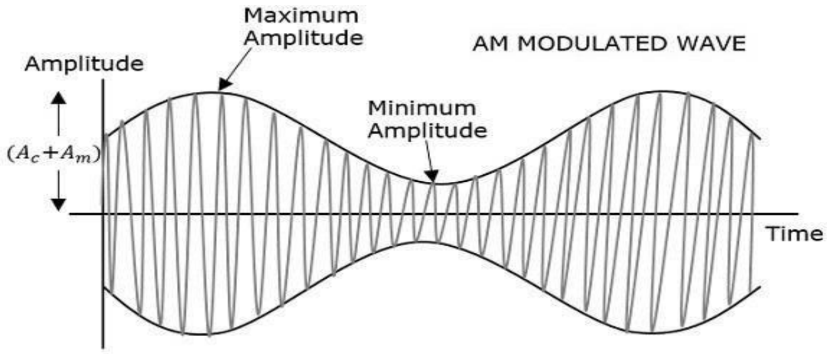
## 2.1 Amplitude modulation & derive the expression for amplitude modulation signal, power relation in AM wave & find Modulation Index:

- **Amplitude modulation**

- In amplitude modulation, the amplitude (signal strength) of the carrier wave is varied in proportion to that of the message signal, such as an audio signal. Here carrier frequency is greater than base band (Original) frequency.






## ➢ Amplitude Modulation Equation Derivation

- Here, one will get a clear knowledge of how to derive an expression for amplitude modulation.
- The mathematical representation of amplitude-modulated waves in the time domain is as follows.

Let,

modulating signal $m(t) = A_m \cos(2\pi f_m t)$

carrier signal $m(t) = A_c \cos(2\pi f_c t)$

So, equation of amplitude modulated wave $s(t) = [\, A_c + A_m \cos(2\pi f_m t) \,] \cos(2\pi f_c t)$

Where,

$A_m$: Amplitude of modulating signal

$A_c$: Amplitude of carrier signal

$f_m$: Frequency of modulating signal

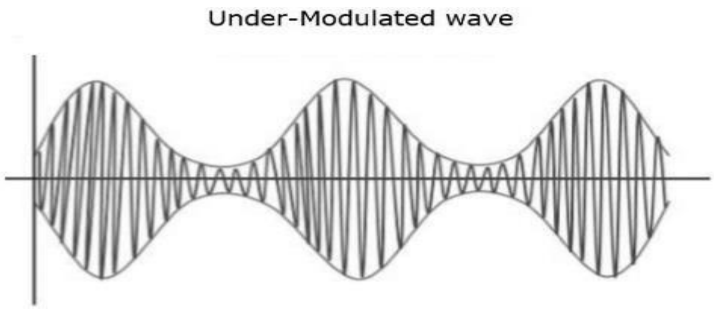
$f_c$: Frequency of carrier signal

## ➢ Modulation Index

- Modulation index defines the extent upto which amplitude of the carrier will be varied about an unmodulated maximum carrier.

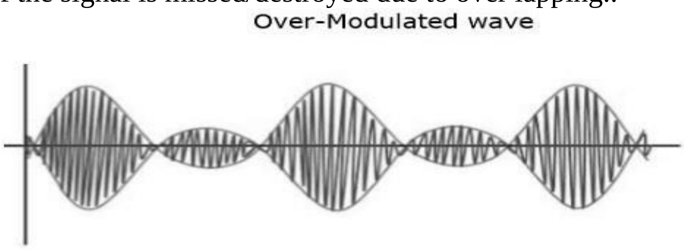
**Mathematically, m =Amplitude of modulating signal/Amplitude of carrier signal= $A_M$ / $A_C$**

- If this value is less than 1, i.e., the modulation index is 0.5, then the modulated output would look like the following figure. It is called as **Under-modulation**.



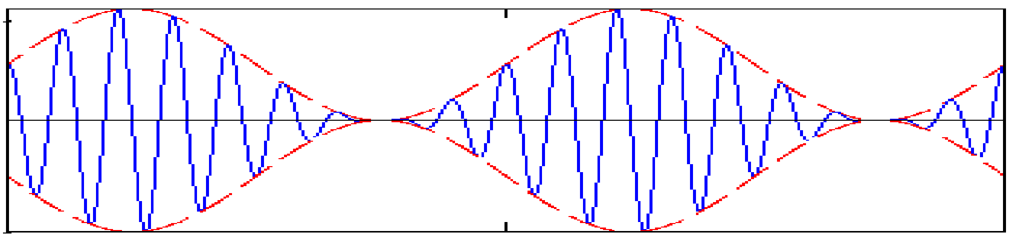

**M<1**

- If the value of the modulation index is greater than 1, i.e., 1.5 or so, then the wave will be an **over-modulated wave**. It would look like the following figure. Here some portion of the signal is missed/destroyed due to over lapping..




**m>1**

- If the value of the modulation index is equal to 1, i.e., 1, then the wave will be an **Equal modulated wave**. It would look like the following figure.



**m=1**

**Power Calculations of AM Wave**

- Consider the following equation of amplitude modulated wave.

$$s(t)=A_c\cos(2\pi f_c t)+\frac{A_m}{2}\cos[2\pi(f_c+f_m)t]+\frac{A_m}{2}\cos[2\pi(f_c-f_m)t]$$

=Carrier signal +upper side band +lower side band

- Power of AM wave is equal to the sum of powers of carrier, upper sideband, and lower sideband frequency components.

$$P_t=P_c+P_{USB}+P_{LSB}$$

We know that the standard formula for power of cos signal is

$$P=v_{rms}^2/R=(V_m)^2/2$$

Where, R is unit resistance

$V_{rms}$ is the rms value of cos signal.
$V_m$ is the peak value of cos signal.

$P_c=(A_C)^2/2$

**$P_{usb}$ = $P_{lsb}$=$Am^2/8$**

**So ,** $P_t=P_c+P_{USB}+P_{LSB}$

$= (A_C)^2/2+ Am^2/8+ Am^2/8$
$=(A_C)^2/2+ Am^2/4$
$= (A_C)^2/2\{1+ (Am/Ac)^2/2\}$
$=(A_C)^2/2\{1+ (m^2/2\}$

$=Pc\ (1+m^2/2)$

## 2.2 Generation of Amplitude Modulation (AM)- Linear level AM modulation only.

### Generation of Amplitude Modulation (AM)

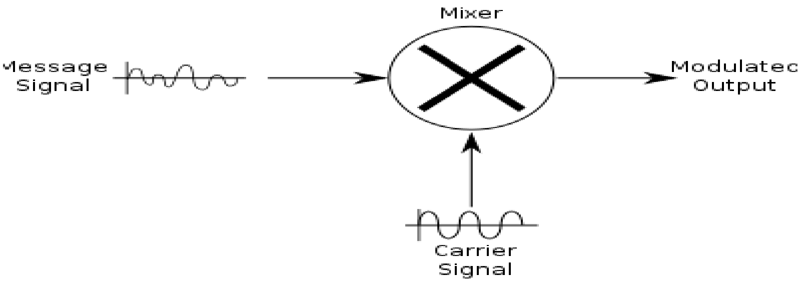
There is a variety of modulator circuits employing tubes or solid-state devices to produce amplitude modulated waves. They are

(a) Linear level modulator circuits.
(b) Square law or non-linear modulator circuits

- **Linear level AM modulation**

Amplitude modulation is characterized by the fact that the amplitude Ac of the carrier Signal Ac Cos(2πfct + φ) is varied in proportion to the baseband (message) signal m(t). Because **the amplitude is linearly related to the message signal**, this technique is also called linear modulation.

Let base band signal will be Am $\cos w_m(t)$ & Carrier signal will be Ac $\cos w_c(t)$. When both signals passes through a product modulator the output will be Am.Ac.$\cos w_m(t)$ Cos wc(t). If amplitude of carrier signal is unit or say 1, Then the out put will be Am/2 (Cos ( $w_c$-$w_m$)(t) +Cos( $w_c$+$w_m$)(t) ). Here we have two side bands having frequency ($w_c$-$w_m$) & ( $w_c$+$w_m$). In this way DSB-SC signals can be generated by a product modulator.




$$\underbrace{V_m \cos(\omega_m t)}_{\text{Message}} \times \underbrace{V_c \cos(\omega_c t)}_{\text{Carrier}} = \underbrace{\frac{V_m V_c}{2}\left[\cos((\omega_m+\omega_c)\,t) + \cos((\omega_m-\omega_c)\,t)\right]}_{\text{Modulated Signal}}$$

## 2.3 Demodulation of AM waves (liner diode detector, square law detector & PLL):

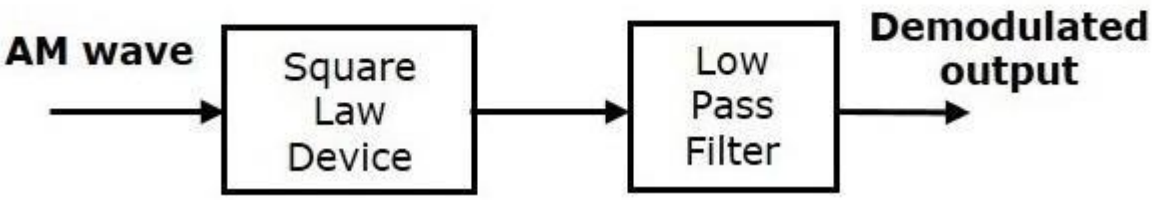
### Demodulation of AM waves:

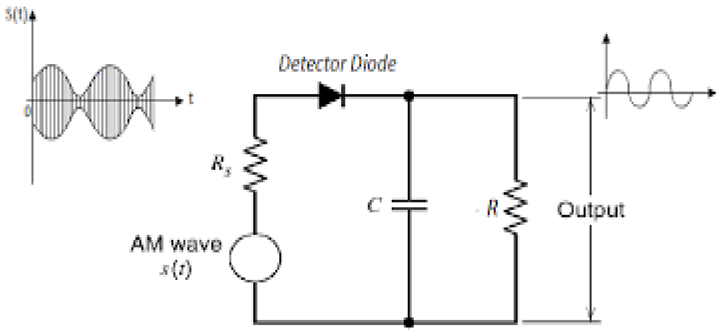
- The process of extracting an original message signal from the modulated wave is known as **detection** or **demodulation**. The circuit, which demodulates the modulated wave is known as the **demodulator**. The following demodulators (detectors) are used for demodulating AM wave.

1) Square Law Demodulator
2) Envelope Detector

### ➢ Square Law Demodulator

- Square law demodulator is used to demodulate low level AM wave. Following is the block diagram of the **square law demodulator**.




- This demodulator contains a square law device and a low pass filter. The AM wave $V_1(t)$ is applied as an input to this demodulator.
- The standard form of AM wave is

$$V_1(t)=A_c[1+mx(t)]\cos(2\pi f_c t)$$

- We know that the mathematical relationship between the input and the output of square law device is

$$V_2(t)=av_1+bv_{1(t)}^2$$

Where,

$V_1(t)$ is the input of the square law device, which is nothing but the AM wave
$V_2(t)$ is the output of the square law device

Substitute $V_1(t)$ in Equation

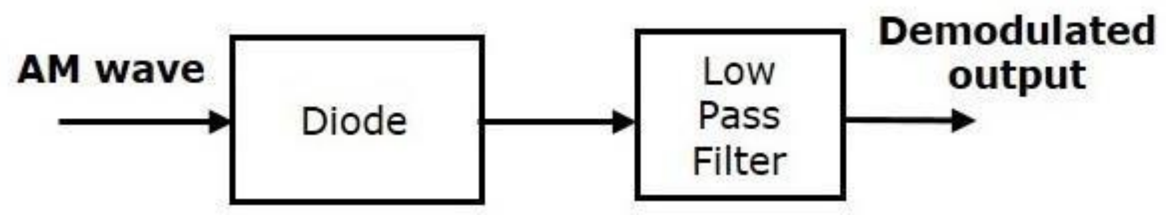
$$V_2(t)=a(Ac[1+mx(t)]\, Cos(2\pi fct)) + b\, Ac^2[1+mx(t)]^2 Cos^2(2\pi fct))$$

$\Rightarrow V2(t)=aAc(1+mx(t)\cos(2\pi fct) + bAc^2/2[1+2m\, x(t)+m^2x^2(t)]\,(1+\cos(4\pi fct))$

as $\cos^2 x= (1+ Cos\ 2x)/2$

In the above equation, the term $bAc^2\, mx(t)$ is the scaled version of the message signal which is due to the $bv_1^2$. Hence the name of this detector is square law detector. It can be extracted by passing the above signal through a low pass filter and the DC component can be eliminated with the help of a coupling capacitor. This means that we have recovered the message signal x(t) at the output of the detector.

- **Envelope Detector**
  - Envelope detector is used to detect (demodulate) high level AM wave. Following is the block diagram of the envelope detector.




**(BLOCK DIAGRAM )**

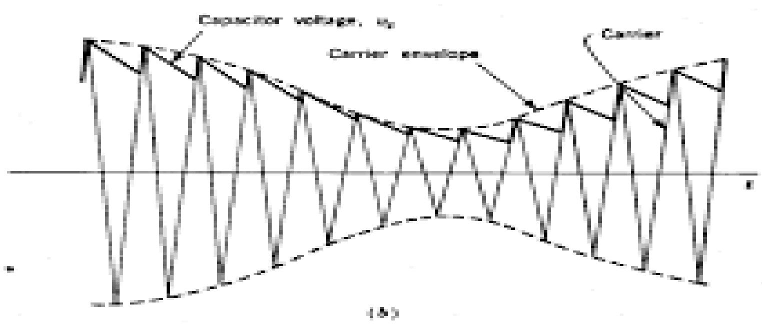
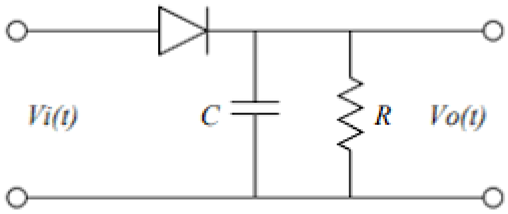



Figure 3.4-2 (a) A demodulator for an AM signal. (b) Input waveform and output voltage $v_c$ across capacitor.



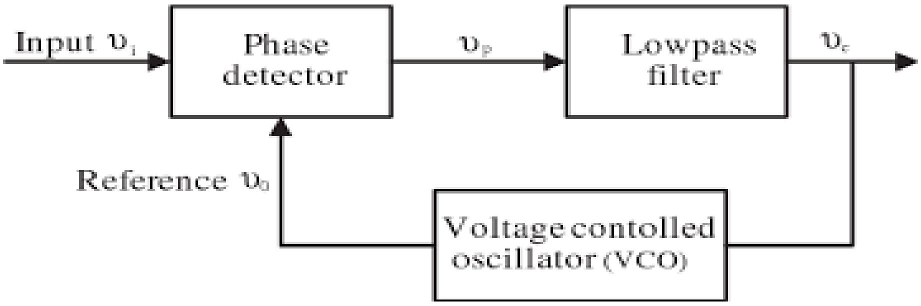

Envelope Detector

  - This envelope detector consists of a diode and low pass filter. Here, the diode is the main detecting element. Hence, the envelope detector is also called as the **diode detector**. The low pass filter contains a parallel combination of the resistor and the capacitor.
  - The AM wave s(t)s(t) is applied as an input to this detector.
  - We know the standard form of AM wave is

$$s(t)=Ac[1+kam(t)]\, Cos(2\pi fct)$$

- In the positive half cycle of AM wave, the diode conducts and the capacitor charges to the peak value of AM wave. When the value of AM wave is less than this value, the diode will be reverse biased.
- Thus, the capacitor will discharge through resistor **R** till the next positive half cycle of AM wave. When the value of AM wave is greater than the capacitor voltage, the diode conducts and the process will be repeated.
- We should select the component values in such a way that the capacitor charges very quickly and discharges very slowly. As a result, we will get the capacitor voltage waveform same as that of the envelope of AM wave, which is almost similar to the modulating signal.

**PLL:-**




- A **phase-locked loop** or **phase lock loop** (**PLL**) is a control system that generates an output signal whose phase is related to the phase of an input signal. The oscillator generates a periodic signal, and the phase detector compares the phase of that signal with the phase of the input periodic signal, adjusting the oscillator to keep the phases matched.
- Keeping the input and output phase in lock step also implies keeping the input and output frequencies the same. Consequently, in addition to synchronizing signals, a phase-locked loop can track an input frequency, or it can generate a frequency that is a multiple of the input frequency.
- Phase-locked loops are widely employed in radio , telecommunications, computers and other electronic applications. They can be used to demodulate a signal, recover a signal from a noisy communication channel, generate a stable frequency at multiples of an input frequency. Since a single integrated circuit can provide a complete phase-locked-loop building block, the technique is widely used in modern electronic devices, with output frequencies from a fraction of a hertz up to many gigahertz.

## 2.4 Explain SSB signal and DSBSC signal

- **Explain SSB signal**

- In radio communications, **single-sideband modulation** (**SSB**) or **single-sideband suppressed-carrier modulation** (**SSB-SC**) is a type of modulation used to transmit information, such as an audio signal, by radio waves..
- It uses transmitter power and bandwidth more efficiently.


- Single sideband modulation, SSB is the main modulation format used for analog voice transmission for two way radio communication on the HF portion of the radio spectrum.
- Its efficiency in terms of spectrum and power when compared to other modes means that for many years it has been the most effective option to use.
- Now some forms of digital voice transmission are being used, but it is unlikely that single sideband will be ousted for many years as the main format used on these bands.

## ➢ Single sideband modulation variants

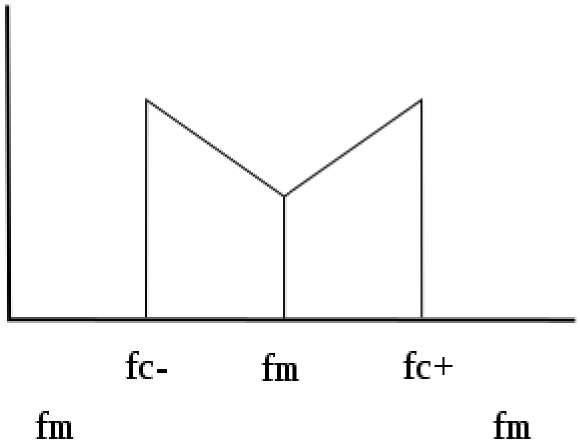
- There are many variants of single sideband modulation that are used, and there are several different abbreviations for them. These are explained below.

  - ***LSB:*** This stands for Lower Sideband. This form of single sideband modulation is formed when the lower sideband only of the original signal is transmitted. Typically, this is used by radio amateurs or radio hams on their allocations below 9 MHz.
  - ***USB:*** This stands for Upper Sideband. This form of single sideband modulation is formed when the upper sideband only of the original signal is transmitted. Typically, this form of SSB modulation is used by professional users on all frequencies and by radio amateurs or radio hams on their allocations above 9 MHz.
  - ***DSB:*** This is Double Sideband and it is a form of modulation where an AM signal is taken and the carrier is removed to leave the two sidebands. Although easy to generate, it does not give any improvements in spectrum efficiency and it is also not particularly easy to resolve. Accordingly, it is rarely used.
  - ***SSB SC:*** This stands for Single Sideband Suppressed Carrier. It is the form of SSB modulation where the carrier is removed completely as opposed to SSB reduced carrier where some of the carrier is left.

## Explain DSBSC signal

- **Double-sideband suppressed-carrier transmission** (DSB-SC) is transmission in which frequencies produced by amplitude modulation (AM) are symmetrically spaced above and below the carrier frequency and the carrier level is reduced to the lowest practical level, ideally being completely suppressed.
- DSB-SC is basically an amplitude modulation wave without the carrier, therefore reducing power waste, giving it a 50% efficiency.
- This is an increase compared to normal AM transmission (DSB) that has a maximum efficiency of 33.333%, since 2/3 of the power is in the carrier which conveys no useful

information and both sidebands containing identical copies of the same information. Single Side Band Suppressed Carrier (SSB-SC) is 100% efficient.




**Spectrum plot of a DSB-SC signal:**

- DSB-SC is generated by a mixer. This consists of a message signal multiplied by a carrier signal.

## 2.5 Methods of generating & detection SSB-SC signal (Indirect method only):

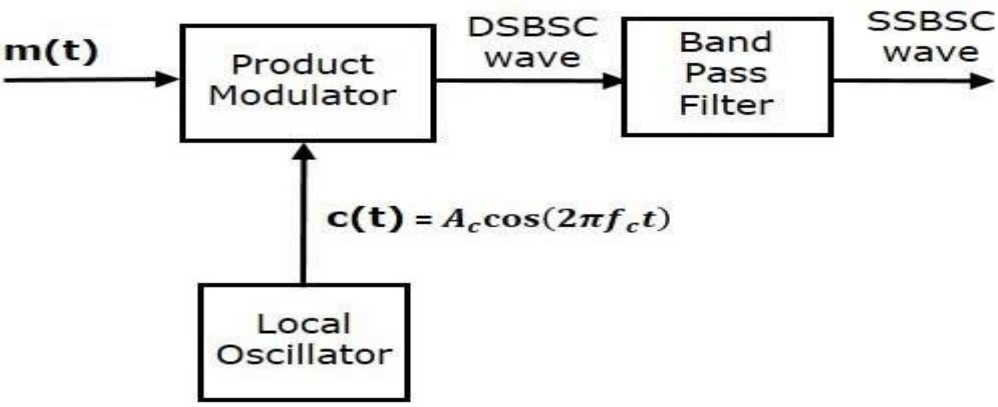
### Methods of generating & detection SSB-SC signal

➢ In this chapter, let us discuss about the modulators, which generate SSBSC wave. We can generate SSBSC wave using the following two methods.

