As per Revised Syllabus 2009

BASICS OF ANALOG AND DIGITAL COMMUNICATION SYSTEM

EXTC

SEMESTER IV - ELECTRONICS ENGINEERING

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Basics of Analog and Digital Communication System

By Priyanka Bhor, Vicky Jain and Yogesh Kothari

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Dedicated To

Our Parents and Friends

Acknowledgement

We would like to thank Prof. Kiran Talele (Professor of Sardar Patel College of Engineering). He was the one who started building our self confidence right from Sem. I of our engineering career by his highly skillful teaching. And at the end, it was our self confidence that made us complete this book.

- Authors

How to Make the Best Use of this Book

Every chapter in this book consists of three main parts

(i) The Theory Part

(ii) The Frequently Asked Questions (FAQ's) Part

(iii) The Problem Part

(i) The Theory Part

This part contains the theoretical knowledge about the chapter. Always read this part first.

(ii) The FAQ's Part

This part contains those questions which are practical. Most of your viva questions will be from this part only. But, keep in mind that before reading the FAQ's you should be aware of the theory. So, always read it at the end, when you are done with your theory.

(iii) The Problems Part

It contains problems. Practice it when you are done with the theory.

> ***A Tip :*** *You can read the chapters in any order, but we would suggest an order so that it becomes easier for you. Read the chapters in the following order.*
>
> *Chapter 1*
> ↓
> *Chapters 3, 4, 5, 6, 7*
> ↓
> *Chapter 2*
> ↓
> *Chapters 8, 9, 10, 11,12*

- Priyanka Bhor, Vicky Jain and Yogesh Kothari

CONTENTS

Syllabus

S.E. Electronics Semester-IV
Basic of Analog and Digital Communication System

Period per week	Lecture	4	
	Practical	2	
	Tutorial	---	
		Hours	Marks
Evaluation System	Theory Examination	3	100
	Practical	---	---
	Oral Examination	---	25
	Term Work	---	25
	TOTAL	---	150

	Detailed Syllabus	Lectures /Week
1.	**Elements of Communication System :** Basic block diagram of communication system, Modulation and Demodulation concept, channels noise in communication system, signal-to-noise ratio, noise factor and noise figure, equivalent noise temperature. Electromagnetic waves propagation : Propagation terms and definitions. ***(Refer chapter 1,2 and 12)***	06
2.	**Amplitude Modulation** Principles of DSB full carrier AM, envelope detector, practical diode detector. Different types of AM : DSB-SC, SSB-SC, VSB, ISB. ***(Refer chapter 3, 4 and 6)***	12
3.	**Angle Modulation** Principles of frequency modulation and phase modulation. FM modulators, types of FM : NBFM and WBFM, FM transmitter, noise triangle, pre-emphasis and de-emphasis circuits. FM detection: frequency discriminator and phase discriminator. ***(Refer chapter 5 and 6)***	12

4.	**Radio Receivers** Receivers characteristics TRF receivers, and super heterodyne. Receivers : Choice of IF, AGC, AFC in AM and FM receivers. ***(Refer chapter 7)***	07
5.	**Analog Pulse Modulation** Sampling theorem for low pass signals, Aliasing error, Sampling techniques, principles, generation, demodulation and spectrum of PAM,PWM,PP. ***(Refer chapter 8)***	07
6.	**Digital Pulse Modulation** Comparison of digital signal transmission over analog signal transmission, significance of regenerative repeaters. Pulse-coded modulation (PCM) : sampling, quantizing, encoding technique, PCM bandwidth, Necessity of companding, PCM waveform formats : Uni-polar and polar NRZ, RZ, AMI Delta modulation (DM), adaptive delta modulation (ADM). Multiplexing : TDM, FDM-principles and applications. ***(Refer chapter 9,10 and 11)***	12

Term Work : The term work shall consists of at least eight laboratory experiment covering the whole of syllabus, duly recorded and graded. This will carry a weightage of fifteen marks. A test shall be conducted and will carry a weightage of ten marks.