- Frequency discrimination method (Filter Method)
- Phase discrimination method ( Phase shift Method)

### ➢ Frequency Discrimination Method

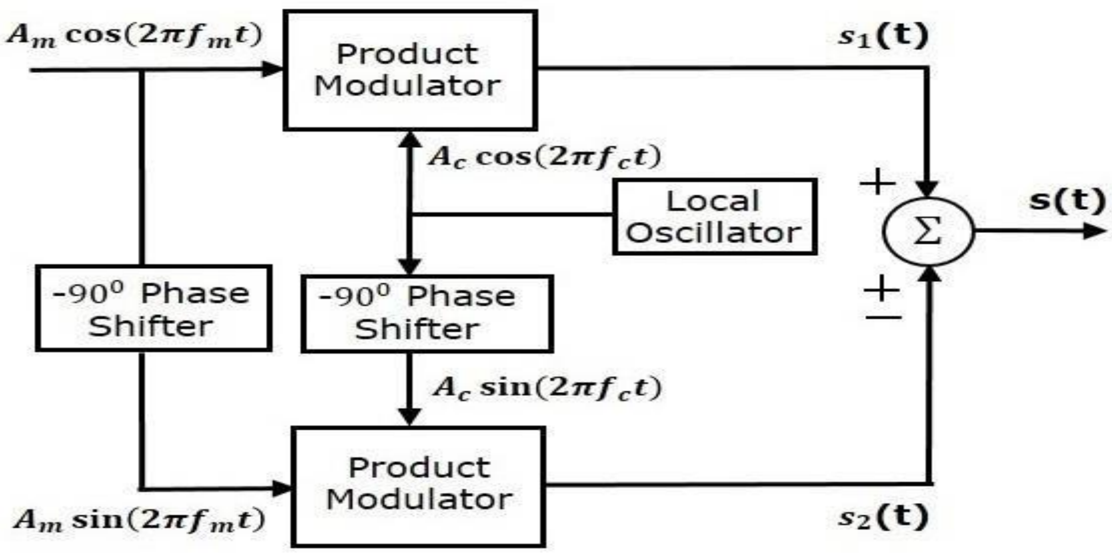
- The following figure shows the block diagram of SSBSC modulator using frequency discrimination method.




➢ In this method, first we will generate DSBSC wave with the help of the product modulator. The output of multiplier is m(t). $\cos w_c t$ which is DSB-SC signal. $\frac{A_m}{2}\cos[2\pi(f_c+f_m)t]+\frac{A_m}{2}\cos[2\pi(f_c-f_m)t]$ .Then, apply this DSBSC wave as an input of band pass filter. This band pass filter has sharp cut off frequency & produces an output, which is SSBSC wave.

- Select the frequency range of band pass filter as the spectrum of the desired SSBSC wave. This means the band pass filter can be tuned to either upper sideband or lower sideband frequencies to get the respective SSBSC wave having upper sideband or lower sideband. Upper side band has a frequency range $(f_c+f_m)$ & Lower side band has a frequency range $(f_c-f_m)$.
- Disadvantages- Design of ideal BPF is not achieved fully.
- **Phase Discrimination Method**

- The following figure shows the block diagram of SSBSC modulator using phase discrimination method.




- This block diagram consists of two product modulators, two $-90^0$ phase shifters, one local oscillator and one summer block. The product modulator produces an output, which is the product of two inputs. The $-90^0$ phase shifter produces an output, which has a phase lag of $-90^0$ with respect to the input.
- The local oscillator is used to generate the carrier signal. Summer block produces an output, which is either the sum of two inputs or the difference of two inputs based on the polarity of inputs.
- The modulating signal $A_m\cos(2\pi f_m t)$ and the carrier signal $A_c\cos(2\pi f_c t)$ are directly applied as inputs to the upper product modulator. So, the upper product modulator produces an output, which is the product of these two inputs.
- The output of upper product modulator is

$$s_1(t)=A_mA_c\cos(2\pi f_m t)\cos(2\pi f_c t)$$

- & Output of lower product modulator is

$$s_1(t)=A_mA_c \,\mathrm{Sin}\,(2\pi f_m t)\ \mathrm{Sin}(2\pi f_c t)$$

## 2.6 Methods of generation DSB-SC signal (Ring Modulator) and detection of DSB-SC signal:

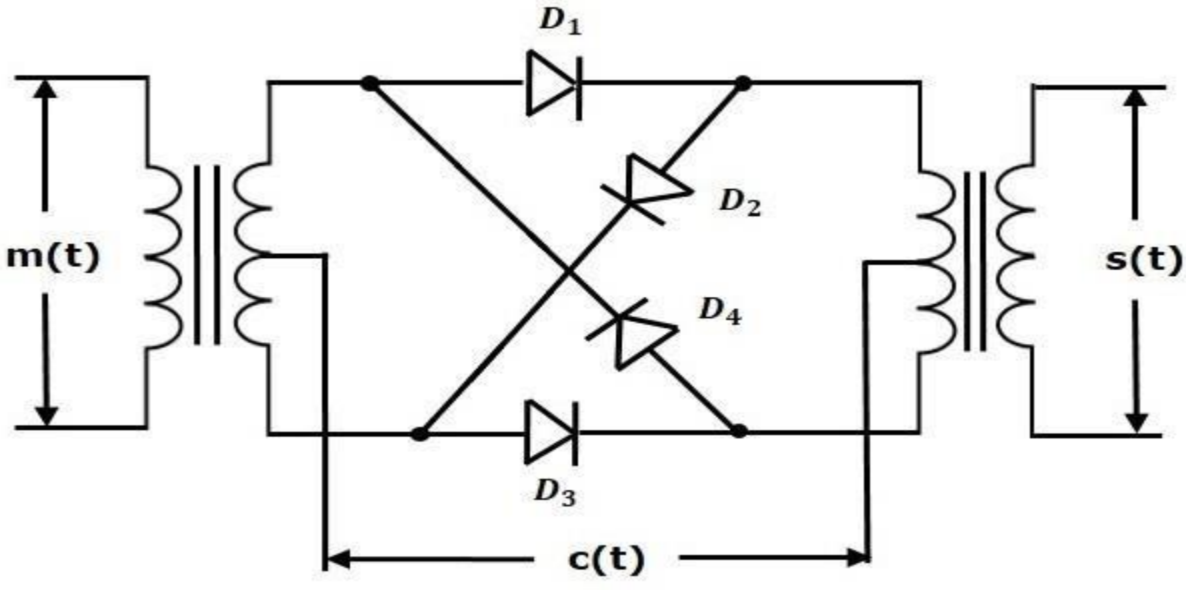
### Methods of generation DSB-SC signal

- The following two modulators generate DSBSC wave.

- Ring modulator (Chopper Modulator)
- Balanced modulator (Synchronous detection)

## Ring Modulator

- Following is the block diagram of the Ring modulator.



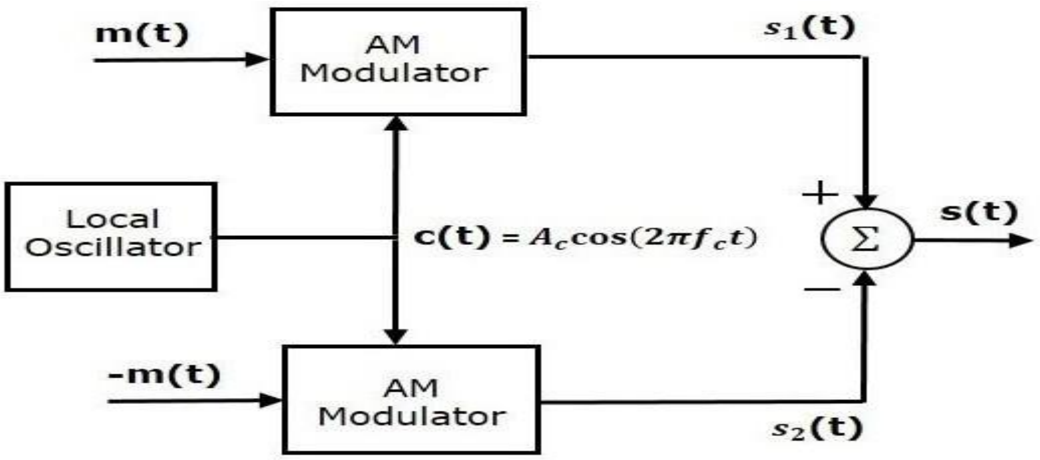

- In this diagram, the four diodes $D_1$, $D_2$, $D_3$ and $D_4$ are connected in the ring structure. Hence, this modulator is called as the **ring modulator**. Two center tapped transformers are used in this diagram. The message signal $m(t)$ is applied to the input transformer. Whereas, the carrier signals $c(t)$ is applied between the two canter tapped transformers.
- For positive half cycle of the carrier signal, the diodes $D_1$ and $D_3$ are switched ON and the other two diodes $D_2$ and $D_4$ are switched OFF. In this case, the message signal is multiplied by +1.
- For negative half cycle of the carrier signal, the diodes $D_2$ and $D_4$ are switched ON and the other two diodes $D_1$ and $D_3$ are switched OFF. In this case, the message signal is multiplied by -1. This results in $180^0$ phase shift in the resulting DSBSC wave.
- From the above analysis, we can say that the four diodes $D_1$, $D_2$, $D_3$ and $D_4$ are controlled by the carrier signal. If the carrier is a square wave, then the Fourier series representation of $c(t)$ is represented as

$$c(t)=4/\pi\sum_{n=1}^{\infty}(-1)^{n-1}/(2n-1)\cos[2\pi f_c t(2n-1)]\ x(t)$$

## 2.7 Concept of Balanced modulators:

### Balanced Modulator

- Following is the block diagram of the Balanced modulator.




- **Balanced modulator** consists of two identical AM modulators. These two modulators are arranged in a balanced configuration in order to suppress the carrier signal. Hence, it is called as Balanced modulator.
- **In this case 2 nonlinear devices(2 diodes) are connected in the balanced mode, so as to supress the carrier signal. So output will be AM-SC or DSB-SC.**
- **For diode input & output characteristic is in the form of $i=av+bv^2$**
  **$= aE_1+bE_1 2$**
- **Applying KVL, $E_1$=Cos Wc(t) +m(t) & $E_2$= CosWc(t) -m(t)**
- **Using KVL , $V_0(t) = V_1 - V_2 = (I_1 - I_2)R$**

**Solving the equation V0(t)= 2Ra Em CosWm(t) +2RbEmCos ( Wc+Wm) t +2RbEm Cos (Wc-Wm) t**

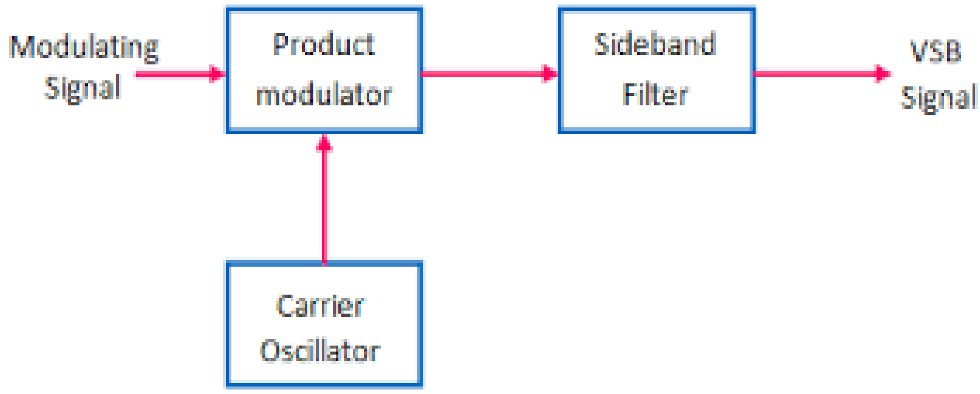
- Signal $W_m$ (t) is very low frequency & not passed through filter. So, other two signals (Both Side Bands) are within BPF range & are available at output side. So final output will be **2RbEm Cos ( Wc+Wm) t +2RbEm Cos ( Wc-Wm) t = KEm/2 (Cos ( Wc+Wm) t +Cos ( Wc-Wm) t** where K is extra as a constant.

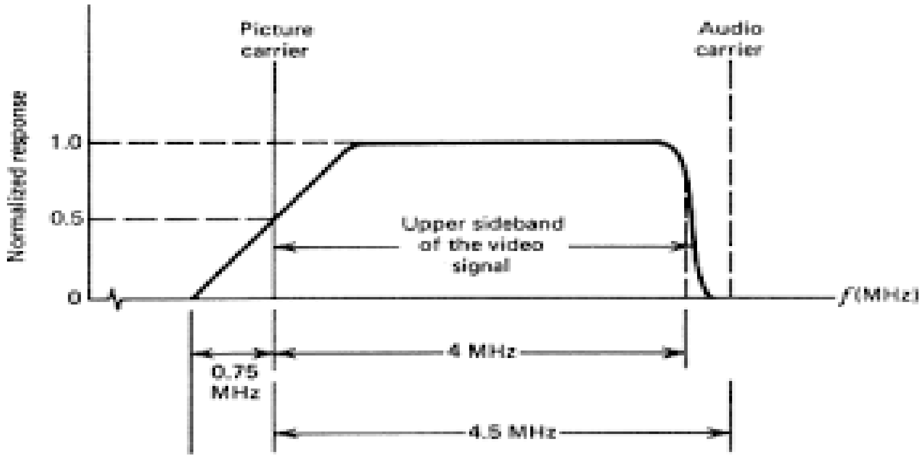
## 2.8 Vestigial Side Band Modulation:

AM vestigial Side Band (VSB) is a form of amplitude modulation in which the carrier & one complete side band are transmitted along with a portion of second side band**.In** VSB lower modulating signal frequencies are transmitted double side band, & higher modulating signal frequencies are transmitted single side band. So lower frequency modulating signal are emphasized & produce larger amplitude signal in the demodulation than the high frequency.

It is generally used in TV transmission, because the video signal needs a large transmission bandwidth if transmitted using DSB-FC or DSB-SC techniques.

The block diagram of a VSB modulator is shown in the figure.




The modulating signal x(t) is applied to a product modulator. The O/P of the carrier oscillator is also applied to the other input of the product modulator.

So $m(t)=x(t) .c(t)= x(t).V_c \cos w_c(t)$ This represents a DSB-SC modulated wave whicg is applied to a sideband shaping filter. This filter will pass the wanted side band & vestige of the unwanted side band.

Hence the spectrum of the VSB modulated signal given by $S(f)= V_c/2 ( x(f-fc) +x(f+fc) )$

## Possible Short type Questions with answers

### 1. Define Balanced modulator.

ANS: **Balanced modulator** consists of two identical AM modulators. These two modulators are arranged in a balanced configuration in order to suppress the carrier signal. Hence, it is called as Balanced modulator.

### 2. Define Amplitude modulation.

ANS: **Amplitude modulation**

In amplitude modulation, the amplitude (signal strength) of the carrier wave is varied in proportion to that of the message signal

### 3. Define Modulation Index of AM. [W-20]

ANS: **Modulation Index**

- Modulation index defines the extent upto which amplitude of the carrier will be varied about an unmodulated maximum carrier.

**Mathematically, $m = A_M / A_C$ Where Am is the maximum amplitude of Base Band signal & Ac is the maximum amplitude of carrier signal.**

4. **Define *LSB.***

ANS: ***LSB:*** This stands for Lower Sideband. This form of single sideband modulation is formed when the lower sideband only of the original signal is transmitted. Here specified BPF is used .Typically, this is used by radio amateurs or radio hams on their allocations below 9 MHz.

5. **Define *USB*.**

ANS: ***USB:*** This stands for Upper Sideband. This form of single sideband modulation is formed when the upper sideband only of the original signal is transmitted. Typically, this form of SSB modulation is used by professional users on all frequencies and by radio amateurs or radio hams on their allocations above 9 MHz.

6. **Write down the application of VSB**.

Ans. Vestigial Side band is used in TV transmission. Here Both Audio & Video signal are transmitted which requires more bandwidth. So VSB is more economical compare to DSB-SC or DSB-FC.

## Possible Long type questions:

**1.Explain Amplitude modulation & derive the expression for amplitude modulation signal with proper block diagram. [W-20]**

**2.Explain Methods of generation DSB-SC signal (Ring Modulator) with Block diagram.**

**3.Explain DSBSC signal with Block diagram.**

**4.Explain Vestigial Side Band.**

**5.Derive the expression for power in AM wave.**

**6.With its block diagram Explain the balance modulator using DSB-SC generation. [W-20].**

# UNIT 3.0

# ANGLE MODULATION SYSTEMS

**LEARNING OBJECTIVES:**

*3.1 Concept of Angle modulation & its types (PM & FM)*
*3.2 Basic principle of Frequency Modulation & Frequency Spectrum of FM Signal*
*3.3 Expression for Frequency Modulated Signal & Modulation Index and sideband of FM signal*
*3.4 Explain Phase modulation & difference of FM & PM- working principle with Block Diagram*
*3.5 Compare between AM and FM modulation (Advantages & Disadvantages)*
*3.6 Methods of FM Generation (Indirect (Armstrong) method only) working principle with Block Diagram*
*3.7 Methods of FM Demodulator or detector (Forster-Seely & Ratio detector)- working principle with Block Diagram*

## 3.1 Concept of Angle modulation & its types (PM & FM):

- **Angle Modulation** is the process changing the phase angle means both phase & frequency in accordance with instantaneous value of modulation signal keeping amplitude constant is known as angle modulation.
- It has two types namely
    1) Frequency modulation and
    2) Phase modulation.

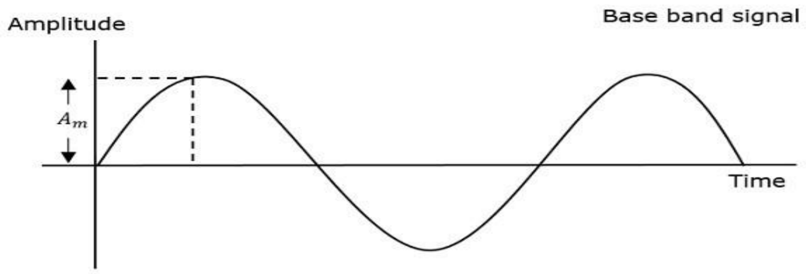
- **Frequency Modulation** is the process of varying the frequency of the carrier signal linearly with the intensity of the message signal.

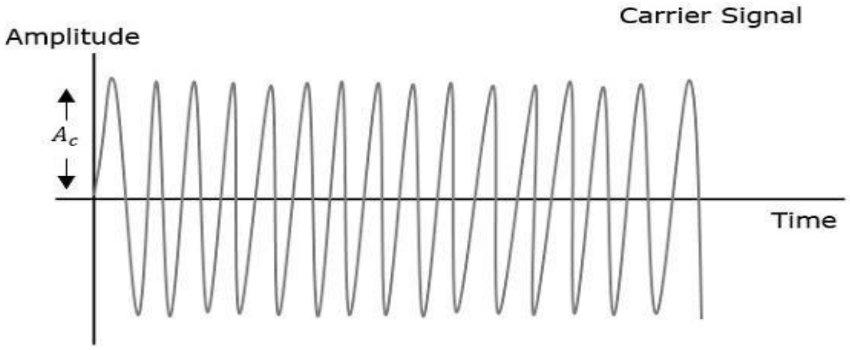
- **Phase Modulation** is the process of varying the phase of the carrier signal linearly with the intensity of the message signal.

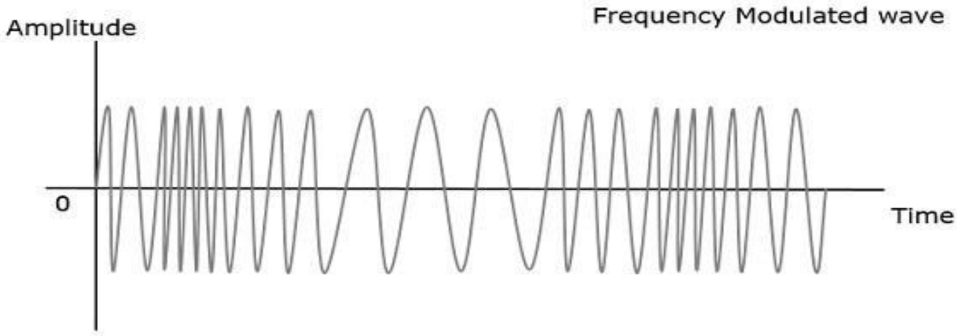
## 3.2 Basic principle of Frequency Modulation & Frequency Spectrum of FM Signal:

### ➢ Frequency Modulation

- In Frequency Modulation (FM), the frequency of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.
- The amplitude and the phase of the carrier signal remains constant whereas the frequency of the carrier changes. This can be better understood by observing the following figures.



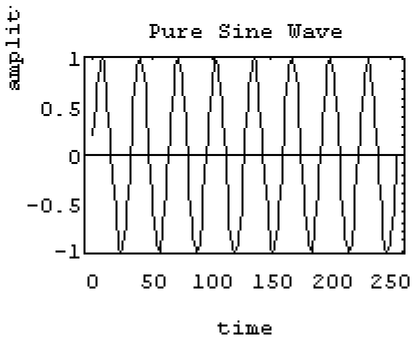



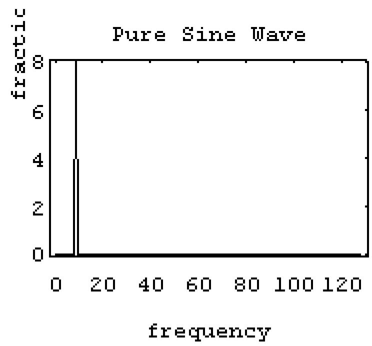




- When the message signal is at zero, the frequency of the modulated wave remains constant as the carrier wave frequency. The frequency increases when the message signal amplitude increases & frequency decreases when amplitude of base band signal decreases.
- Which means, with the increase in amplitude of the modulating or message signal, the carrier frequency goes on increasing. Likewise, with the decrease in the amplitude of the modulating signal, the frequency goes on decreasing.

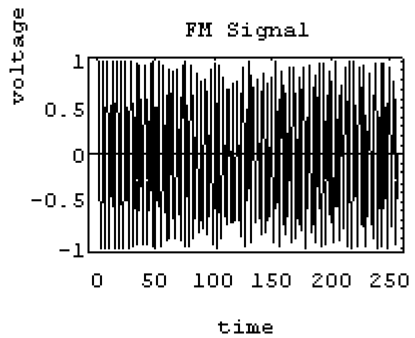
## ➢ FM Spectrum of FM Signal

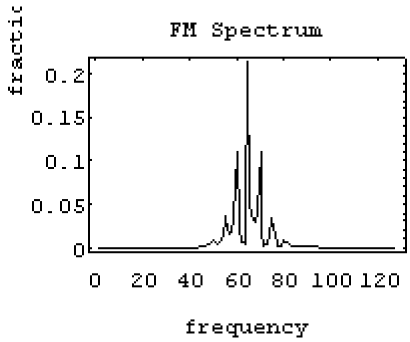
- A spectrum represents the relative amounts of different frequency components in any signal. It is a well-known fact of mathematics, that any function (signal) can be decomposed into purely sinusoidal components.
- In technical terms, the sines and cosines wave form a complete set of functions, also known as a basis in the infinite-dimensional vector space of real-valued functions.
- Given that any signal can be thought to be made up of sinusoidal signals, the spectrum then represents the "recipe card" of how to make the signal from sinusoids. Like: 1 part of 50 Hz and 2 parts of 200 Hz. Pure sinusoids have the simplest spectrum of all, just one component:




- In this example, the carrier has 8 Hz and so the spectrum has a single component with value 1.0 at 8 Hz
- The FM spectrum is considerably more complicated. The spectrum of a simple FM signal looks like as under




## 3.3 Expression for Frequency Modulated Signal & Modulation Index and sideband of FM signal:

- **Expression for Frequency Modulated Signal**

In this case the frequency of carrier signal is varied in accordance with the instantaneous amplitude of the modulating signal

In FM, the amplitude of carrier remains constant. The variation in carrier frequency from the unmodulated carrier is called frequency deviation ***(δ)***.

- FM is non-linear process & information contain in frequency deviation.

**Mathematical Expression of FM:**

- Let instantaneous frequency of frequency modulated wave is;

***Wi=Wc+ δ where Wc is the carrier frequency & δ is the deviation***

- The frequency deviation above or below ***Wc*** depends on the instantaneous amplitude of modulating signal i.e ***δ=kVm Cos Wm (t)***

where ***k=*** frequency deviation sensitivity of FM

***So, Wi= Wc + δ = Wc+ kVm cos Wm (t)*** --------------------- (i)

- The frequency modulated signal is represented mathematically as

$V_{fm\,(t)}$*=Vc sin θ* ..............------...............(ii)

If θ instantaneous phase then it can be determined from instantaneous frequency as

***Wi = dθ /dt***

***Wi dt = dθ***

$\int$***dθ=*** $\int$***Wi dt = Θ***

Substituting the value of Wi from equation (i) in above equation;

***θ=*** $\int$ ***{Wc+kVm cos Wm(t)}dt***

***θ= Wc (kVm/Wm) sin Wm (t)***

***θ= Wc+βsin Wm (t)***.......................................................(iii)

where ***β*** (modulation Index) = ***(kVm/Wm)***

Substituting the value of θ from equation (iii) in equation (ii);

$V_{fm\,(t)}$ ***= Vc sin (Wc(t)+ β sin Wm (t)***

Above is the equation of frequency modulation.

## ➢ Frequency modulation index

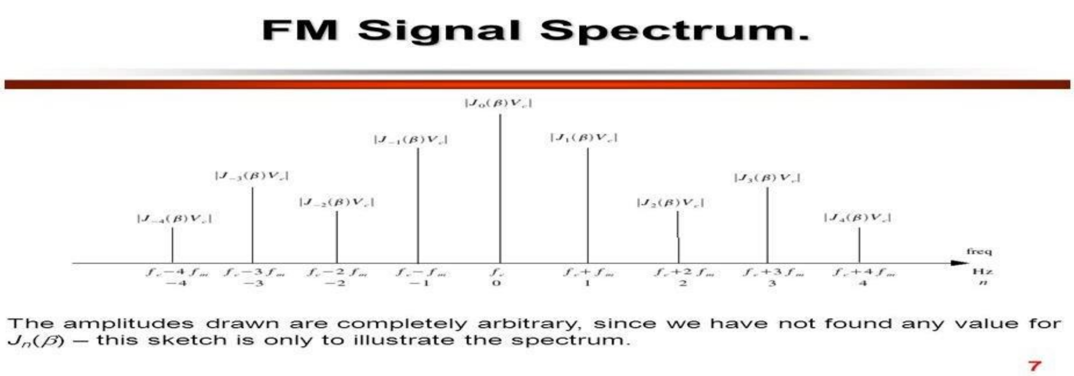
- In FM, modulation index is equal to the ratio of the frequency deviation to the modulating frequency.

**β=frequency deviation/modulation frequency=** $\Delta f/f_m$ **=*(kVm/Wm)***

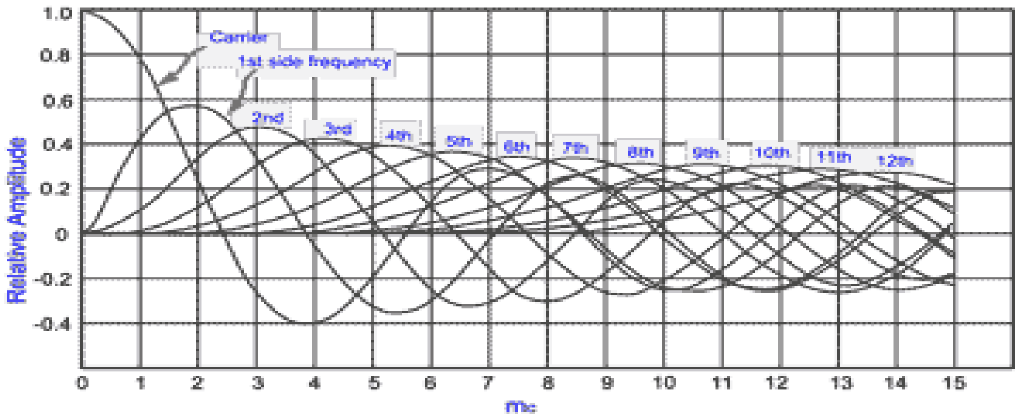
- From the formula and definition of the modulation index, it can be seen that there is no term that includes the carrier frequency and this means that it is totally independent of the carrier frequency.

## ➢ Frequency modulation sidebands

- The modulation of any carrier in any way produces sidebands. For amplitude modulated signals, the way in which these sidebands are created and their bandwidth and amplitude are quite straightforward. The situation for frequency modulated signals is rather different.
- The FM sidebands are dependent on both the level of deviation and the frequency of the modulation. In fact, the total spectrum for a frequency modulated signal consists of the carrier plus an infinite number of sidebands spreading out on either side of the carrier at integral multiples of the modulating frequency.



$$x_m(t) \triangleq e^{j\beta \sin(\omega_m t)} = \sum_{k=-\infty}^{\infty} J_k(\beta) e^{jk\omega_m t}.$$



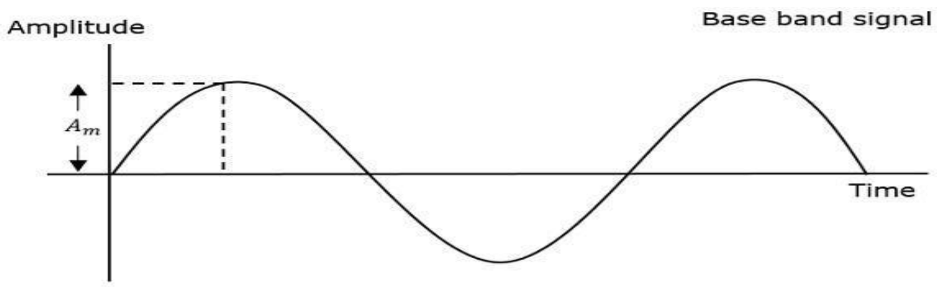

**Frequency modulation sideband levels**

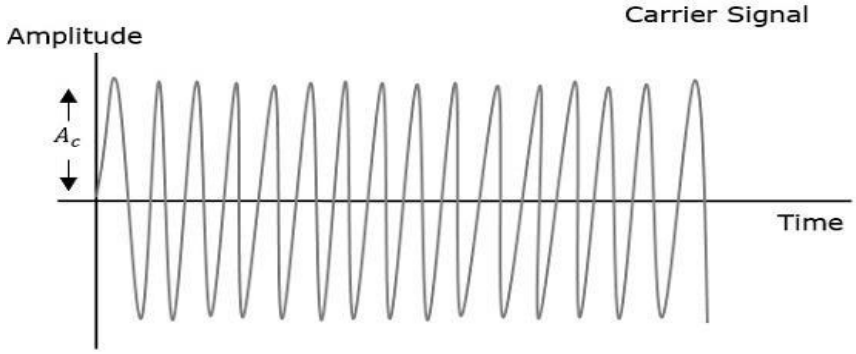
- From the Graph & Bessel's equation it is seen that frequency spectrum extends up to infinite but amplitude goes on decreasing. At infinite the amplitude will be zero. So depending upon MI(Modulation Index) value FM is two types i.e. NBFM & WBFM . For WBFM(Wide band FM)Modulation Index is large, then large no of side bands are produced, hence bandwidth is large. $\beta \gg 1$,then BW= 2 (fm+ δ)= 2 fm(1+ δ/ fm) =2 fm(1+ β )
- So BW =2 fm β
- But in NBFM (Narrow Band FM), $\beta \ll 1$ so Band width will be 2fm just like AM. So, channel BW of FM depends upon Modulation Index of FM.

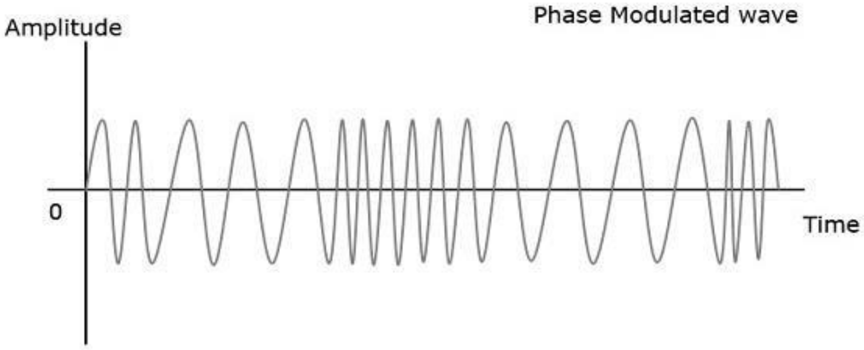
## 3.4 Explain Phase modulation & difference of FM & PM)-working principle with Block Diagram:

### ➢ Phase Modulation

- In frequency modulation, the frequency of the carrier varies. But in **Phase Modulation (PM)**, the phase of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.
- The amplitude and the frequency of the carrier signal remains constant whereas the phase of the carrier changes. This can be better understood by observing the following figures.




- The phase of the modulated wave has got infinite points where the phase shift in a wave can take place. The instantaneous amplitude of the modulating signal, changes the phase of the carrier. When the amplitude is positive, the phase changes in one direction and if the amplitude is negative, the phase changes in the opposite direction.

## ➢ Difference between Frequency Modulation and Phase Modulation:

| **Frequency Modulation** | **Phase Modulation** |
|---|---|
| 1.In Frequency Modulation amplitude and phase remain the same but frequency changes. | In Phase Modulation, the frequency and amplitude remain the same but phase changes. |
| 2. Frequency Modulation is proportional to modulating voltage. | Phase Modulation is proportional to modulating voltage. |
| 3. It is possible to receive FM on a PM receiver. | It is possible to receive PM on a FM receiver. |

| | | | |
|---|---|---|---|
| 4. | Noise immunity is poor than AM and PM | Noise immunity is better than AM but worse than PM. |
| 5. | Signal to noise ratio is better than in phase modulation. | Signal to noise ratio is poor than in frequency modulation |

.

## 3.5 Compare between AM and FM modulation (Advantages & Disadvantages):

**Difference Between AM and FM**

| S.No | Parameters | AM | FM |
|---|---|---|---|
| 1. | Full form | Amplitude modulation | Frequency modulation |
| 2. | Origin | AM method of audio transmission was successfully carried out in the mid-1870s. | FM radio was developed in the United States in the 1930s by Edwin Armstrong. |
| 3. | Modulating differences | In AM, a radio wave is known as the "carrier" or "carrier wave" is modulated in amplitude by the signal that is to be transmitted. | In FM, a radio wave is known as the "carrier" or "carrier wave" is modulated in frequency by the signal that is to be transmitted. |
| 4. | Constant parameters | The frequency and phase remain the same. | The amplitude and phase remain the same. |
| 5. | Quality | AM has poor sound quality, and a lower bandwidth but is cheaper and can be transmitted over long distances | FM is less affected by interference, but FM signals are impacted by physical barriers. They have a better sound quality due to higher bandwidth. |
| 6 | Band width | 2fm | More |
| 7 | Efficiency | More | Less |

**Advantages of AM (Amplitude modulation)**:

➡AM signal are reflected back to earth from ionosphere layer. Due to this fact, AM signals can reach far places which are thousands of miles from source. Hence AM radio has coverage wider compare to FM radio.

**Disadvantages of AM:**

➡The most natural as well as manmade radio noise are of AM type. The AM receivers do not have any means to reject this kind of noise.

➡Weak AM signals have low magnitude compare to strong signals. This requires AM receiver to have circuitry to compensate for signal level difference.

### Advantages and disadvantages of FM.

### Advantages

- The noise can be reduced by increasing the deviation that's why FM is more immune to noise.
- Due to higher efficiency most, the power is utilized and the FM transmissions can be used for the stereo sound transmission due to a large number of sidebands.

### Disadvantages

- The bandwidth of FM is 10 times larger than the AM. Hence a wide channel would be required for FM transmission.
- The transmitting and receiving equipment are very complicated in FM.

## 3.6 Methods of FM Generation (Indirect (Armstrong) method only) working principle with Block Diagram

- **Armstrong Method for the Generation of FM**

- In the direct methods of generation of FM, LC oscillators are to be used. The crystal oscillator cannot be used. The LC oscillators are not stable enough for the communication or broadcast purpose. Thus, the direct methods cannot be used for the broadcast applications.
- The alternative method is to use the indirect method called as the Armstrong method of FM generation.
- In this method, the FM is obtained through phase modulation. A crystal oscillator can be used hence the frequency stability is very high and this method is widely used in practice.

- Figure.1 shows the block diagram of the Armstrong method.

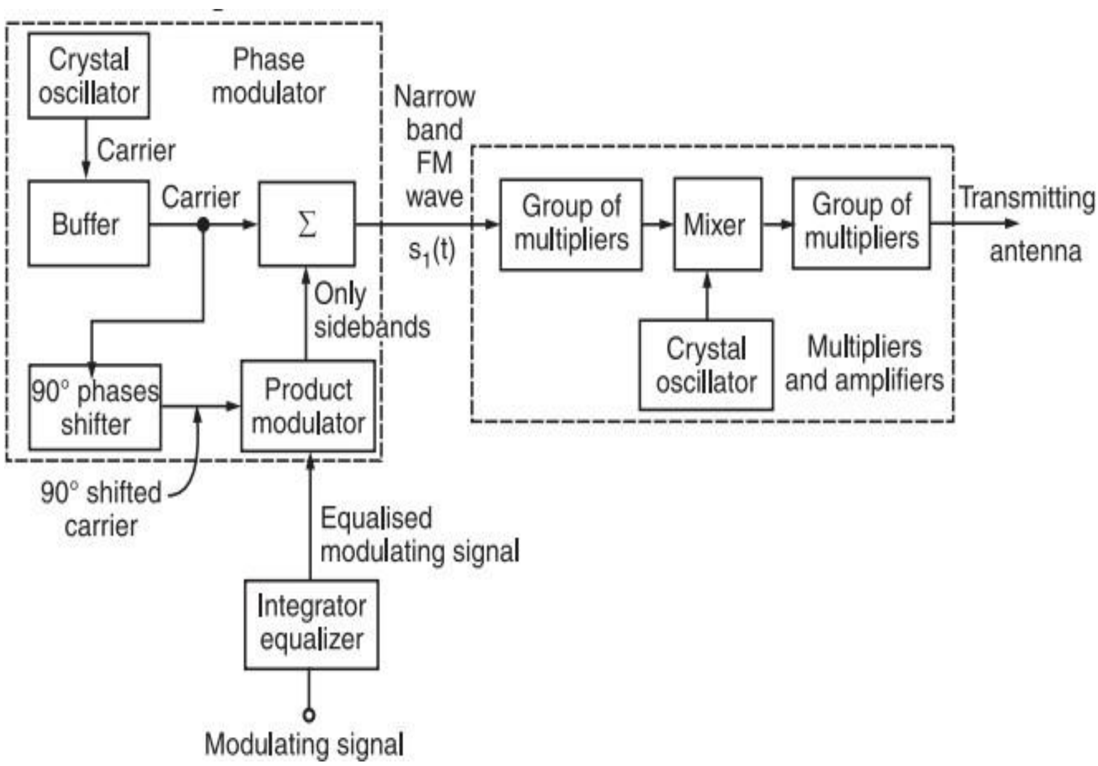



**Fig.1 : Armstrong Method for FM Generation**

**Working Principle**

- The crystal oscillator produces a stable unmodulated carrier which is applied to the 90° phase shifter as well as the combining network through a buffer.
- The 90° phase shifter produces a 90° phase shifted carrier. It is applied to the balanced modulator along with the modulating signal.
- Thus, the carrier used for modulation is 90° shifted with respect to the original carrier.
- At the output of the product modulator, we get DSB SC signal i.e., AM signal without carrier.
- This signal consists of only two sidebands with their resultant in phase with the 90° shifted carrier.
- The two sidebands and the original carrier without any phase shift are applied to a combining network (∑). At the output of the combining network, we get the resultant of vector addition of the carrier and two sidebands as shown in figure 2.

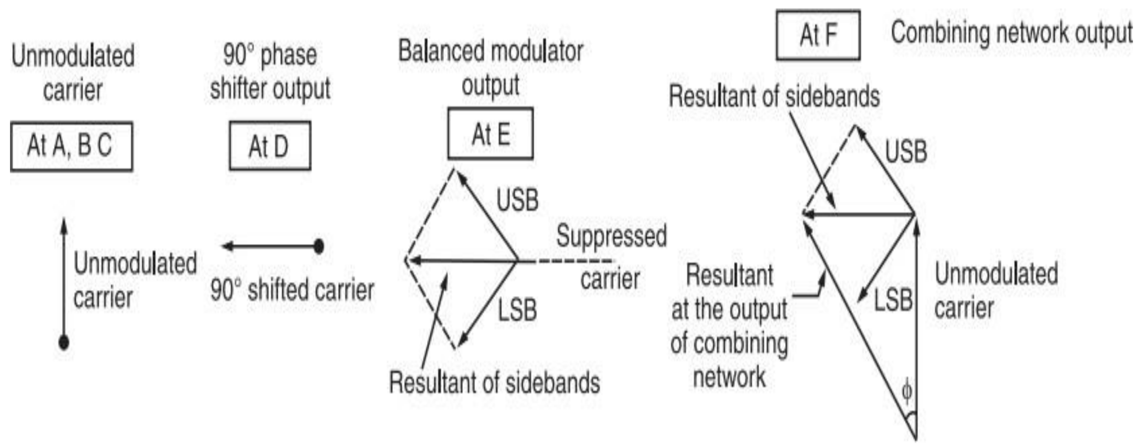



Fig.2: Phasors explaining the generation of PM

- Now, as the modulation index is increased, the amplitude of sidebands will also increase. Hence, the amplitude of their resultant increases. This will increase the angle Φ made by the resultant with unmodulated carrier.
- The angle Φ decreases with reduction in modulation index as shown in figure 3.

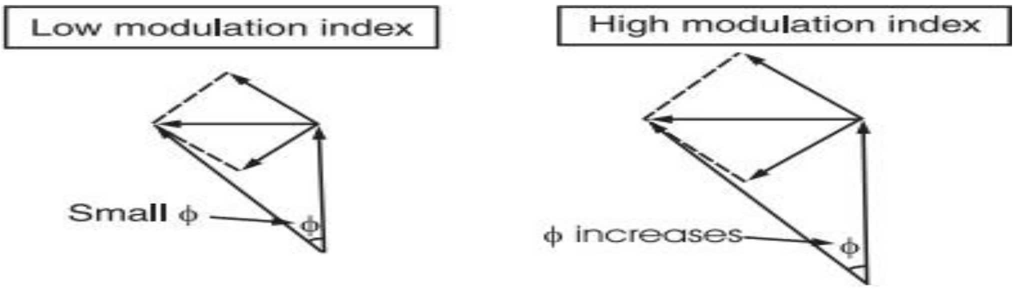



Fig.3 : Effect of modulation index on frequency f

- Thus, the resultant at the output of the combining network is phase modulated. Hence, the block diagram of figure.1 operates as a phase modulator.
- In this method first we have to generate NBFM. Here m(t) signal is passed through integrator & its output is ∫***m(t).***
  Then it fed to phase modulator with the help of crystal oscillator output Vc Cos wc(t), & the final output will be $V_{fm}{}^{(t)}$ = *Vc sin (Wc(t)+ β sin Wm (t)*
  This signal is called NBFM.
- NBFM signal is to be passed frequency multipliers to obtain WBFM. Its function is to multiply the carrier frequency as many times as frequency deviation.