Suggested List of Experiments

(1) Amplitude modulation and demodulation

(2) DSB-SC and SSB-SC modulation, demodulation

(3) Frequency modulation and demodulation

(4) Study of Superhetrodyne receiver characteristics

(5) Sampling and reconstruction of sampled signals

(6) Pulse modulation (PAM,PWM,PPM)

(7) Delta modulation

(8) Time division multiplexing of PCM signals

(9) Line codes (NRZ, RZ, AMI-RZ)

(10) Simulation on AM, FM/Multiplexing

1 Elements of Communication System

Topic	Theory imp	Oral imp
Communication		
Signal		☆
Block diagram	☆☆	
Modulation	☆	☆☆
Need for Modulation	☆☆	☆☆☆☆
FAQ's		☆☆☆
Problems		

1.0 Communication

Communication means transfer of information from one entity to other entity through a medium. Entity is a real world object e.g. Human being, Computer, Cell phones etc. From our subject point of view it means sending, receiving and processing of information by electronic medium.

Timeline

It shows the evolution of wireless communication.

Year 1844		Year 1906		Year 1947
Telegraph line	⇨	Invention of triode 1st phase wireless	⇨	Invention of transistor 2nd phase wireless

1.1 Signal

Signal is some way in which information is represented.

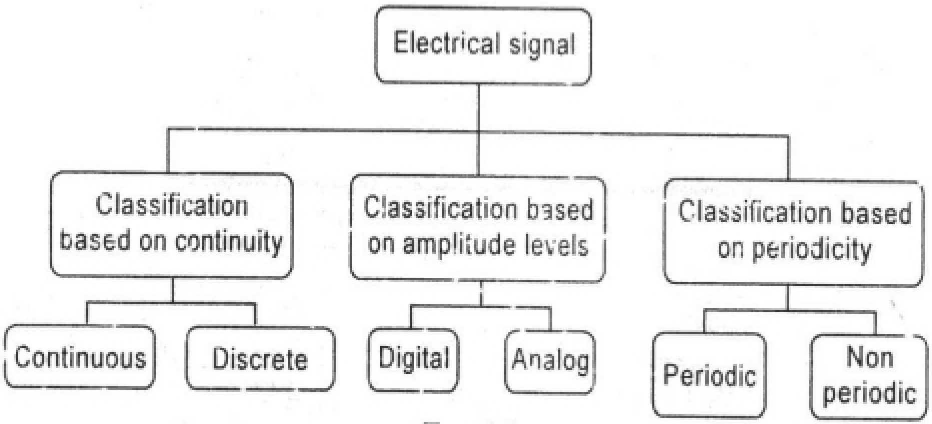
E.g. when a cricket umpire signals and raises his hand with a finger pointing towards sky, it represents the information that the batsman is out.

Our main concern here is an electrical signal i.e. it has information in the form of electrical energy.

Technical Definition of Signal

Signal is a function of dependent or independent variables which varies with any physical quantity, like time, frequency etc.

Classification of Electrical Signal



Tree 1.1

All signals in real life are non periodic, continuous and analog. Some examples of signals are as shown in *figure 1.1*.