## 3.7 Methods of FM Demodulator or detector (Forster-Seely & Ratio detector)- working principle with Block Diagram

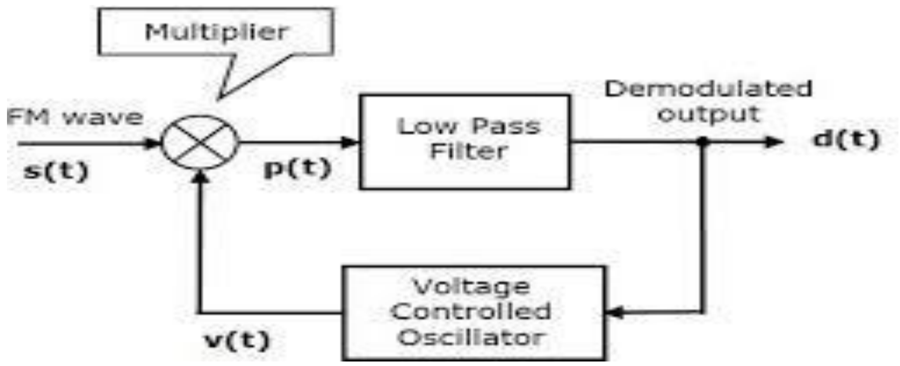
### ➢ Methods of FM DEMODULATORS

- The following two methods demodulate FM wave.

(1) Frequency discrimination method (Balanced FM Slope Detector (Balanced Frequency Discriminator)
(2) Phase discrimination method (Foster Seeley FM Demodulator)

### Phase discrimination method (Foster Seeley FM Demodulator)

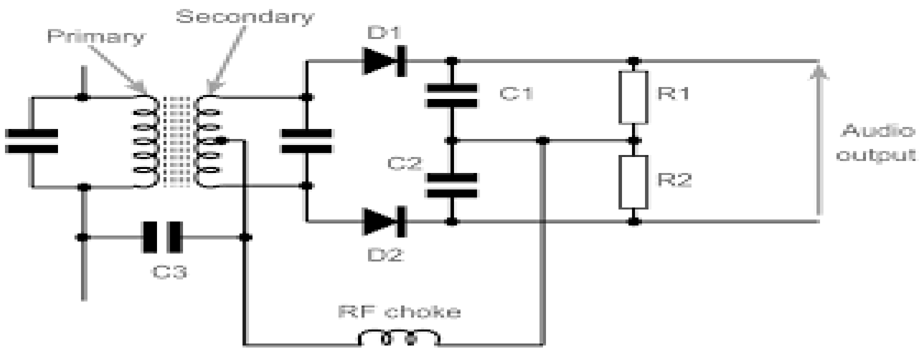
- The following figure shows the block diagram of FM demodulator using phase discrimination method.




**Foster Seeley FM Demodulator (Block diagram)**

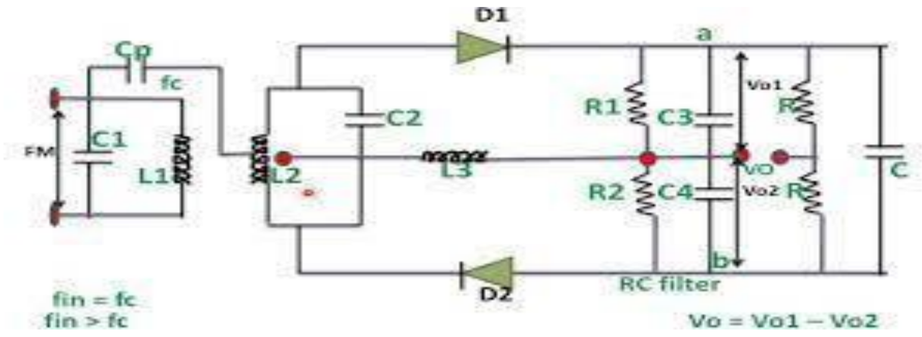
- This block diagram consists of the multiplier, the low pass filter, and the Voltage Controlled Oscillator (VCO).
- VCO produces an output signal v(t), whose frequency is proportional to the input signal voltage d(t). Initially, when the signal d(t) is zero, adjust the VCO to produce an output signal v(t), having a carrier frequency and $-90^0$ phase shift with respect to the carrier signal.
- FM wave s(t) and the VCO output v(t) are applied as inputs of the multiplier.
- The multiplier produces an output, having a high frequency component and a low frequency component.
- Low pass filter eliminates the high frequency component and produces only the low frequency component as its output.

## Foster-Seeley FM discriminator




- In many respects the Foster Seeley FM demodulator resembles the circuit of a full wave bridge rectifier - the format that uses a centre tapped transformer, but additional components are added to give it a frequency sensitive aspect.
- The basic operation of the circuit can be explained by looking at the instances when the instantaneous input equals the carrier frequency, the two halves of the tuned transformer circuit produce the same rectified voltage and the output is zero. If the frequency of the input changes, the balance between the two halves of the transformer secondary changes, and the result is a voltage proportional to the frequency deviation of the carrier.
- Looking in more detail at the circuit, the Foster-Seeley circuit operates using a phase difference between signals. To obtain the different phased signals a connection is made to the primary side of the transformer using a capacitor, and this is taken to the centre tap of the transformer. This gives a signal that is 90° out of phase.
- When an un-modulated carrier is applied at the centre frequency, both diodes conduct, to produce equal and opposite voltages across their respective load resistors. These voltages cancel each one another out at the output so that no voltage is present. As the carrier moves off to one side of the centre frequency the balance condition is destroyed, and one diode conducts more than the other. This results in the voltage across one of the resistors being larger than the other, and a resulting voltage at the output corresponding to the modulation on the incoming signal.
- The choke is required in the circuit to ensure that no RF signals appear at the output. The capacitors C1 and C2 provide a similar filtering function.
- Both the ratio detector and Foster-Seeley detectors are expensive to manufacture. As a result, the Foster Seeley discriminator as well as the ratio detector circuits are rarely used in modern radio receivers as FM demodulators.

## Ratio Detector in FM




**FM ratio discriminator / detector circuit**

- The operation of the ratio detector centres around a frequency sensitive phase shift network with a transformer and the diodes that are effectively in series with one another. When a steady carrier is applied to the circuit the diodes act to produce a steady voltage across the resistors R1 and R2, and the capacitor C3 charges up as a result.

- The transformer enables the circuit to detect changes in the frequency of the incoming signal. It has three windings. The primary and secondary act in the normal way to produce a signal at the output. The third winding is un-tuned and the coupling between the primary and the third winding is very tight, and this means that the phasing between signals in these two windings is the same.

- The primary and secondary windings are tuned and lightly coupled. This means that there is a phase difference of 90 degrees between the signals in these windings at the centre frequency. If the signal moves away from the centre frequency the phase difference will change. In turn the phase difference between the secondary and third windings also varies. When this occurs, the voltage will subtract from one side of the secondary and add to the other causing an imbalance across the resistors R1 and R2. As a result, this causes a current to flow in the third winding and the modulation to appear at the output.

- The capacitors C1 and C2 filter any remaining RF signal which may appear across the resistors. The capacitor C4 and R3 also act as filters ensuring no RF reaches the audio section of the receiver.

**Ratio detector advantages & disadvantages**

**Advantages of ratio detector**

- Simple to construct using discrete components
- Offers good level of performance and reasonable linearity

**Disadvantages of Ratio FM discriminator:**

- High cost of transformer ,so it is not used by now.
- Typically lends itself to use in only circuits using discrete components and not integrated within an IC

.**Possible short type questions with Answer**

1. **Define Frequency Modulation. (2005-S, 2014-S)**

   ANS: **Frequency Modulation** is the process of varying the frequency of the carrier signal linearly with the message signal.

2. **Define Angle Modulation**
   ANS **Angle Modulation** is the process in which the frequency or the phase of the carrier varies according to the message signal.

- This is further divided into
  i) Frequency modulation and
  ii) Phase modulation.

3. **Define Frequency deviation. (2012-S,2010 -S, 2005-S)**
   Ans-Frequency deviation is defined as the ratio of maximum allowable frequency deviation to the maximum modulating frequency for FM broadcasting.
4. **Define Modulation Index for frequency modulation (2018-S,2019-S)**
   Ans-It is defined as the measure of extend of frequency variation about an unmodulated maximum carrier .
   $\beta$=**frequency deviation/modulation frequency= $\Delta f/f_m$ =*(kVm/Wm)***

5. **Define Armstrong Method for the Generation of FM.**
   ANS: In the direct methods of generation of FM, LC oscillators are to be used. The crystal oscillator cannot be used. The LC oscillators are not stable enough for the communication or broadcast purpose. Thus, the direct methods cannot be used for the broadcast applications.

## Possible Long Type Questions

1. Derive the expression for frequency modulated signal .(2005-S,2021-S & 2018 -S)
2. Explain Phase discrimination method (Foster Seeley FM Demodulator) with Block diagram. {2007-S,2008-S,2010-S,2012-S,2014-S, 2019-S}
3. Explain Basic principle of Frequency Modulation with Block diagram.
4. What are the advantages of FM over AM [2007-S,2010-S]
5. Explain Difference between Frequency Modulation and AM. [2005-S,2006_s,2008-S,2014-S,2018-S,2019-S 2020-S]
6. Explain in detail the generation of FM using Armstrong method with a neat block diagram. [2005-S,2012-S,2013-S , 2019-S,2020-W]

# Unit-4
# AM & FM TRANSMITTER & RECEIVER

**LEARNING OBJECTIVES:**

*4.1 Classification of Radio Receivers*
*4.2 Define the terms Selectivity, Sensitivity, Fidelity and Noise Figure*
*4.3 AM transmitter - working principle with Block Diagram*
*4.4 Concept of Frequency conversion, RF amplifier & IF amplifier, Tuning, S/N ratio*
*4.5 Working of super heterodyne radio receiver with Block diagram*
*4.6 Working of FM Transmitter & Receiver with Block Diagram*

## 4.1 Classification of Radio Receivers:

**Classification of Radio receivers: -**

- There are number of different types of radio receivers:
  - **a) Tuned radio frequency Receiver**
  - **b) Regenerative receiver**
  - **c) Super regenerative receiver**
  - **d) Super heterodyne receiver**
  - **e) Direct conversion receiver**

**1. Tuned radio frequency Receiver, TRF:**

• The very first radio receivers of this type simply consisted of a tuned circuit and a detector. Crystal sets were early forms of TRF radios.

• Later amplifiers were added to boost the signal level, both at the radio frequencies and audio frequencies. The main problems with this form of receiver is the lack of selectivity.

**2. Regenerative receiver:**

•The regenerative radio receiver significantly improved the levels of gain and selectivity obtainable.

•It uses positive feedback and ran at the point just before oscillation occurred. Major improvements in gain were obtained.

**3. Super regenerative receiver:**

•The super regenerative radio receiver takes the concept of further regeneration stage. Using a second lower frequency oscillation within the same stage, this second oscillation interrupts the oscillation of the main regeneration.

•In this way the main regeneration can be run so that the stage is effectively in oscillation where it provides very much higher levels of gain.

**4. Super heterodyne receiver:**

•The super heterodyne form of radio receiver was developed to provide additional levels of selectivity.

•It uses the heterodyne or mixing process to convert signals done to a fixed intermediate frequency. Changing the frequency of the local oscillator effectively tunes the radio.

5. **Direct conversion receiver:**

•This type of radio format converts the signal directly down to the baseband frequency.

•Initially it was used for AM and SSB transmissions, but now it is widely used for digital communications where demodulators are used to take advantage of the variety of phase shift keying (PSK), and quadrature amplitude modulation (QAM) signals.

## 4.2 Define the terms Selectivity, Sensitivity, Fidelity and Noise Figure:

- **Selectivity: -**

•Selectivity is one of the key attributes of any radio receiver. It needs to accept signals on the wanted frequency only and rejects signals of others frequencies. Here signals have narrow bandwidth.

•Selectivity is usually measured as a ratio in decibels (dB).

- **Sensitivity: -**

•The ability of a receiver to identify and amplify signals at the receiver's input is called Receiver Sensitivity. It is expressed in dBm.

•The receiver sensitivity level tells us the weakest signal that a receiver will be able to identify and process.

- **Fidelity: -**

•Fidelity of a receiver is its ability to reproduce the exact replica of the transmitted signals at the receiver output.

•For better fidelity, the amplifier must pass high bandwidth signals to amplify the frequencies of the outermost sidebands, while for better selectivity the signal should have narrow bandwidth.

- **Noise Figure: -**

•Noise figure is a number by which the noise performance of an amplifier or a radio receiver can be specified. NF is a measure of the degradation in signal to noise ratio & it can be used in association with radio receiver sensitivity & it is essential element of RF circuit design of any radio receiver.

The Noise Factor can be derived simply taking the SNR at the input & dividing it by the SNR at the output.

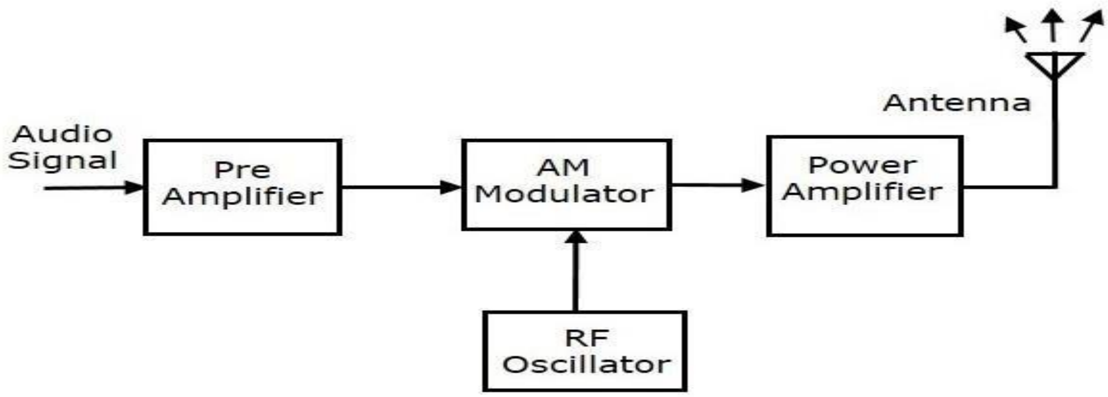
•The lower the value of the noise figure, the better is the performance.

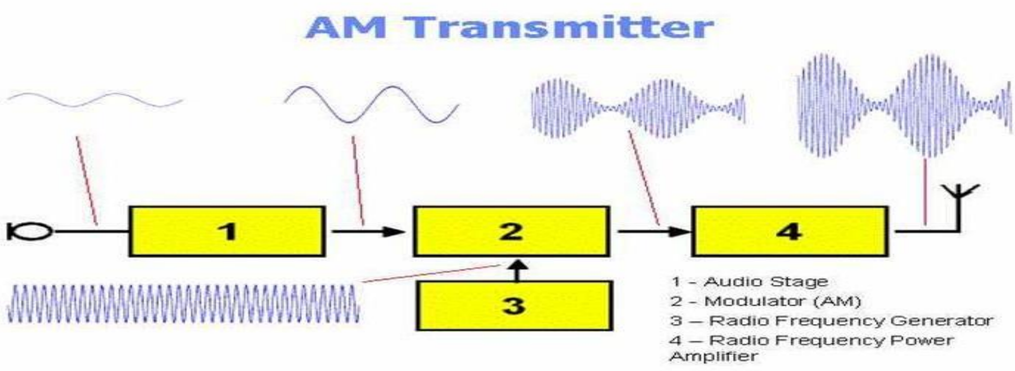
## 4.3 AM transmitter - working principle with Block Diagram:

- **AM Transmitter: -**

•AM transmitter takes the audio signal as an input and delivers amplitude modulated wave to the antenna as an output to be transmitted.

•The block diagram of AM transmitter is shown in the following figure.




- The working of AM transmitter can be explained as follows.

i. **Audio signal: -** The audio signal from the output of the microphone is sent to the pre-amplifier.

i. **Pre-amplifier: -** Pre-amplifier boosts the level of the modulating signal.

ii. **RF oscillator: -** The RF oscillator generates the carrier signal.

iii. **AM modulator: -** Both the modulating and the carrier signal is sent to AM modulator.

iv. **Power amplifier: -**Power amplifier the final stage of the block diagram. It makes the signal stronger so that it can be transmitted into the aerial.

It is used to increase the power levels of AM wave. This wave is finally passed to the antenna to be transmitted.

## 4.4 Concept of Frequency Conversion, RF amplifier & IF amplifier, Tuning, S/N ratio:

### Frequency conversion: -

- Frequency conversion may refer to different processes affecting frequency of physical phenomena:

•The conversion of modulated signal to base band signal is done in a single frequency conversion. This avoids the complexity of the super heterodyne two or more frequency conversion, IF stages or Image Rejection issues. The received radio frequency signal is fed directly into a frequency mixer. Frequency of the local oscillator is not offset from but identical to the received signal's frequency.

•A Heterodyne is used in signal electronics to convert frequencies.

### RF amplifier: -

•A radio frequency power amplifier is a type of electronic amplifier that converts a low-power radio-frequency signal into a higher power signal.

•Typically, RF power amplifiers drive the antenna of a transmitter.

### IF amplifier: -

Intermediate-frequency (IF) amplifiers are amplifier stages used to raise signal levels in radio and television receivers, at frequencies intermediate to the higher radio-frequency (RF) signal from the antenna and the lower (baseband) audio or video frequency that the receiver is recovering.

### Tuning: -

A tuner is a subsystem that receives radio frequency (RF) transmissions and converts. The verb tuning in radio contexts means adjusting the receiver to detect the desired radio signal carrier frequency that a particular radio station uses. Changing the value of inductor(coil) by sliding or rotating the knob, the frequency can be changed & matched suitably to desired frequency. Similarly, if ganged capacitor is used then also frequency can be changed as desired. Because both capacitor & inductor can change the frequency.

### S/N ratio: -

•Signal-to-noise ratio (S/N ratio) is a measure used in communication engg. that compares the level of a desired signal to the level of background noise.

•SNR is defined as the ratio of signal power to the noise power, often expressed in decibels. A ratio higher than 1:1 indicates more signal than noise.

## 4.5 Working of superheterodyne radio receiver with Block diagram:

**<u>Super heterodyne receiver block diagram explanation:</u>**

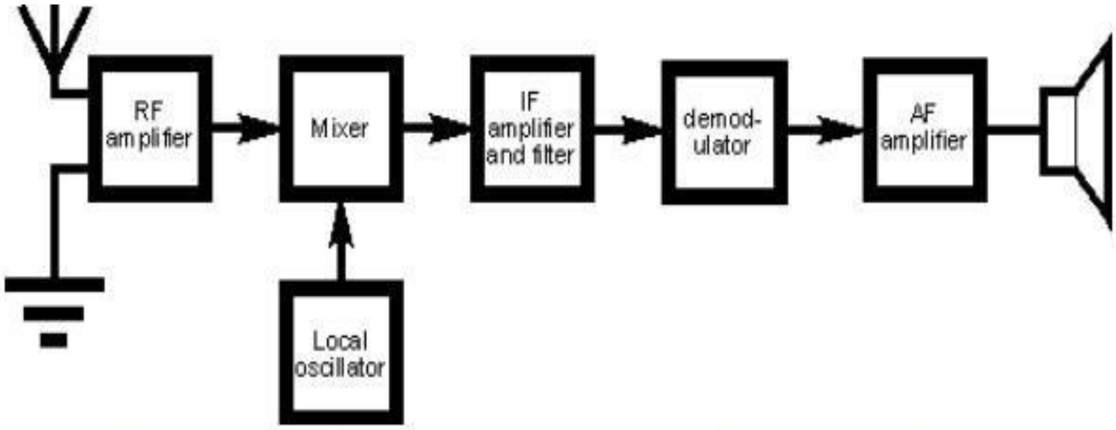



FIG 1.10 Block Diagram of a Basic Superheterodyne Radio Receiver

### i. RF amplifier: -

Signals enter the receiver from the antenna and are applied to the RF amplifier where they are tuned to remove the image signal and also reduce the noise level of unwanted signals of other frequencies that are not required.

### ii. Mixer: -

The signals are then applied to the mixer along with the local oscillator where the wanted signal is converted down to the intermediate frequency. Here frequency conversion are made.

### iii. IF amplifier & Filter: -

Here significant levels of amplification are made and the signals are filtered by using desired filter circuits. This filtering selects signals on one channel against those on the next.

The advantage of the IF filter as opposed to RF filtering is that the filter can be designed for a fixed frequency. This allows for much better tuning.

Variable filters are never able to provide the same level of selectivity that can be provided by fixed frequency ones.

### iv. Demodulator: -

The Output of IF amplifier & filter is fed to the super heterodyne demodulator. This could be for amplitude modulation, single sideband, frequency modulation, or indeed any form of modulation. It is also possible to switch different demodulators in according to the mode being received.

### v. AF amplifier: -

The final element in the super heterodyne receiver block diagram is shown as an audio frequency amplifier, although this could be any form of circuit block that is used to process or amplified the demodulated signal. The out put of this stage is fed to power amplifier say loud speaker or output device etc.

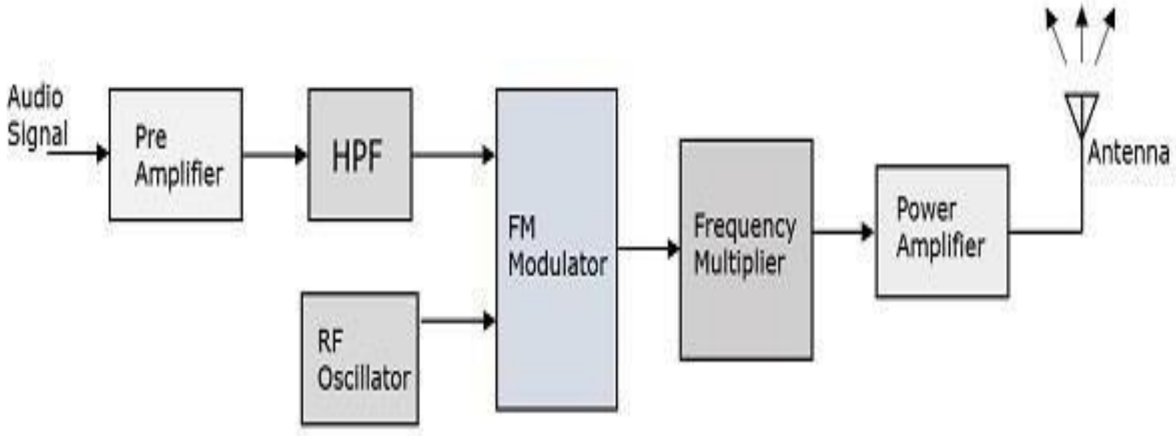
## 4.6 Working of FM Transmitter & Receiver with Block Diagram:

### FM Transmitter: -

The FM transmitter is a low power transmitter and it uses FM waves for transmitting the sound. This transmitter transmits the audio signals through the carrier wave by the difference of frequency.

### Block Diagram of FM Transmitter: -

- FM transmitter is the whole unit, which takes the audio signal as an input and delivers FM wave to the antenna as an output to be transmitted.
- **The block diagram of FM transmitter is shown in the following figure.**



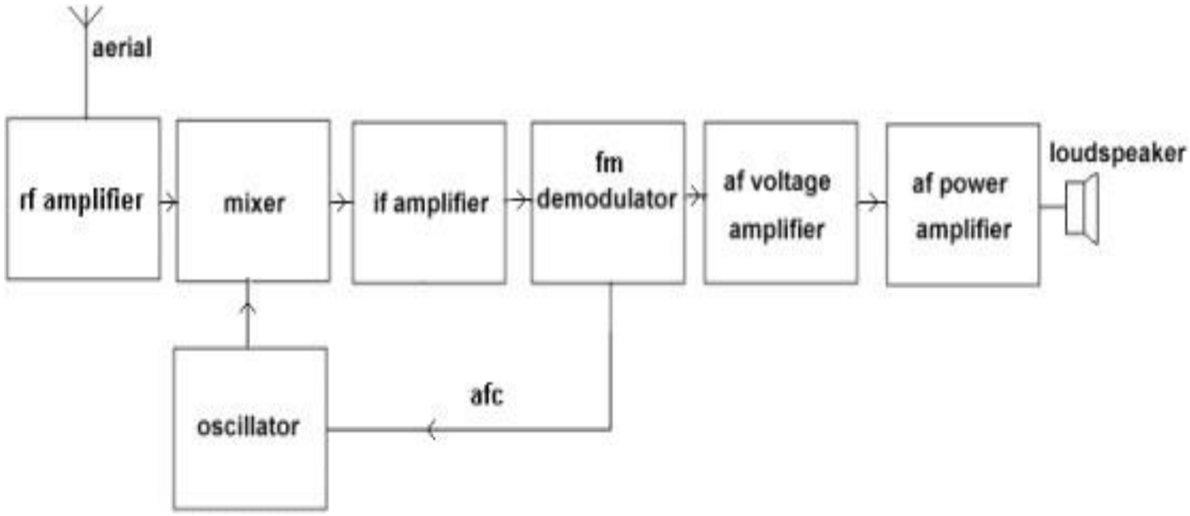

- The working of FM transmitter can be explained as follows.
  1) The audio signal from the output of the microphone is sent to the pre-amplifier, which boosts the level of the modulating signal.
  2) This signal is then passed to high pass filter (HPF), which acts as a pre-emphasis network to filter out the noise and improve the signal to noise ratio.
  3) This signal is further passed to the FM modulator circuit.
  4) The oscillator circuit generates a high frequency carrier, which is sent to the modulator along with the modulating signal.
  5) Several stages of frequency multiplier are used to increase the operating frequency. Here the power of the signal may not sufficient to transmit.
  6) Hence, a RF power amplifier is used at the end to increase the power of the modulated signal.
  7) This FM modulated output is finally passed to the antenna to be transmitted.

## FM Receiver: -

• A radio or FM receiver is an electronic device that receives radio waves and converts the information carried by them to a usable form.

• An antenna is used to catch the desired frequency waves. Frequency modulation is widely used for FM radio broadcasting.

## Block Diagram of FM Receiver: -




### i. RF section: -

• Consists of a pre-selector and an amplifier

• Pre-selector is a broad-tuned band pass filter with an adjustable centre frequency used to reject unwanted radio frequency and to reduce the noise bandwidth.

• RF amplifier determines the sensitivity of the receiver and a predominant factor in determining the noise figure for the receiver.

### ii. Mixer/converter section: -

• Consists of a radio-frequency oscillator and a mixer.

• Choice of oscillator depends on the stability and accuracy desired.

• Mixer is a nonlinear device to convert radio frequency to intermediate frequencies (i.e. heterodyning process).

• The shape of the envelope, the bandwidth and the original information contained in the envelope remains unchanged although the carrier and sideband frequencies are translated from RF to IF.

### iii. IF section: -

• Consists of a series of IF amplifiers and band pass filters to achieve most of the receiver gain and selectivity.

• The IF is always lower than the RF because it is easier and less expensive to construct high-gain, stable amplifiers for low frequency signals.

• IF amplifiers are also less likely to oscillate than their RF counterparts.

### iv. Detector section: -

• To convert the IF signals back to the original source information (demodulation).

• Can be as simple as a single diode or as complex as a PLL or balanced demodulator.

**v. Audio amplifier section: -**

Comprises several cascaded audio amplifiers and one or more speakers

**vi. AGC (Automatic Gain Control): -**

It Adjusts the IF amplifier gain according to signal level (to the average amplitude signal almost constant).

•AGC is a system by means of which the overall gain of radio receiver is varied automatically with the variations in the strength of received signals, to maintain the output constant.

## Possible Short Type Questions with Answers

**1.Define Tuned radio frequency, TRF.**

**Ans:** This type of radio receiver was one of the first that was used. The very first radio receivers of this type simply consisted of a tuned circuit and a detector. Crystal sets were early forms of TRF radios.

**2. Define Super heterodyne receiver.**

**Ans:** The super heterodyne form of radio receiver was developed to provide additional levels of selectivity.

**3. Define Selectivity.**

**Ans:** Selectivity is one of the key attributes of any radio. It needs to accept signals on the wanted frequency and reject others on different frequencies.

•Selectivity is usually measured as a ratio in decibels (dB).

**4. Define Sensitivity.**

**Ans: -** The ability of a receiver to identify and amplify signals at the receiver's input is called Receiver Sensitivity.

- It is expressed in dBm.

**5. Define Fidelity.**

**Ans: -** Fidelity of a receiver is its ability to reproduce the exact replica of the transmitted signals at the receiver output.

**6.Define AM Transmitter.**

**Ans: -** AM transmitter takes the audio signal as an input and delivers amplitude modulated wave to the antenna as an output to be transmitted.

**7.Define FM Transmitter.**

**Ans: -** The FM transmitter is a low power transmitter and it uses FM waves for transmitting the sound, this transmitter transmits the audio signals through the carrier wave by the difference of frequency.

**8. Define FM Receiver.**

**Ans: -** A radio or FM receiver is an electronic device that receives radio waves and converts the information carried by them to a usable form.

**9. Define AGC (Automatic Gain Control).**

**Ans: -** Adjust the IF amplifier gain according to signal level (to the average amplitude signal almost constant).

•AGC is a system by means of which the overall gain of radio receiver is varied automatically with the variations in the strength of received signals, to maintain the output constant.

**10. Define Tuning.**

**Ans: -** A tuner is a subsystem that receives radio frequency (RF) transmissions and converts. The verb tuning in radio contexts means adjusting the receiver to detect the desired radio signal carrier frequency that a particular radio station uses. Changing the value of inductor(coil) by sliding or rotating the knob, the frequency can be changed & matched suitably to desired frequency. Similarly, if ganged capacitor is used then also frequency can be changed as desired. Because both capacitor & inductor can change the frequency.

## Possible Long Type Questions

**1) Define AM Transmitter & Explain with block diagram.**

**2)Explain Super heterodyne receiver with block diagram. [W-20]**

**3) Define FM Transmitter & explain with block diagram.**

**4) Define FM Receiver & explain with block diagram.**

**5) Explain Classification of Radio receiver.**

# CHAPTER 5.0

# ANALOG TO DIGITAL CONVERSION & PULSE MODULATION SYSTEM

***LEARING OBJECTIVES:***

*5.1 Concept of Sampling Theorem, Nyquist rate & Aliasing*

*5.2 Sampling Techniques (Instantaneous, Natural, Flat Top)*

*5.3 Analog Pulse Modulation - Generation and detection of PAM, PWM & PPM system with the help of Block diagram & comparison of all above*

*5.4 Concept of Quantization of signal & Quantization error*

*5.5 Generation & Demodulation of PCM system with Block diagram & its applications.*

*5.6 Companding in PCM & Vocoder*

*5.7 Time Division Multiplexing & explain the operation with circuit diagram*

*5.8 Generation & demodulation of Delta modulation with Block diagram*

*5.9 Generation & demodulation of DPCM with Block diagram*

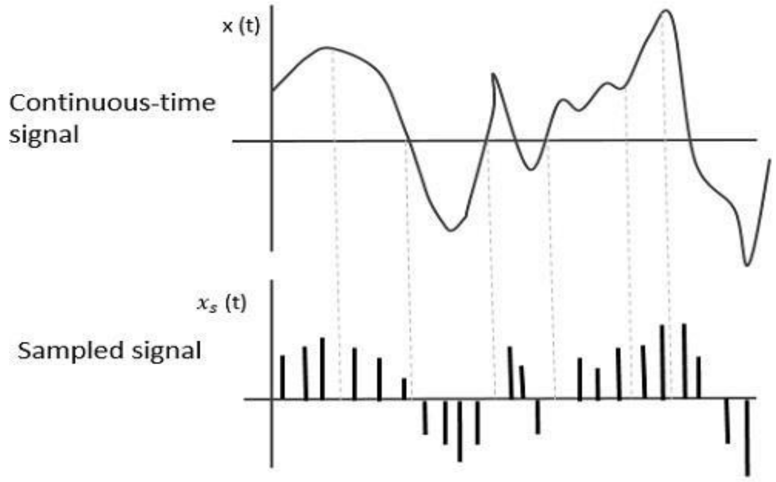
*5.10 Comparison between PCM, DM, ADM & DPCM*

## 5.1 Concept of Sampling Theorem, Nyquist rate & Aliasing

- **Sampling Theorem: -**
  - The Sampling Theorem states that **a signal can be exactly reproduced if it is sampled at a frequency f**, where f is greater than or equal to twice the maximum frequency in the signal. i.**e.** $f_s \geq 2f_m$

- Sampling
  - The process of converting continuous time signals into equivalent discrete time signals, is called **Sampling**. A certain instant of data is continually sampled in the sampling process
    Here two signals are multiplied with the help of multiplier & we obtained a sampled signal

  - The following figure indicates a continuous-time signal **x(t)** and a sampled signal **$x_s$(t)**. When **x(t)** is multiplied by a periodic impulse train, the sampled signal **$x_s$(t)** is obtained.




- A **sampling signal** is a periodic train of pulses, having **unit amplitude**, sampled at equal intervals of time $\mathbf{T_s}$, which is called as the **Sampling time**. This data is transmitted at the time instants $\mathbf{T_s}$ and the carrier signal is transmitted at the remaining time.

➢ **<u>Nyquist rate & Aliasing: -</u>**

**NYQUIST RATE:**

- Nyquist rate is the rate at which sampling of a signal is done so that overlapping of frequency does not take place. When the sampling rate become exactly equal to $2f_m$ samples per second, then the specific rate is known as Nyquist rate. It is also known as the minimum sampling rate and given by: $\mathbf{f_s = 2f_m}$

**Effect of Under sampling: ALIASING**

- It is the effect in which overlapping of a frequency component takes place at the frequency higher than Nyquist rate. Signal loss may occur due to aliasing effect. We can say that aliasing is the phenomena in which a high frequency component in the frequency spectrum of a signal takes identity of a lower frequency component in the same spectrum of the sampled signal.
- Because of overlapping due to process of aliasing, sometimes it is not possible to overcome the sampled signal x(t) from the sampled signal g(t) by applying the process of low pass filtering since the spectral components in the overlap regions. hence this causes the signal to destroy.

➢ **The Effect of Aliasing can be reduced:**

1) Pre alias filter must be used to limit band of frequency of the required signal $f_m$ Hz.

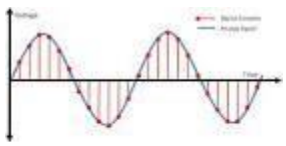
**2)** Sampling frequency $f_s$ must be selected such that $f_s > 2f_m$.

## 5.2 Sampling Techniques (Instantaneous, Natural, Flat Top)

- **Sampling** is the process of converting analog signal into a discrete signal or making an analog or continuous signal to occur at a particular interval of time, this phenomena is known as sampling.

❖ **SAMPLING THEOREM: -**

- Sampling theorem states that a band limited signal having no frequency components higher than fm hertz can be sampled if its sampling freq is equal to or greater than Nyquist rate.



Analog Signal Representation

➢ **Sampling Techniques**

- There are basically three types of Sampling techniques, namely:

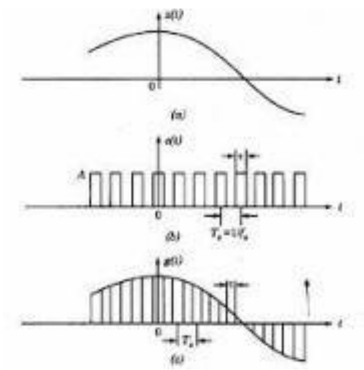
  **1.** Natural Sampling

  **2.** Flat top Sampling

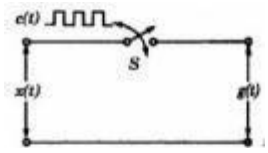
  **3.** Ideal Sampling

**1. Natural Sampling:**

- Natural Sampling is a practical method of sampling in which pulse have finite width equal to τ. Sampling is done in accordance with the carrier signal which is digital in nature.



Natural Sampled Waveform



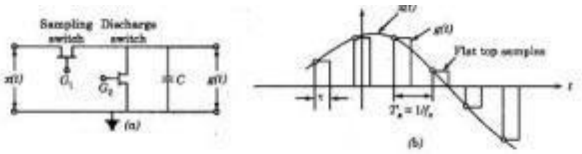

Functional Diagram of Natural Sampler

- With the help of functional diagram of a Natural sampler, a sampled signal g(t) is obtained by multiplication of sampling function c(t) and the input signal x(t).
- Spectrum of Natural Sampled Signal is given by:

**$G(f) = A\tau / T_s .[\, \Sigma \sin c(n\, f_s.\tau)\, X(f\text{-}n\, f_s)]$**

**2. Flat Top Sampling:**

- Flat top sampling is like natural sampling i.e; practical in nature. In comparison to natural sampling flat top sampling can be easily obtained. In this sampling techniques, the top of the samples remains constant and is equal to the instantaneous value of the message signal x(t) at the start of sampling process. Sample and hold circuit are used in this type of sampling.




Block Diagram and Waveform

- F**igure(a),** shows functional diagram of a sample hold circuit which is used to generate fat top samples.
- **Figure(b),** shows the general waveform of the flat top samples. It can be observed that only starting edge of the pulse represent the instantaneous value of the message signal x(t).

**Spectrum of Flat top Sampled Signal is given by: $G(f) = f_s .[\, \Sigma\, X(f\text{-}n\, f_s).\, H(f)]$**

**3. Ideal Sampling:**

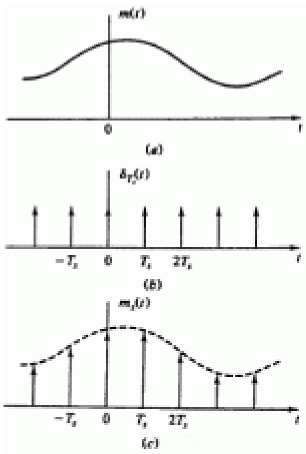


Fig. 5-2 Ideal signal sampling

Ideal Sampling Wave form

- Ideal Sampling is also known as Instantaneous sampling or Impulse Sampling.Train of impulse is used as a carrier signal for ideal sampling.
- In this sampling technique the sampling function is a train of impulses and the principle used is known as multiplication principle.

**Here,**

**Figure (a)**, represent message signal or input signal or signal to be sampled.

**Figure (b),** represent the sampling function.

**Figure (c),** represent the resultant signal.

**Spectrum of Ideal Sampled Signal is given by: $G(f) = f_s .[ \Sigma X(f-n f_s)]$**

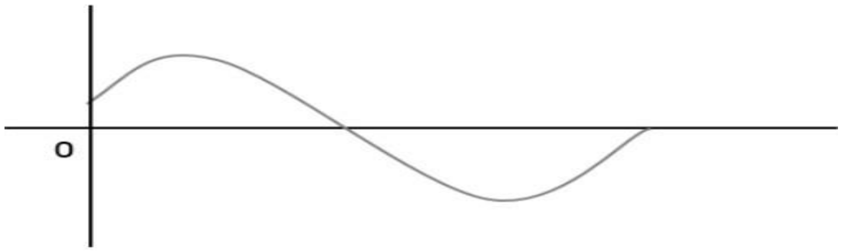
## 5.3 Analog Pulse Modulation - Generation and detection of PAM, PWM & PPM system with the help of Block diagram & comparison of all above

- **Analog Pulse Modulation**

- After the continuous wave modulation, the next division is Pulse modulation. Pulse modulation is further divided into analog and digital modulation. The analog modulation techniques are mainly classified into Pulse Amplitude Modulation, Pulse Duration Modulation/Pulse Width Modulation, and Pulse Position Modulation.

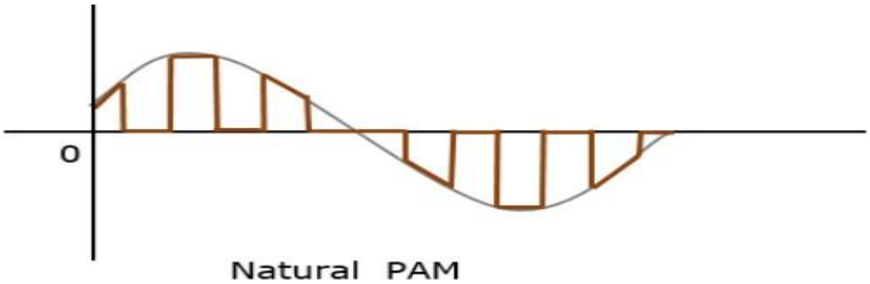
### Pulse Amplitude Modulation (PAM)

- **Pulse Amplitude Modulation (PAM)** is an analog modulating scheme in which the amplitude of the pulse carrier varies proportional to the instantaneous amplitude of the message signal.
- The pulse amplitude modulated signal, will follow the amplitude of the original signal, as the signal traces out the path of the whole wave. In natural PAM, a signal sampled at the Nyquist rate is reconstructed, by passing it through an efficient **Low Pass Frequency (LPF)** with exact cutoff frequency
- The following figures explain the Pulse Amplitude Modulation.



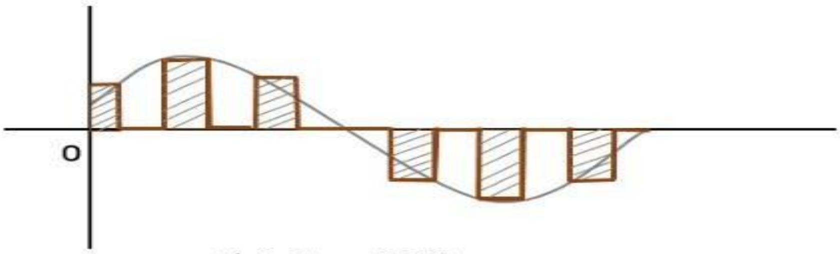
Modulating signal

Carrier Pulse train



Natural PAM

- Though the PAM signal is passed through an LPF, it cannot recover the signal without distortion. Hence to avoid this noise, flat-top sampling is done as shown in the following figure.