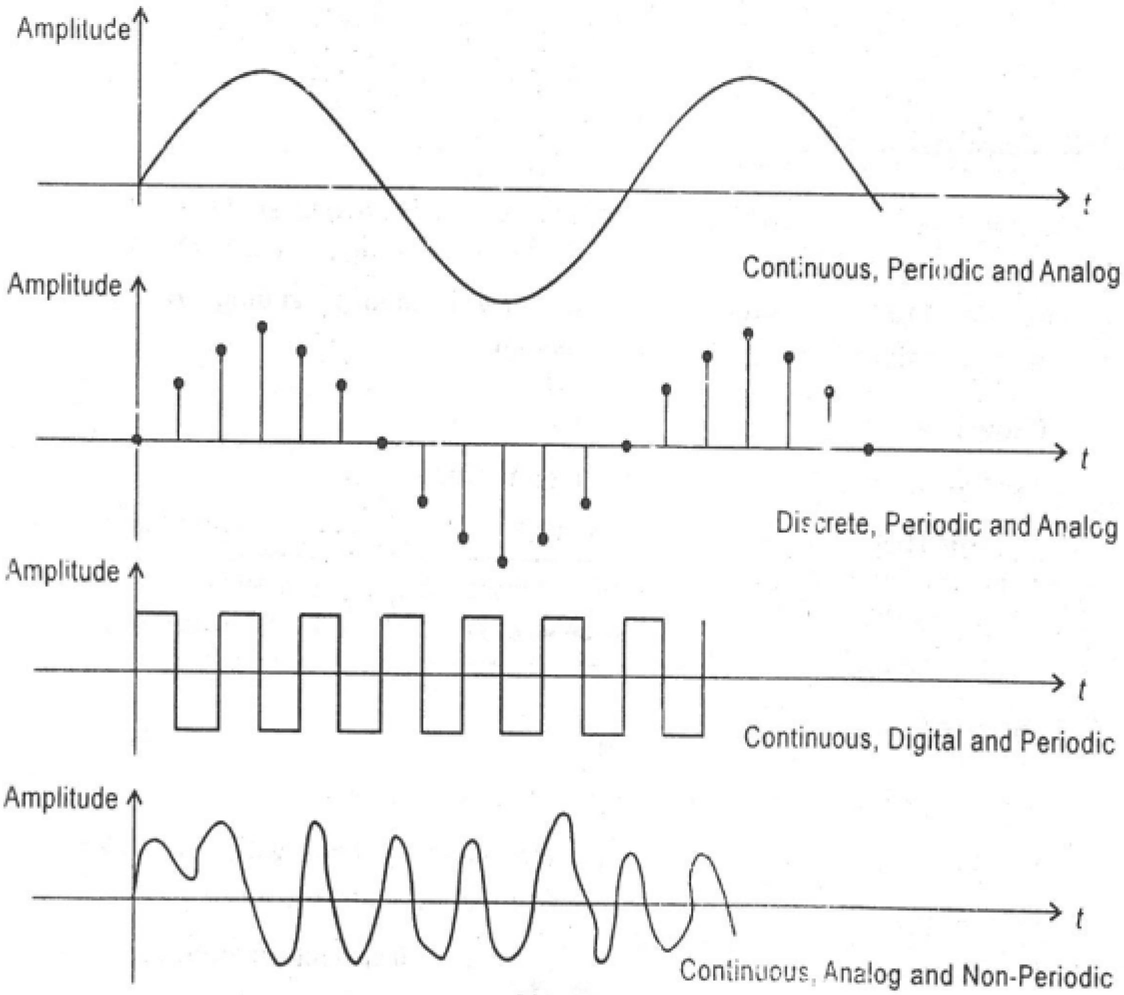


Fig. 1.1

1.2 Basic Block Diagram

Q. Draw the block diagram of communication systems and explain each and every block.

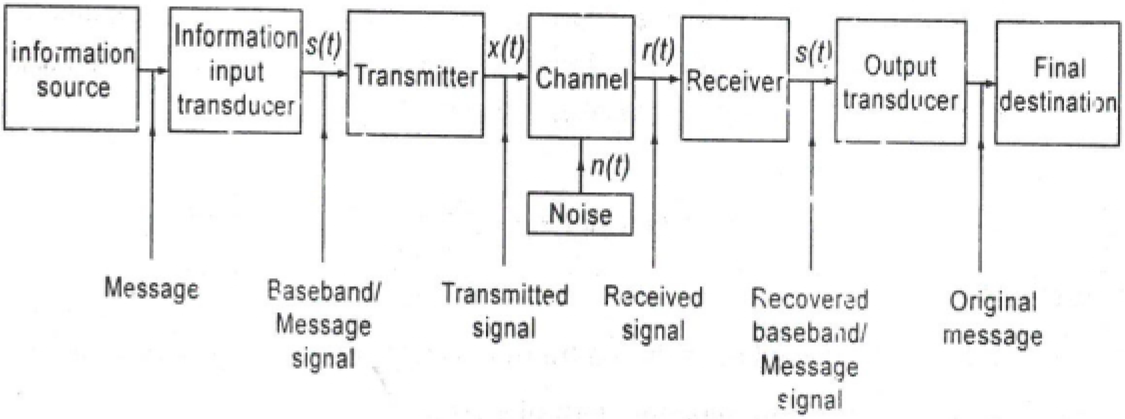


Fig. 1.2

(a) Information Source

- Information in broader sense is meaningful data.
- It is not a physical quantity.
- We cannot measure it.
- Physical form of information is a signal.
- A generator of information is information source.
- E.g. a sentence spoken by man "I want to learn P.Com."