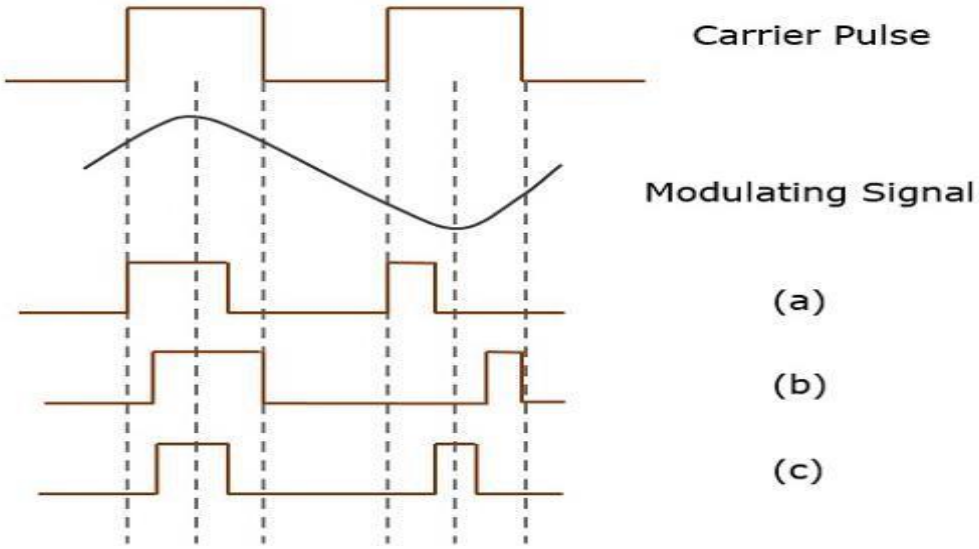
Flat-Top PAM

- **Flat-top sampling** is the process in which sampled signal can be represented in pulses for which the amplitude of the signal cannot be changed with respect to the analog signal, to be sampled. The tops of amplitude remain flat. This process simplifies the circuit design.

## ➢ <u>Pulse Width Modulation (PWM)</u>: -

- **Pulse Width Modulation (PWM)** or **Pulse Duration Modulation (PDM)** or **Pulse Time Modulation (PTM)** is an analog modulating scheme in which the duration or width or time of the pulse carrier varies proportional to the instantaneous amplitude of the message signal.

- The width of the pulse varies in this method, but the amplitude of the signal remains constant. Amplitude limiters are used to make the amplitude of the signal constant. These circuits clip off the amplitude, to a desired level and hence the noise is limited.
- The following figures explain the types of Pulse Width Modulations.



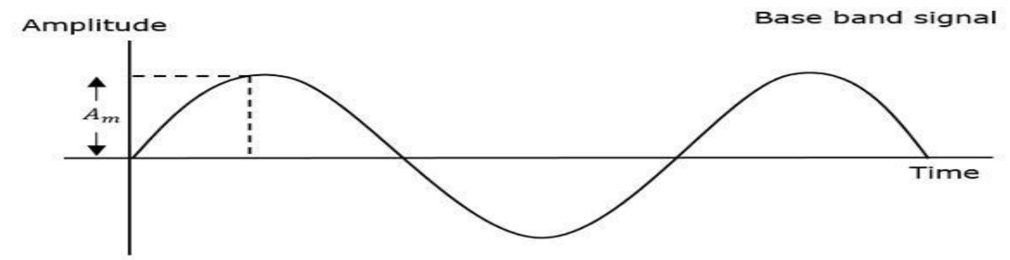

There are three variations of PWM. They are −

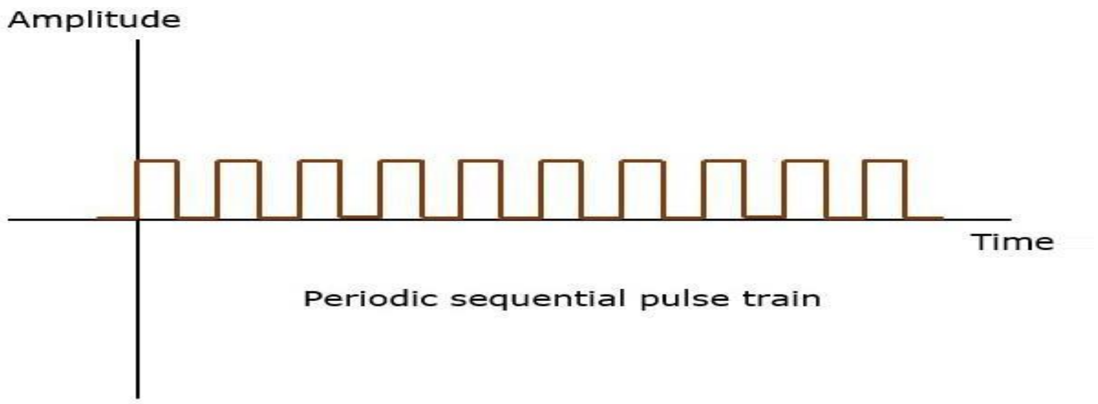
- The leading edge of the pulse being constant, the trailing edge varies according to the message signal.
- The trailing edge of the pulse being constant, the leading edge varies according to the message signal.
- The centre of the pulse being constant, the leading edge and the trailing edge varies according to the message signal.

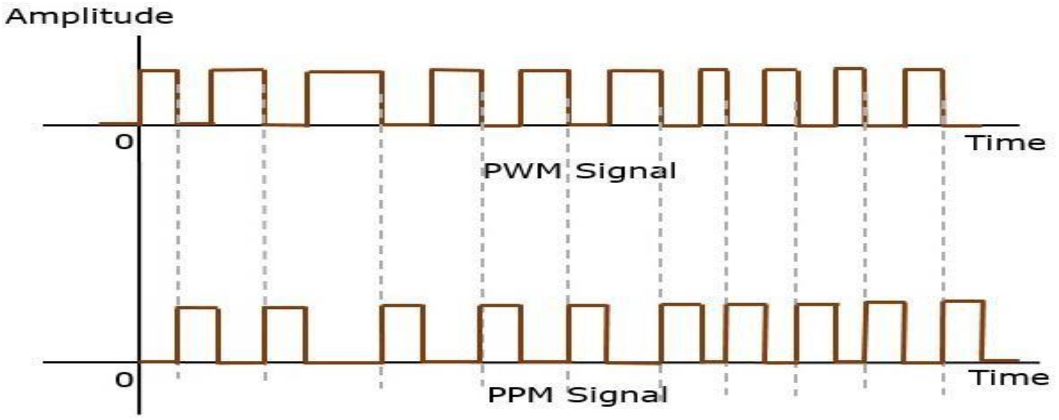
These three types are shown in the above given figure, with timing slots.

## ➢ Pulse Position Modulation (PPM)

- **Pulse Position Modulation (PPM)** is an analog modulating scheme in which the amplitude and width of the pulses are kept constant, while the position of each pulse, with reference to the position of a reference pulse varies according to the instantaneous sampled value of the message signal.
- The transmitter has to send synchronizing pulses (or simply sync pulses) to keep the transmitter and receiver in synchronism. These sync pulses help maintain the position of the pulses.
- The following figures explain the Pulse Position Modulation.




- Pulse position modulation is done in accordance with the pulse width modulated signal. Each trailing of the pulse width modulated signal becomes the starting point for pulses in PPM signal. Hence, the position of these pulses is proportional to the width of the PWM pulses.

## Advantage

- As the amplitude and width are constant, the power handled is also constant.

## Disadvantage

- The synchronization between transmitter and receiver is a must.

## ➢ Comparison between PAM, PWM, and PPM

The comparison between the above modulation processes is presented in a single table.

| PAM | PWM | PPM |
|---|---|---|
| Amplitude is varied | Width is varied | Position is varied |
| Bandwidth depends on the width of the pulse | Bandwidth depends on the rise time of the pulse | Bandwidth depends on the rise time of the pulse |
| Instantaneous transmitter power varies with the amplitude of the pulses | Instantaneous transmitter power varies with the amplitude and width of the pulses | Instantaneous transmitter power remains constant with the width of the pulses |
| System complexity is high | System complexity is low | System complexity is low |
| Noise interference is high | Noise interference is low | Noise interference is low |
| It is similar to amplitude modulation | It is similar to frequency modulation | It is similar to phase modulation |

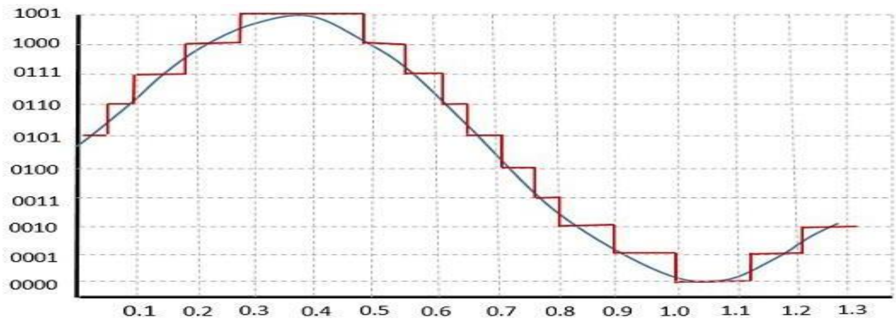
# 5.4 Concept of Quantization of signal & Quantization error

## ➢ Quantization

- The digitization of analog signals involves the rounding off of the values which are approximately equal to the analog values. The method of sampling chooses few points on the analog signal and then these points are joined to round off the value to a near stabilized value. Such a process is called as **Quantization**.

### ➢ Quantization of signal & Quantization error

- The quantizing of an analog signal is done by discretizing the signal with a number of quantization levels. Quantization is representing the sampled values of the amplitude by a finite set of levels, which means converting a **continuous-amplitude sample** into a **discrete-time signal**.
- The following figure shows how an analog signal gets quantized. The blue line represents analog signal while the red one represents the quantized signal.




- Both sampling and quantization results in the loss of information. The quality of a Quantizer output depends upon the number of quantization levels used.

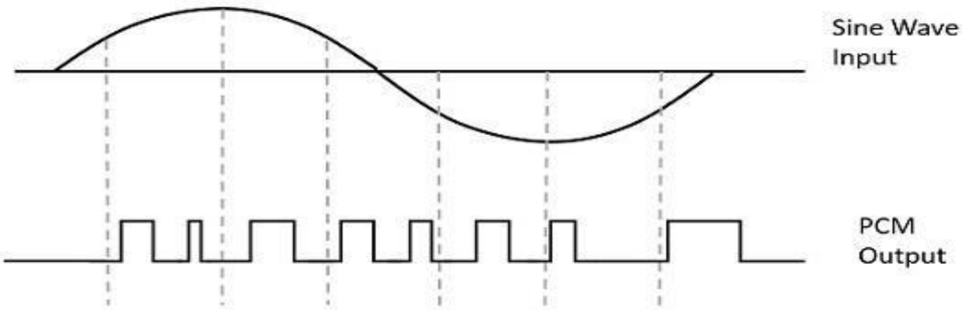
## ➢ Quantization error

- Quantization error is **the difference between the analog signal and the closest available digital value at each sampling instant from the A/D converter**. Quantization error also introduces noise, called quantization noise, to the sample signal.

# 5.5 Generation & Demodulation of PCM system with Block diagram & its applications.

## ➢ Pulse Code Modulation

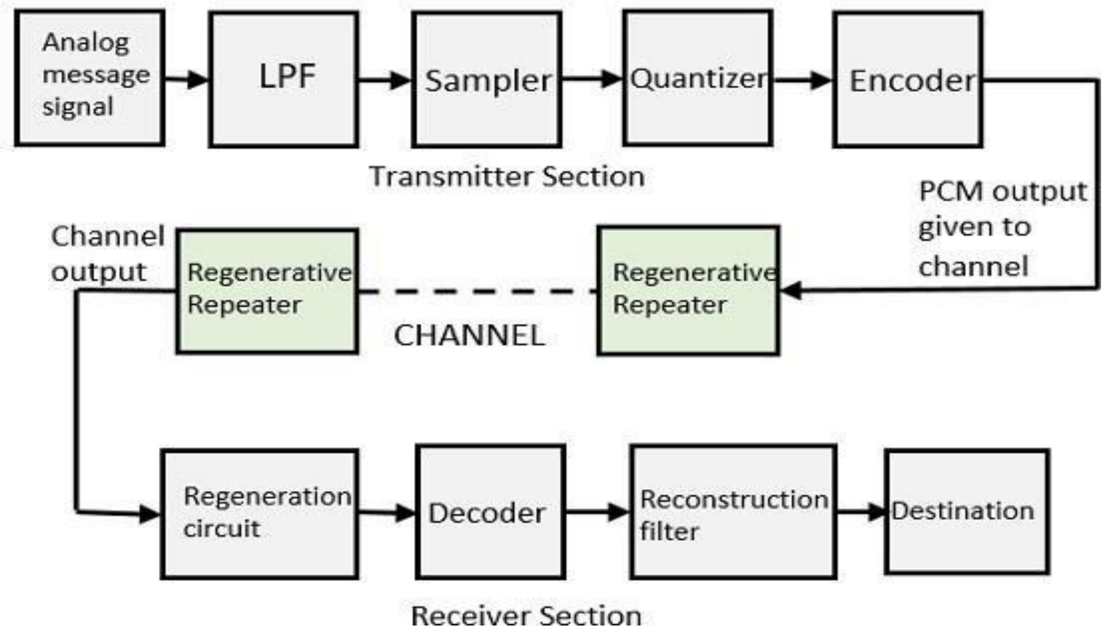
- A signal is Pulse Code modulated to convert its analog information into a binary sequence, i.e., 1s and 0s. The output of a **Pulse Code Modulation (PCM)** will resemble a binary sequence.
- The following figure shows an example of PCM output with respect to instantaneous values of a given sine wave.




- Instead of a pulse train, PCM produces a series of numbers or digits, and hence this process is called as digital. Each one of these digits, though in binary code, represent the approximate amplitude of the signal sample at that instant.
- In Pulse Code Modulation, the message signal is represented by a sequence of coded pulses. This message signal is achieved by representing the signal in discrete form in both time and amplitude.

# Basic Elements of PCM

- The transmitter section of a Pulse Code Modulator circuit consists of **Sampling, Quantizing** and **Encoding**, which are performed in the **analog-to-digital converter** section. The low pass filter prior to sampling prevents aliasing of the message signal.
- The basic operations in the receiver section are **regeneration of impaired signals, decoding,** and **reconstruction** of the quantized pulse train.
- The following figure is the block diagram of PCM which represents the basic elements of both the transmitter and the receiver sections.




## Low Pass Filter (LPF)

- This filter eliminates the high frequency components present in the input analog signal which is greater than the highest frequency of the message signal, to avoid aliasing of the message signal.

## Sampler

- This is the circuit which uses the technique that helps to collect the sample data at instantaneous values of the message signal, so as to reconstruct the original signal. The sampling rate must be greater than twice the highest frequency component **W** of the message signal, in accordance with the sampling theorem.

## Quantizer

- Quantizing is a process of reducing the excessive bits and confining the data. The sampled output when given to Quantizer, reduces the redundant bits and compresses the value.

## Encoder

- The digitization of analog signal is done by the encoder. It designates each quantized level by a binary code. The sampling done here is the sample-and-hold process. These three sections will act as an analog to the digital converter. Encoding minimizes the bandwidth used.

## Regenerative Repeater

- The output of the channel has one regenerative repeater circuit to compensate the signal loss and reconstruct the signal. It also increases the strength of the signal.

## Decoder

- The decoder circuit decodes the pulse coded waveform to reproduce the original signal. This circuit acts as the **demodulator**.

## Reconstruction Filter

- After the digital-to-analog conversion is done by the regenerative circuit and the decoder, a low pass filter is employed, called as the reconstruction filter to get back the original signal.
- Hence, the Pulse Code Modulator circuit digitizes the analog signal given, codes it, and samples it. It then transmits in an analog form. This whole process is repeated in a reverse pattern to obtain the original signal.

# 5.6 Companding in PCM & Vocoder

- ➢ **Companding in PCM**

    - For digital audio signals, companding is used in pulse code modulation (PCM). The process involves **decreasing the number of bits used to record the strongest (loudest) signals**. In the digital file format, companding improves the signal-to-noise ratio at reduced bit rates.

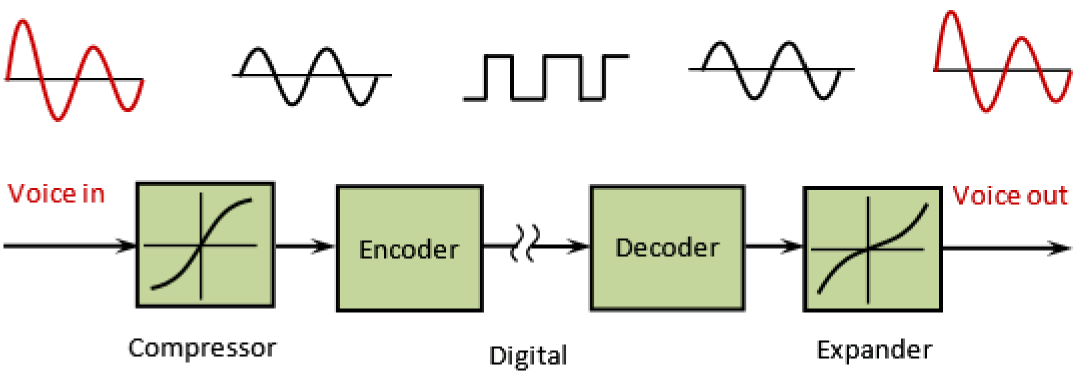
Non uniform quyantizers are difficult to make and expencive.

An alternative is to first pass the speech signal through a nonlinearity before quantising with a uniform quantizer.

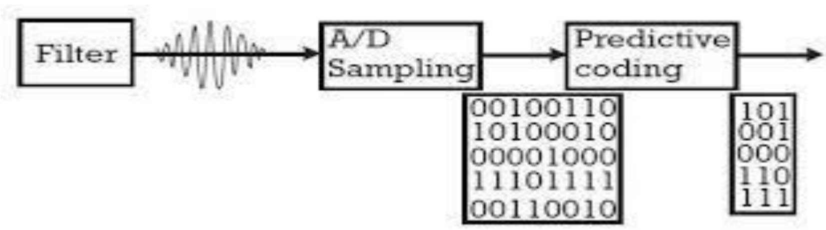
The nonlinearity causes the signal amplitude to be compressed.The input to the quantizer will have a more uniform distribution.

At the receiver the signsl is explanded by an inverse to the nonlinearity.

The process of compressing & expanding is called companding.




- **Vocoder**
    - A vocoder **analyzes and transfers the sonic character of the audio signal arriving at its analysis input to synthesizer sound generators**.

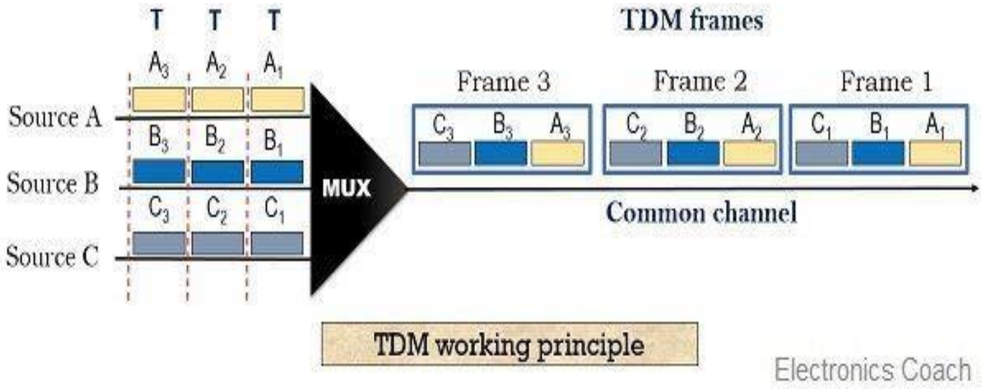



Function of Vocoder

## 5.7 Time Division Multiplexing & explain the operation with circuit diagram

- **Time Division Multiplexing (TDM):**
    - As we know, multiplexing allows the transmission of several signals over a common channel. However, one may need to differentiate between the various signal for proper data transmission.
    - So, in **time division multiplexing, the complete signal gets transmitted by occupying different time slots**.

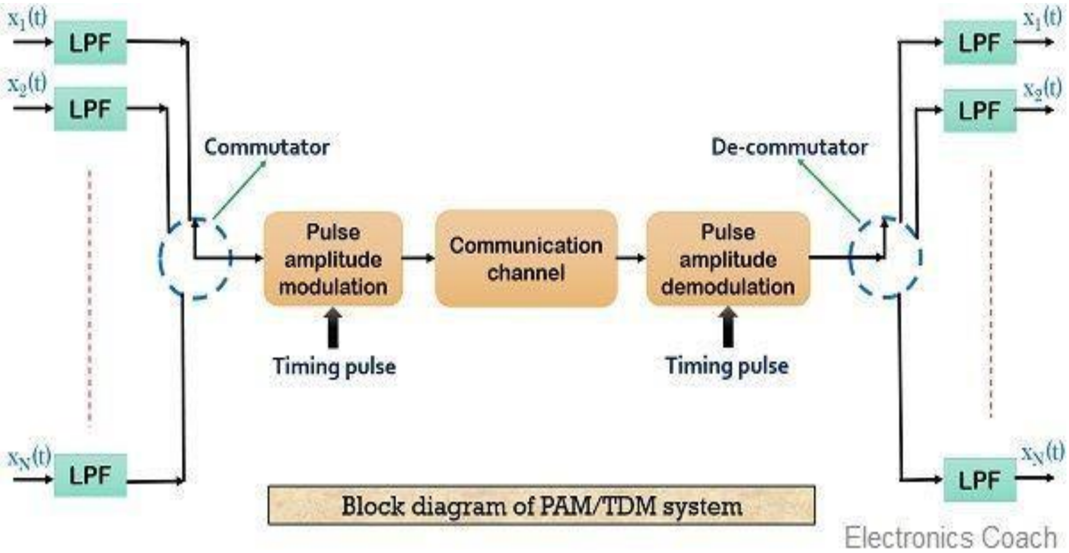
- Let us have a look at the figure below in order to have a better understanding of the TDM process.