  Source - man

  Information - I want to learn P.Com.

  Physical form - sound signal.

(b) Input Transducer

- Transducer converts one form of energy to other form

  E.g. Microphone converts sound energy into electrical energy i.e. Sound signal to electrical signal.
- In communication systems usually target form of energy is electrical energy i.e. electrical signal.
- The electrical signal received at this stage is called *baseband signal* denoted by $s(t)$.

(c) Transmitter

- This block is responsible for modulation (processing) of the baseband signal.
- It is mainly responsible for either of the following two operations

(1) Transmit signal as it is.

(2) Modulate the signal and then transmit.

- What is modulation ?

  It is basically changing the characteristics or parameters (i.e. amplitude, frequency or phase) *of a high frequency* (carrier signal) *signal on the basis of a low frequency signal* (baseband signal).

  Modulated signal is $x(t)$.

  (Modulation is explained in detail in the later part of this chapter.)

(d) Channel

- The transmitting medium between transmitter and receiver is called a *channel*.
- The channel is a very important part of system.
- It adds many constraints to the system.
- Noise signal is also added to signal in the channel.
- Noise is one of the main characteristics of channel.

  E.g. of channel :

  Transmission lines or wires,

  Optical fibers,

  Atmosphere.

(e) Noise

- It is unwanted signal of random and unpredictable nature.
- Does not contain any useful information.
- Denoted by $n(t)$.

  (A separate chapter on noise will give you full explanation)

(f) Receiver

- Function is to generate the original signal from the received signal.
- Received signal is $r(t)$.

  $r(t) = x(t) + n(t)$.

- So we have to remove the noise to recover the $x(t)$.
- At transmitter end, we modulate the signal so, we have to demodulate the signal at receiver i.e.

  Get back $s(t)$ from $x(t)$

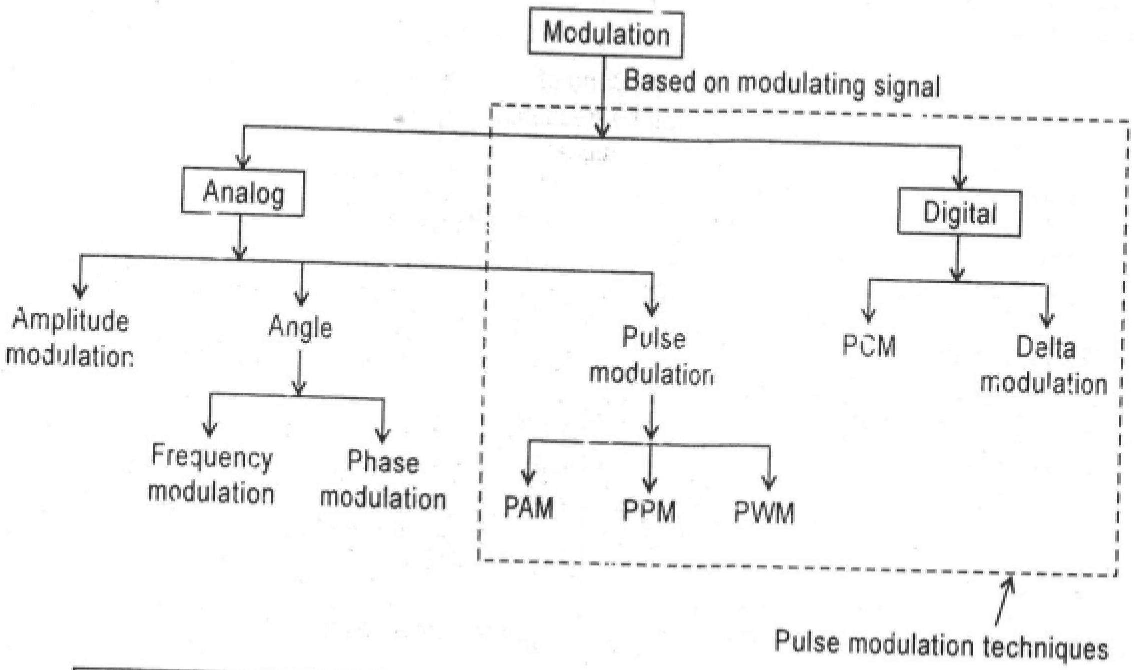
(g) Output Transducer

- It converts the electrical signal to required form.

1.3 Modulation

Q. What is modulation ?

- *It is basically changing the characteristics or parameters* (i.e. amplitude, frequency or phase) *of a high frequency* (carrier signal) signal *on the basis of a low frequency signal* (baseband signal).
- Two signals are involved :
  (1) Carrier signal
  (2) Modulating or the base band signal
- **Types of Modulation :** The following hierarchy shows different types of modulation.



This book covers the explanation and study of the entire above heirarchy

Tree 1.2

1.4 Need for Modulation

Q. Explain the need of modulation and what are different types of modulation techniques ?

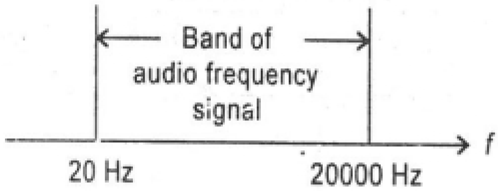
Note : *Write about types of modulation from previous section*

Modulation increases the cost and complexity of the system. This is the main disadvantage of using modulation. Then too, modulation is extensively used because if modulation is not used and the base band signal is transmitted as it is, then the following problems can be encountered.

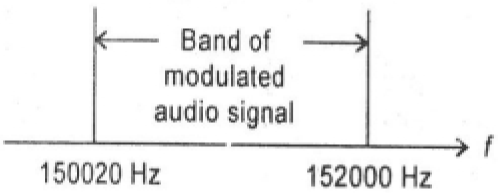
- Antenna height.
- Narrow banding.
- Poor radiation and penetration.
- Diffraction angle.
- Multiplexing.

To study the need for modulation let us recollect what modulation exactly is with a practical example.

Lets take the normal audio signal which consists of frequencies from 20 Hz to 20 kHz. Thus, the lowest frequency is 20 Hz and highest frequency is 20 kHz. The frequency band of the audio signal is



Now, if this audio signal modulates a carrier of 1.5 MHz then the band of the modulated audio signal is as follows.



This is how baseband signal of a low frequency spectrum is translated to a high frequency spectrum.

Let us discuss the problem's which may be encountered if modulation is not used in a system and how modulation solves those problems.

1.4.1 Antenna Height

According to a study on antennas, for proper transmission and reception of radio waves, the height of antenna should be the order of one fourth or half the wavelength of the frequency of the transmitted signal.

$$\text{i.e. } h = \frac{\lambda}{4} \text{ or } h = \frac{\lambda}{2}$$

where h : Height of antenna

λ : Wavelength of the transmitted signal

$$\text{Also } \lambda = \frac{c}{f}$$

$$\therefore \quad \boxed{h = \frac{c}{4f}} \qquad \left(\text{by considering } h = \frac{\lambda}{4} \text{ (for e.g.)}\right)$$

Without Modulation :

$$\text{Lowest frequency} = 20 \text{ Hz}$$

$$\text{Highest frequency} = 20000 \text{ Hz}$$

$$\therefore \text{ Height } h_L = \frac{3 \times 10^8}{4 \times 20} = 3750 \text{ km} \quad (\because c = 3 \times 10^8 \text{ m/s})$$

$$\text{and } h_H = \frac{3 \times 10^8}{4 \times 20 \times 10^3} = 3.75 \text{ km}$$

h_L : Height corresponding to lowest frequency

h_H : Height corresponding to highest frequency

Thus without modulation the antenna height should lie between 3.75 km to 3750 km which is practically not feasible.

With Modulation :

$$\text{Lowest frequency} = 1500020 \text{ Hz}$$

$$\text{Highest frequency} = 1520000 \text{ Hz}$$

$$\therefore h_L = 49.999333 \text{ m}$$

$$\text{and } h_H = 49.342 \text{ m}$$

$$\text{thus } h_L \approx h_H \approx 50 \text{ m}$$

Thus we need an antenna of height 50 m which is practically available.