- As we can see that source A, B and C wants to transmit data through a common medium. Thus, the signal from the 3 sources, is divided into multiple frames each having their fixed time slot. Here, 3 units from each source are taken into consideration, that jointly form the actual signal.

## ➢ <u>TDM system</u>

- The figure below shows the block diagram of a TDM system employing both transmitter and receiver section.




- 

- The technique efficiently utilizes the complete channel for data transmission hence sometimes known as PAM/TDM. This is so because a TDM system uses a pulse amplitude modulation.

- In this modulation technique, each pulse holds some short time duration allowing maximal channel usage.

- Here at the beginning, the system consists of multiple LPF depending on the number of data inputs. These low pass filters are basically anti-aliasing filters that eliminate the aliasing of the data input signal.

- The output of the LPF is then fed to the commutator. As per the rotation of the commutator the samples of the data inputs are collected by it.

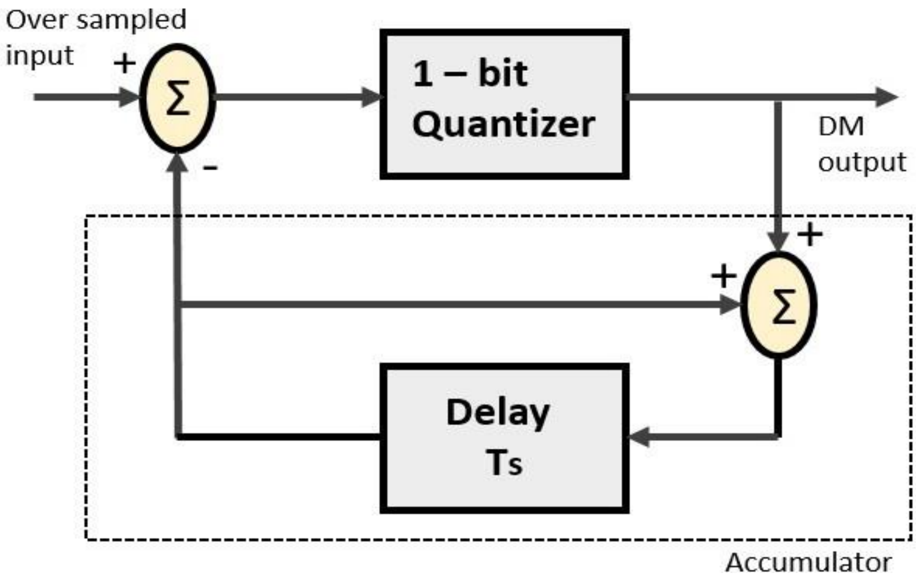
- Here, $f_s$ is the rate of rotation of the commutator, thus denotes the sampling frequency of the system.

## 5.8 Generation & demodulation of Delta modulation with Block diagram

- The type of modulation, where the sampling rate is much higher and in which the stepsize after quantization is of smaller value **Δ**, such a modulation is termed as **delta modulation**.

## Delta Modulator

- The **Delta Modulator** comprises of a 1-bit quantizer and a delay circuit along with two summer circuits. Following is the block diagram of a delta modulator.

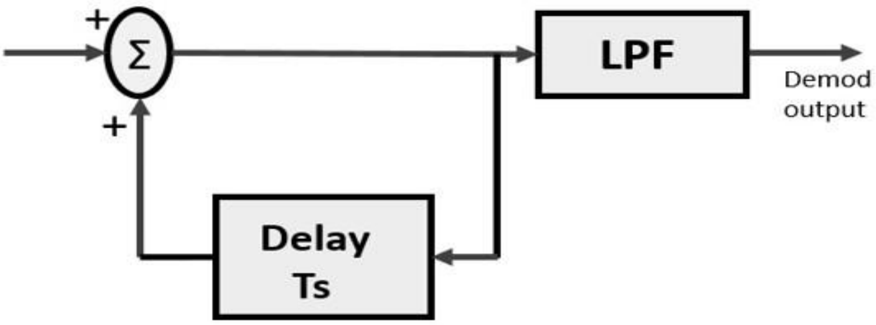



- A stair-case approximated waveform will be the output of the delta modulator with the step-size as delta (**Δ**). The output quality of the waveform is moderate.

## Delta Demodulator

- The delta demodulator comprises of a low pass filter, a summer, and a delay circuit. The predictor circuit is eliminated here and hence no assumed input is given to the demodulator.

- Following is the block diagram for delta demodulator.

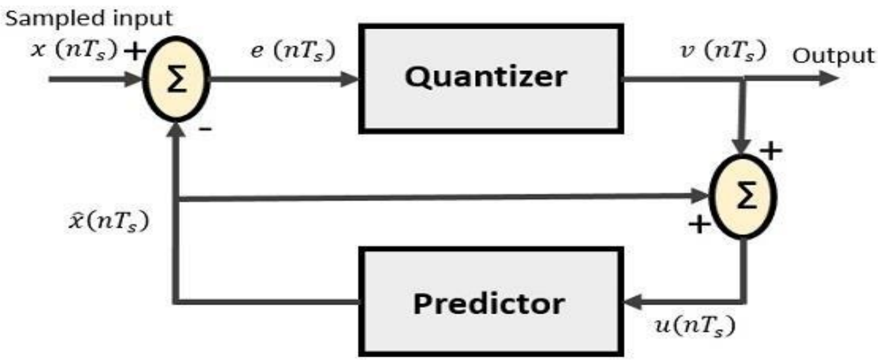



- Low pass filter is used for many reasons, but the prominent one is noise elimination for out-of-band signals. The step-size error that may occur at the transmitter is called **granular noise**, which is eliminated here. If there is no noise present, then the modulator output equals the demodulator input.

## 5.9 Generation & demodulation of DPCM with Block diagram

### ➢ DPCM Transmitter

- The DPCM Transmitter consists of Quantizer and Predictor with two summer circuits. Following is the block diagram of DPCM transmitter.

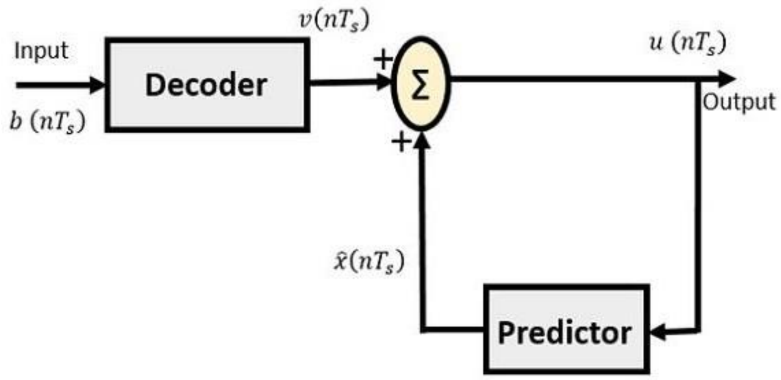



The signals at each point are named as −

- $x(nT_s)$ is the sampled input
- $\hat{x}(nT_s)$ is the predicted sample
- $e(nT_s)$ is the difference of sampled input and predicted output, often called as prediction error
- $v(nT_s)$ is the quantized output
- $u(nT_s)$ is the predictor input which is actually the summer output of the predictor output and the quantizer output

## ➢ DPCM Receiver

- The block diagram of DPCM Receiver consists of a decoder, a predictor, and a summer circuit. Following is the diagram of DPCM Receiver.




- The notation of the signals is the same as the previous ones. In the absence of noise, the encoded receiver input will be the same as the encoded transmitter output.
- As mentioned before, the predictor assumes a value, based on the previous outputs. The input given to the decoder is processed and that output is summed up with the output of the predictor, to obtain a better output.

## ➢ Applications of DPCM

- The DPCM technique mainly used Speech, image and audio signal compression. The DPCM conducted on signals with the correlation between successive samples leads to good compression ratios.

# 5.10 Comparison between PCM, DM, ADM & DPCM

### ➢ Advantages of DM over DPCM

- 1-bit quantizer
- Very easy design of modulator & demodulator

However, there exists some **noise in DM** and following are the types of noise.

- Slope Over load distortion (when Δ is small)
- Granular noise (when Δ is large)

### ➢ Adaptive Delta Modulation

- In digital modulation, we come across certain problems in determining the step-size, which influences the quality of the output wave.
- The larger step-size is needed in the steep slope of modulating signal and a smaller stepsize is needed where the message has a small slope. As a result, the minute details get missed.
- Hence, it would be better if we can control the adjustment of step-size, according to our requirement in order to obtain the sampling in a desired fashion. This is the concept of **Adaptive Delta Modulation (ADM)**.

## Possible Short Type Questions with Answers

1. **Define Quantizer?**
   ANS: Quantizing is a process of reducing the excessive bits and confining the data. The sampled output when given to Quantizer, reduces the redundant bits and compresses the value.
2. **Define Delta Modulator?**
   ANS: The **Delta Modulator** comprises of a 1-bit quantizer and a delay circuit along with two summer circuits. Following is the block diagram of a delta modulator.

3. **Define Pulse Amplitude Modulation (PAM)?**

   ANS: **Pulse Amplitude Modulation (PAM)** is an analog modulating scheme in which the amplitude of the pulse carrier varies proportional to the instantaneous amplitude of the message signal.
4. **Define Sampling?**

   ANS: **Sampling** is the process of converting analog signal into a discrete signal or making an analog or continuous signal to occur at a particular interval of time, this phenomena is known as sampling.
5. **Define Sampling Theorem?**

   ANS: **Sampling Theorem: -**

   - The Sampling Theorem states that **a signal can be exactly reproduced if it is sampled at a frequency F**, where F is greater than twice the maximum frequency in the signal.

## Possible Long Type Questions

1. **Explain Demodulation of DPCM with Block diagram.**
2. **Explain Delta Modulator with Block diagram.**
3. **Explain TDM system with Block diagram. (W-20)**
4. **Explain Basic Elements of PCM with Block diagram. (W-18)**
5. **Explain Pulse Width Modulation (PWM) with Block diagram.**
6. **Explain Comparison between PAM, PWM, and PPM.**
7. **Describe the operation of PAM with neat wave form. (W-20)**

# Unit-6

# DIGITAL MODULATION TECHNIQUES

***LEARING OBJECTIVES:***

*6.1 Concept of Multiplexing (FDM & TDM)- (Basic concept, Transmitter & Receiver) & Digital modulation formats*

*6.2 Advantages of digital communication system over Analog system*

*6.3 Digital modulation techniques & types*

*6.4 Generation and Detection of binary ASK, FSK, PSK, QPSK, QAM, MSK, GMSK*

*6.5 Working of T1-Carrier system*

*6.6 Spread Spectrum & its applications*

*6.7 Working operation of Spread Spectrum Modulation Techniques (DS-SS & FH-SS)*

*6.8 Define bit, Baud, symbol & channel capacity formula. (Shannon Theorems)*

*6.9 Application of Different Modulation Schemes*

*6.10 Types of Modem & its application*

## 6.1 Concept of Multiplexing (FDM & TDM)- (Basic concept, Transmitter & Receiver) & Digital modulation formats.

### Concept of Multiplexing (FDM & TDM): -

➢ **Multiplexing: -**

Multiplexing is a technique which combines multiple signals into one signal, suitable for transmission over a single communication channel such as coaxial cable or optical fiber.

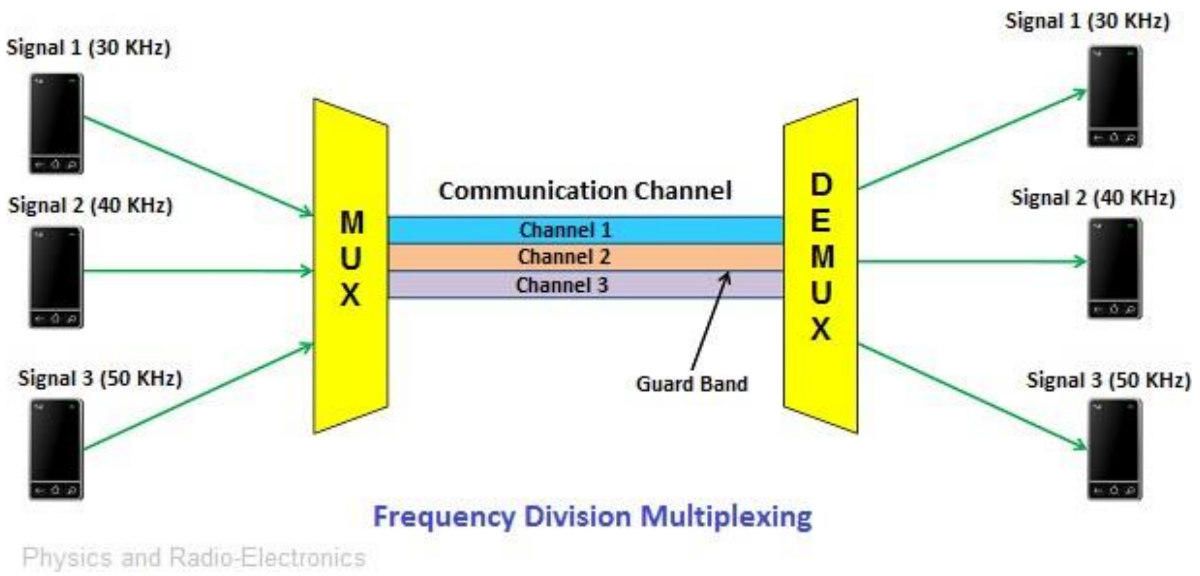
➢ **Types of Multiplexing: -**

There are mainly two types of multiplexing namely

1. Frequency Division Multiplexing (FDM).
2. Time Division Multiplexing (TDM).

### 1. Frequency Division Multiplexing (FDM): -

- This technique combines multiple signals into one signal and transmitted over the single communication channel & combining more than one signal over a shared medium.
- In FDM, signals of different frequencies are combined for concurrent transmission.
- Frequency division multiplexing is an analog technique. It is the most popular & used extensively in TV and radio transmission.
- In this technique, the bandwidth of the communication channel should be greater than the combined bandwidth of individual signals.




**Advantages of Frequency Division Multiplexing (FDM): -**

a) It transmits multiple signals simultaneously.

**b)** In this case, the demodulation process is easy.

**Disadvantages of Frequency Division Multiplexing (FDM): -**

1.It needs a large bandwidth communication channel.

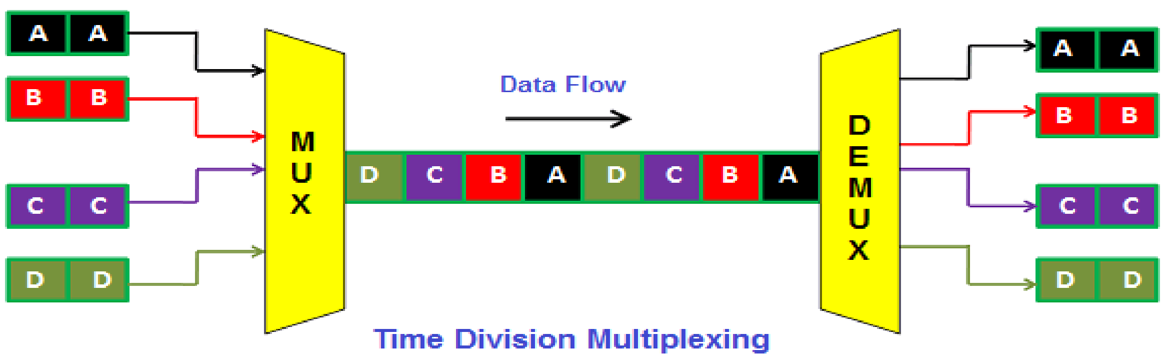
**Applications of Frequency Division Multiplexing (FDM): -**

a) Frequency division multiplexing is used for FM and AM radio broadcasting.

b) It is used in television broadcasting.

## 2. Time Division Multiplexing (TDM): -

- Time Division Multiplexing is a technique in which multiple signals are combined and transmitted one after another in series on the same communication channel.
- At the receiver side, the received signals are separated . Each signal is received by an user at a different time.
- In time division multiplexing, all signals operate with the same frequency but transmitted at different times.
- In time division multiplexing, the sharing of a channel is done on the basis of time.



- In time division multiplexing, each user is allotted a particular time interval called time slot during which data is transmitted.
- For example, as shown in the above figure, at first, we send signal A. Then after second signal B and then after third signal C and finally last signal D. Thus, each user occupies an entire bandwidth for a short period of time.

**Advantages of Time Division Multiplexing (TDM):-**

1. Full bandwidth is utilized by a user at a particular time.

2. The time division multiplexing technique is more flexible than frequency division multiplexing.

**Disadvantages of Time Division Multiplexing (TDM): -**

1.In time division multiplexing, synchronization is required.

**Applications of Time Division Multiplexing (TDM): -**

1. Communication system

2. Computer memory

3. Telephone systems

## Digital modulation formats: -

- Digital modulation formats often use an "IQ" signal representation in order to represent a given modulation scheme, however this can appear abstract at first.

### 6.2 Advantages of digital communication system over Analog system

There are many advantages of digital communication over analog communication, such as

1. The effect of distortion, noise, and interference is less.
2. Digital circuits are easy to design, reliable and cheaper than analog circuits.
3. The hardware implementation in digital circuits, is more flexible than analog.
4. The probability of error occurrence is reduced by employing error detecting and error correcting codes.
5. Combining digital signals using time division multiplexing (TDM) is easier than combining analog signals using frequency division multiplexing (FDM).
6. The configuring process of digital signals is easier than analog signals.

## 6.3 Digital modulation techniques & types

- Digital Modulation provides more information capacity, high data security, quicker system availability with great quality communication.
- ➢ The most fundamental digital modulation techniques & types are: -

1. ASK (amplitude-shift keying)
2. FSK (frequency-shift keying)
3. PSK (phase-shift keying)
4. QAM (quadrature amplitude modulation)
5. QPSK (Quadrature Phase Shift Keying)
6. MSK (Minimum Shift Keying)
7. GMSK (Gaussian Minimum Shift Keying)

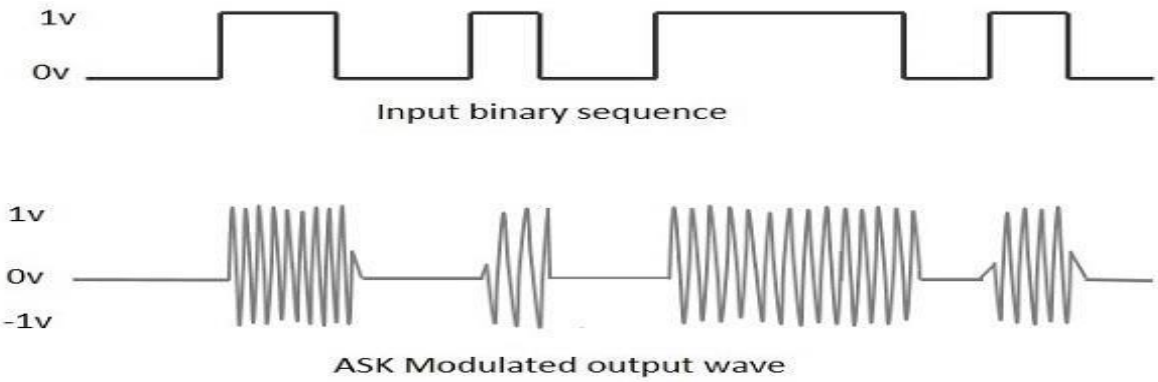
# 6.4 Generation and Detection of binary ASK, FSK, PSK, QPSK, QAM, MSK, GMSK

## 1. ASK (Amplitude-shift keying): -

- 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.

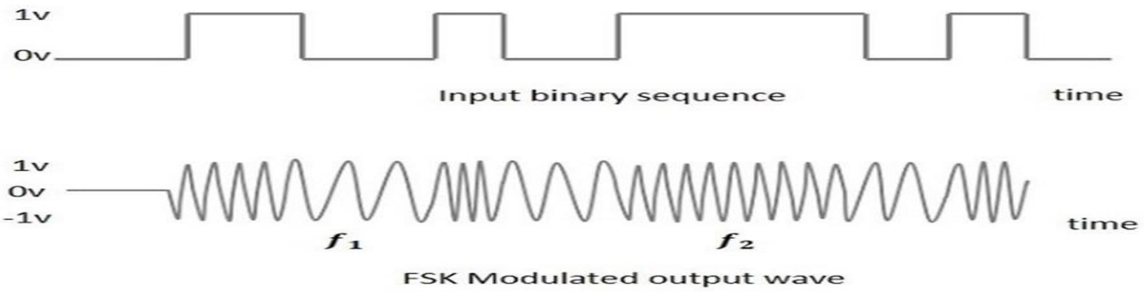
Any modulated signal has a high frequency carrier. The output of an ASK modulated wave is high in amplitude for a binary High input and is low in amplitude for a binary Low input.

- The following figure represents ASK modulated waveform along with its input.




## 2. FSK (frequency-shift keying): -

- 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 an FSK modulated wave is high in frequency for a binary High input and is low in frequency for a binary Low input.
- The following image is the diagrammatic representation of FSK modulated waveform along with its input.