1.4.2 Narrow Banding

We know that in radio communication systems, the frequency of transmitted signal decides the height of the antenna. Thus, if an audio signal (20 Hz to 20 kHz) is to be transmitted, then the system requires as many antennas as the number of frequency components. The solution to this problem is the analysis of *band-edge*

> **Note :** *Band edge ratio is defined as* ***the ratio of the lowest and highest frequencies of the transmitting frequency band.***
>
> *If the band edge ratio is 1 : 1 or (≈1) then a single antenna can be used to transmit the entire range of frequencies.*

Without Modulation :

$$\text{Lowest frequency} = 20 \text{ Hz}$$

$$\text{Highest frequency} = 20000 \text{ Hz}$$

$$\therefore \text{ band edge ratio} = \frac{20}{20000} = 0.0001$$

Thus transmitting the entire range (20 Hz → 20 kHz) is not possible using a single antenna.

With Modulation :

$$\text{Band edge ratio} = \frac{1500020}{1520000} \approx 1$$

Thus, a single antenna can be used to transmit the entire range of frequencies. Thus a wide frequency band, ranging from, 20 Hz to 20 kHz is practically narrowed so that the ratio of the lowest to highest frequency components is 1 : 1 using modulation. This process is called *Narrow Banding*.

1.4.3 Poor Radiation and Penetration

According to the electromagnetic theory, the radiated power of a signal is directly proportional to the frequency of signal.

i.e. $\boxed{\text{Power} \propto \text{Frequency}}$

Without Modulation :

Since frequency is 20 Hz-20 kHz, radiated power is relatively less.

With Modulation :

Since frequency of the signal is high (1500020 Hz to 1520000 Hz), the radiated power is relatively very high.

Note : *If the power is less, the level of penetration is also less.*

1.4.4 Diffraction Angle

In some radio communication systems, such as microwave links and RADAR systems, radio waves are radiated in the form of narrow beams. When, the narrow beam travels through the channel, it may strike some obstacles resulting in diffraction and the beam spreads.

Also, it is proved that the spreading of a beam is directly proportional to the diffraction angle.

And, the diffraction angle is inversely proportional to the frequency of the signal.

$$\boxed{\text{Diffraction angle} \propto \frac{1}{f}}$$

Without Modulation :

Thus, for low frequencies diffraction angle will be high and thus, signal will spread more and hence power will reduce.

With Modulation :

Thus, because of modulation the diffraction angle will be low thus, spreading of signal will reduce.

1.4.5 Multiplexing

Multiplexing is the technique that is most widely used in nearly all types of communication systems. It facilitates the simultaneous transmission of multiple messages over a single transmission channel.

To understand multiplexing, consider the following example.

Consider two audio signals from two senders A and B. These two signals are to be received by C and D respectively, and these signals are transmitted over the same channel.

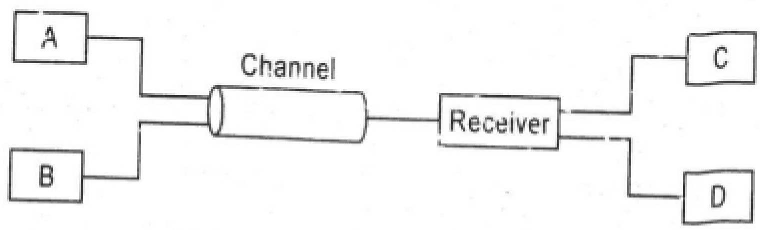


Fig. 1.3

Now, both the audio signals have the same range of frequencies (20 Hz to 20 kHz). Thus, there is a possibility that some frequency components may overlap and also at the receiver end there is no way to separate the two signals.

Now, suppose that the speech signal from A is used to modulate a carrier of 1.5 MHz. Thus, the range of frequency for the modulated signal will be (1500020 to 1520000 Hz), and the speech signal from B is used to modulate a carrier of 2.0 MHz then the range of frequencies will be 2000020 to 2020000 Hz.

Thus, if these signals are transmitted through a single channel, there is no chance of the signals getting overlapped and also the signals can be separated at the receiver end.