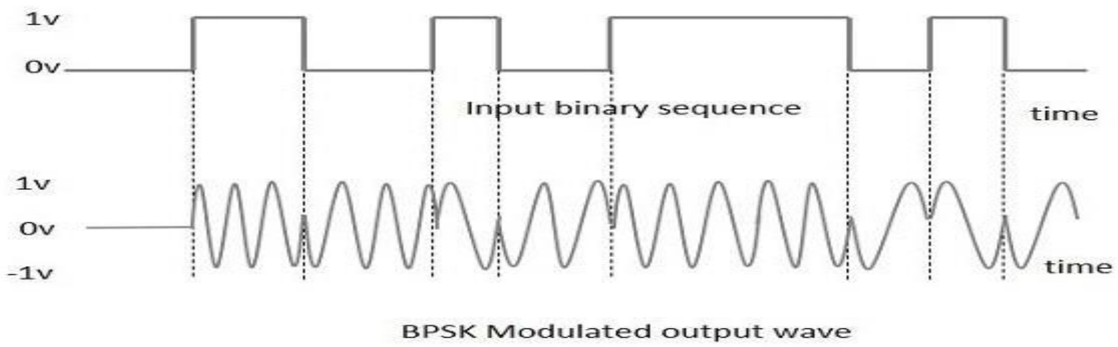



## 3. PSK (phase-shift keying): -

- 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 –

## 1) 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°.

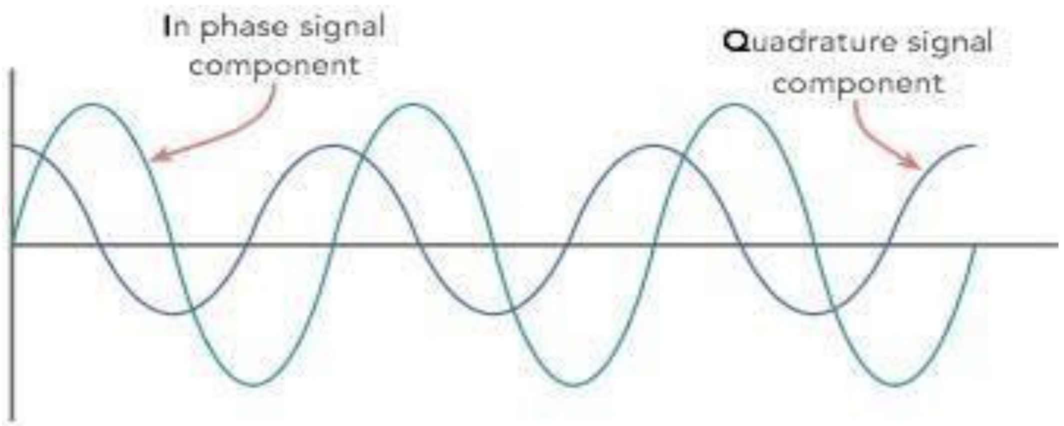



## 2) Quadrature Phase Shift Keying (QPSK): -

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

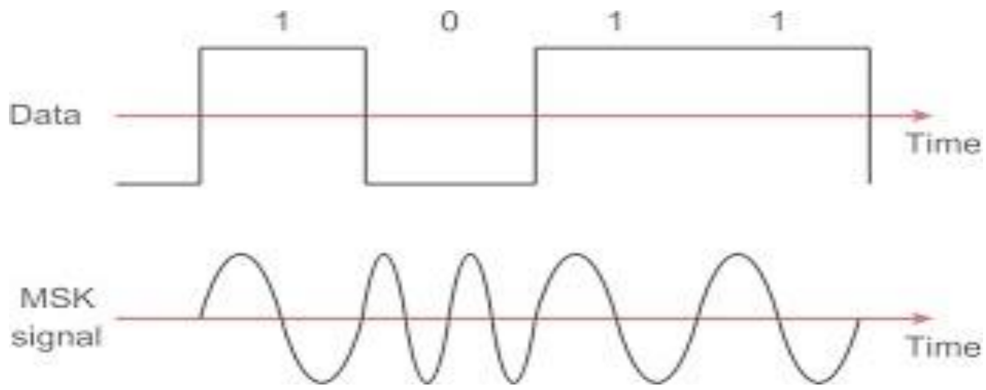
## 4. QAM (quadrature amplitude modulation):-

- Quadrature Amplitude Modulation, QAM is a signal in which two carriers shifted in phase by 90 degrees (i.e. sine and cosine) are modulated and combined.
- The resultant overall signal consisting of the combination of both I and Q carriers contains of both amplitude and phase variations. I means in phase signal omponent & Q means quadrature component.



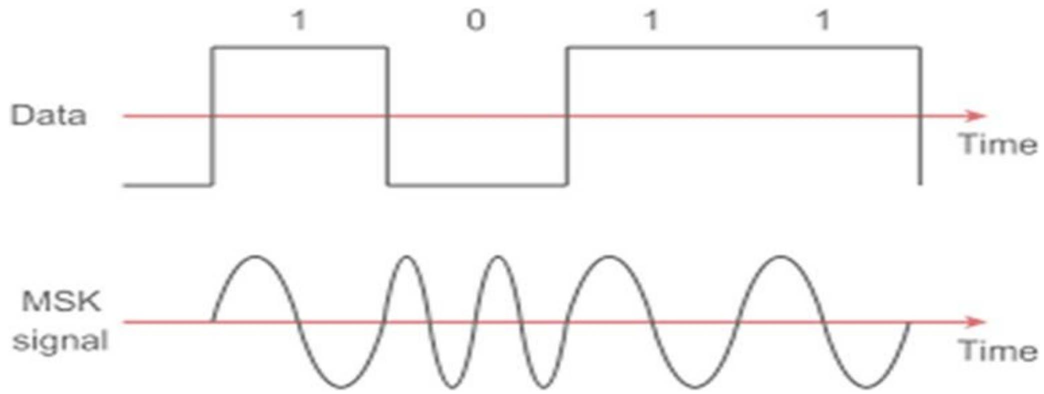

## 5. MSK (Minimum Shift Keying): -

- MSK, Minimum Shift Keying is a form of continuous phase frequency shift keying, providing spectrum efficiency & enabling efficient RF power amplifier operation.
- Minimum shift keying, MSK, is a form frequency modulation based on a system called continuous-phase frequency-shift keying.




### 6. <u>GMSK (Gaussian Minimum Shift Keying): -</u>

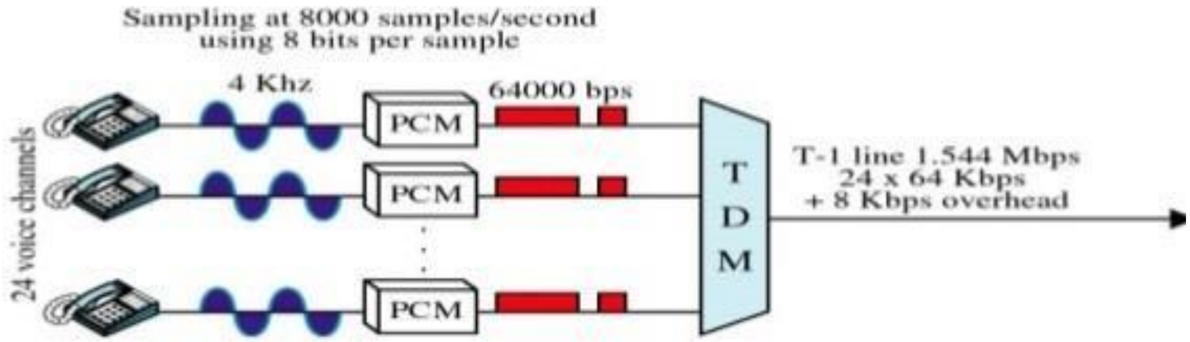
- Gaussian Minimum Shift Keying (GMSK) is a form of modulation based on frequency shift keying that has no phase discontinuities and provides efficient use of spectrum as well as enabling high efficiency radio power amplifiers.




## 6.5 Working of T1-Carrier system

## <u>Working of T1-Carrier system: -</u>

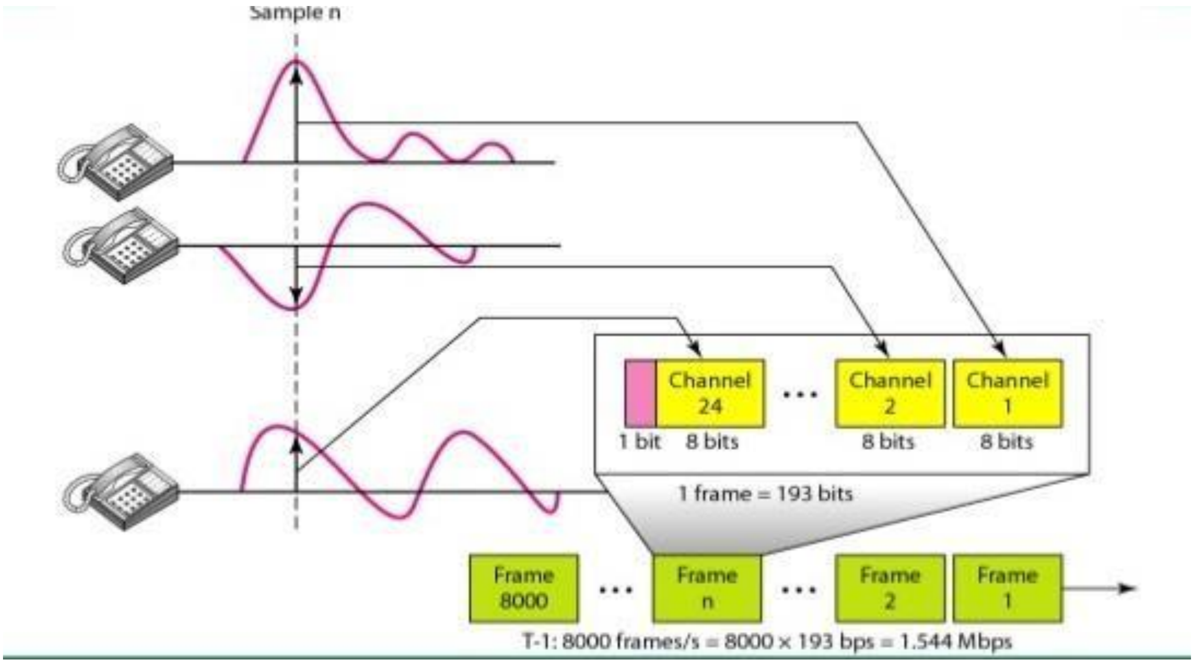
- T1 digital carrier system is a North American digital multiplexing standard since 1963. T1 stands for transmission one and specifies a digital carrier system using PCM encoded analog signal.
- A T1 carrier system is time division multiplexes PCM encoded samples from 24 voice band channels for transmission over a single metallic wire pair or optical fiber transmission line. Each voice band channel has BW around 300Hz to 3000KHz.



- A multiplexer is simply a digital switch with 24 independent inputs and onetime division multiplexed output. The PCM output signals from 24 voice band channels are sequentially selected and connected through the multiplexer to the transmission line.
- With T1 carrier system, there is sampling, encoding and multiplexing of 24 voice band channels. Each channel contains an 8-bit PCM code and sampled 8000 times a second. Each channel is sampled at same rate but not at same time.
- The figure shows that, each channel is sampled once in each frame, but not at same time. Each channel's sample is offset from previous channel's sample by 1/24 of total frame time.
- Therefore, one 64Kbps PCM encoded sample is transmitted for each voice band channel during each frame. The line Speed is calculated as:

$$\frac{24\ Channels}{frame} \times \frac{8Bits}{Channel} = 192\text{bits per frame}$$

$$Thus\ \frac{192\ bits}{frame} \times \frac{8000\ frames}{second} = 1.536Mbps$$




- An additional bit (called framing bit) is added to each frame. The framing bit occurs once per frame (8000bps rate) and recovered in receiver, where it is used to maintain frame and sample synchronization between TDM transmitter and receiver.
- So, each frame contains 193 bits and line speed for T1 digital carrier system is

$$\frac{193\ bits}{frame} \times \frac{8000\ frames}{second} = 1.544Mbps$$

## 6.6 Spread Spectrum & its applications

**Spread Spectrum: -**

In telecommunication and radio communication, spread-spectrum techniques are methods by which a signal (e.g., an electrical, electromagnetic signal) generated with a particular bandwidth is deliberately spread in the frequency domain, resulting in a signal with a wider bandwidth.

**Its applications: -**

**CDMA radios:** It is useful in multiple access communications wherein many users communicate over a shared channel.

**High Resolution Ranging:** SS Communications is often used in high resolution ranging.

**WLAN:** Wireless LAN (Local Area Networks) widely use spread spectrum communications.

## 6.7 Working operation of Spread Spectrum Modulation Techniques (DS-SS & FH-SS)

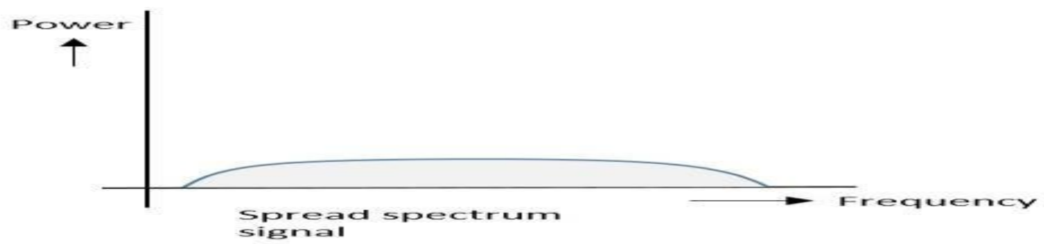
### Working operation of Spread Spectrum Modulation Techniques (DS-SS & FH-SS): -

### Spread Spectrum Modulation: -

- A collective class of signalling techniques are employed before transmitting a signal to provide a secure communication, known as the Spread Spectrum Modulation.
- The main advantage of spread spectrum communication technique is to prevent "interference" whether it is intentional or unintentional.

### Spread Spectrum Signals: -

- 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.
- ➢ Spread spectrum multiple access techniques are of two types.
  1) Frequency Hopped Spread Spectrum (FHSS)
  2) Direct Sequence Spread Spectrum (DSSS)

## 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 re-use.
- 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.

### 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.

## 6.8 Define bit, Baud, symbol & channel capacity formula. (Shannon Theorems)

### Shannon's Theorem: -

Shannon's Theorem gives an upper bound to the capacity of a link, in bits per second (bps), as a function of the available bandwidth and the signal-to-noise ratio of the link.

The Theorem can be stated as:

$$C = B * \log_2(1+ S/N)$$

where C is the achievable channel capacity, B is the bandwidth of the line, S is the average signal power and N is the average noise power.

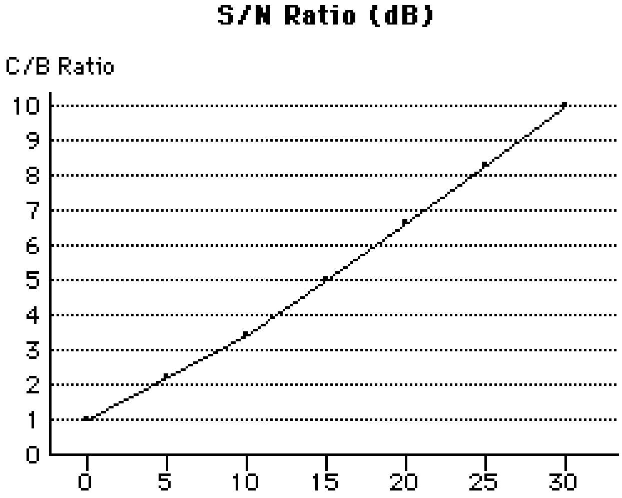
The signal-to-noise ratio (S/N) is usually expressed in decibels (dB) given by the formula:

$$10 \log_{10}(S/N)$$

so, for example a signal-to-noise ratio of 1000 is commonly expressed as **10 log10(1000) = 30 dB.**

Here is a graph showing the relationship between C/B and S/N (in dB):

| S/N Ratio (dB) | Units: C/B Ratio |
|---|---|
| 0 | 1.0 |
| 5 | 2.19 |
| 10 | 3.46 |
| 15 | 5.03 |
| 20 | 6.66 |
| 25 | 8.31 |
| 30 | 9.97 |




6.9

## Application of Different Modulation Schemes

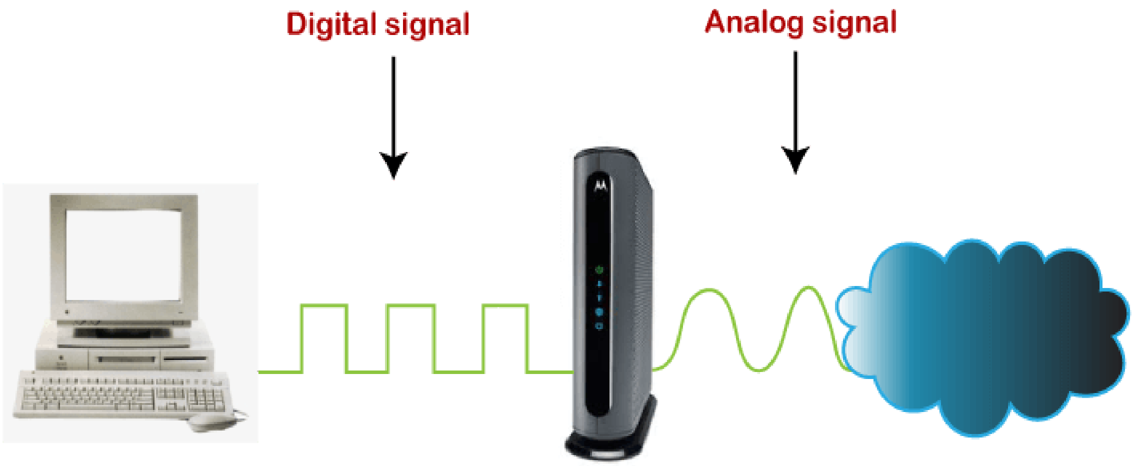
- Grouping various signals with similar frequencies to transmit them over similar bandwidths.
- Amplification of base frequencies to improve the quality of transmission.
- Convert digital signals to analog and vice versa.
- It is used in **Aircraft for communication** between station to pilot and vice versa.

## 6.10 Types of Modem & its application

Modem stands for modulator that converts analog signal at the senders end & it converts digital signal back to analog signals at the receiving end.

**Types of Modem**

- External Modem: Modem that are connected externally to a computer as other peripherals are connected.
- Internal Modem: Modem that are fixed within a computer.
- Wireless Modem. It bypass the telephone system & connect directly to a wireless network through it can directly access the internet connectivity provided by an Internet service provider (ISP).
- Dial-up Modem. Dial up connections use moderns to decode audio signals into data to send to a router or computer & to encode signals from the latter two devices to send to another modem.
- Cable Modem. A cable modem is a type of network bridge that provides bi-directional data communication via radio frequency channels on a hybrid fibre-co axial (HFC) radio frequency over glass ($RF_0G$) & co-axial cable infrastructure.
- DSL Modem. There are three types of DSL connections. SDSL,VDSL & ADSL. A modem is a small box that connects your device to the internet using cables.
- Satellite Modem. A satellite modem or sat modem used to establish data transfers using a communications satellite as a relay. Its main function is to transform an input bitstream to a radio signal & vice versa. These devices are used in satellite internet access.
- Half-duplex Modem. Half duplex devices can only transmit in one direction at one time. In this case data can move in two directions, but not at the same time. The word Duplex refers to the capability to send & receive data.




It is used in data transmission, remote management, broadband back up, point of sale etc.. It is used to send a voice massage to a telephone/mobile when an alarm is set off, Cell phone tower maintenance on circuits etc.

## Short Type Questions with Answers

1. **Define frequency division Multiplexing. (W 2020)**

Ans-It is a technique by which total BW available in communication system is divided into series of non-overlapping frequency bands.

2. **Define Spread Spectrum.**

**Ans:** In telecommunication and radio communication, spread-spectrum techniques are methods by which a signal (e.g., an electrical, electromagnetic signal) generated with a particular bandwidth is deliberately spread in the frequency domain, resulting in a signal with a wider bandwidth.

**3. Define GMSK.**

- **Ans:** Gaussian Minimum Shift Keying (GMSK) is a form of modulation based on frequency shift keying that has no phase discontinuities and provides efficient use of spectrum as well as enabling high efficiency radio power amplifiers.

**4.Define Spread Spectrum Modulation.**

**Ans: Spread Spectrum Modulation:**

- A collective class of signalling techniques are employed before transmitting a signal to provide a secure communication, known as the Spread Spectrum Modulation.

**5. Define Time Division Multiplexing.**

- **Ans:** Time Division Multiplexing is a technique in which multiple signals are combined and transmitted one after another on the same communication channel.

**6. What is Modem & give its application. (W-2020)**

Ans- Modem stands for modulator that converts analog signal at the sender's end & it converts digital signal back to analog signals at the receiving end. It is used in data transmission, remote management, broadband back up, point of sale etc.. It is used to send a voice massage to a telephone/mobile when an alarm is set off, Cell phone tower maintenance on circuits etc.

## Possible Long Type Questions

1 **Describe the modulation & demodulation process of ASK in detail. (W-2020)**

**2. Explain the Working of T1-Carrier system with Block Diagram.**

**3. Explain the Working operation of Spread Spectrum Modulation Techniques (DS-SS & FH-SS).**

**4. Define Digital modulation techniques & classify it with figures.**

## REFERENCE :

**Table:**

| Sl.No. | Name of the Book/Source | Name of the Author | Name of the Publication |
|---|---|---|---|
| 01 | Communication systems(Analog &Digital) | By Sanjay Sharma | KATSON |
| 02 | Communication system | By V. Chandrasekhar | OX FORD |
| 03 | Advanced Communication | By Thomasi | PHI |