This is multiplexing which is possible only through modulation.

Note : *Multiplexing is explained in detail in chapter 10*

1.5 Frequently Asked Questions

Q.1. What is trunking ?

Ans. Traditional radio equipment works because all parties involved in the communication agree on what frequencies they will utilize. Traditional radio scanners work by scanning for and then listening to those frequencies.

Trunking radios, on the other hand, constantly re-negotiate the frequencies utilized for the conversation. This allows for more efficient utilization of limited frequencies because each conversation does not require a dedicated channel.

Q.2. What radio frequencies are used for what purposes ?

Ans. The Federal Communications Commission (FCC) licenses the frequency spectrum in the United States.
Useful charts showing frequency allocations are available from the National Telecommunications and Information Administration (NTIA)
at http://www.ntia.doc.gov/osmhome/allochrt.html.

This table shows commonly used names for various frequency ranges :

Frequency	Description
30 - 300 GHz	Extremely High Frequency
3 - 30 GHz	Super High Frequency
300 MHz - 3 GHz	Ultra High Frequency (UHF)
30 - 300 MHz	Very High Frequency (VHF)
3 - 30 MHz	High Frequency (HF)
300 kHz - 3 MHz	Medium Frequency
30 kHz - 300 kHz	Low Frequency (LF)
3 - 30 kHz	Very Low Frequency (VLF)
300 Hz - 3 kHz	Voice Frequency
Below 300 Hz	Extremely Low Frequency

Table 1.1

Q.3. What is crystal radio ?

Ans. The crystal radio is a simple radio receiver that can be made from a few easy-to-obtain and inexpensive parts. It is unique for this type of radio that it does not require a battery pack, has no moving parts, and can be built using ordinary *household* materials, yet it actually works. The only power it receives comes from the radio signals.

Q.4. *What is intermodulation ?*

Ans. Intermodulation is a *special case where two (or more) sinusoids effect one another to produce undesired products*, i.e. *Unwanted Frequencies*. Again, this can only occur when both waves share the same nonlinear device.

Q.5. *What is linear device ?*

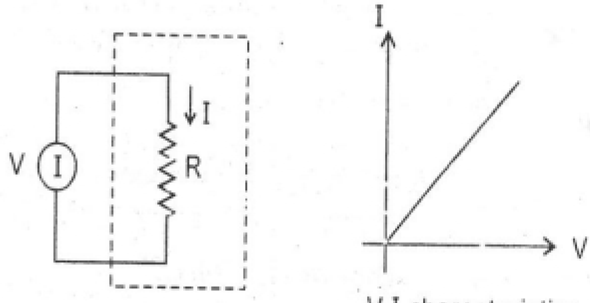
Ans. A device for which the output within a given dynamic range, is linearly proportional to the input, is a *linear device*.

e.g. a resistor

Here $V = IR$

where V : Input

I : Output and they are linearly proportional



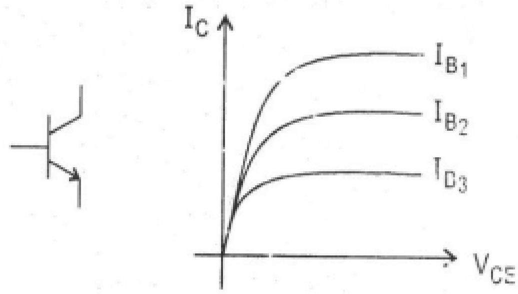
V-I characteristics of resistor

Fig. 1.4

Q.6. *What is nonlinear device ?*

Ans. An active device for which the output is not linearly proportional to the input is a non-linear device.

e.g. a transistor



V-I characteristics of a transistor in common emitter mode

Fig. 1.5

Q.7. *What is cross modulation ?*

Ans. If you have ever been listening to a distant FM station while driving by an AM Broadcast station's transmitting tower; you can hear both stations - one on top of the other. That effect is "Cross Modulation," a form of Inter-modulation.

Q.8. *What is bandwidth ?*

Ans. In electronic communication, bandwidth is the width of the range (or *band*) of frequencies that an electrical *signal* uses on a given transmission medium. In this usage, bandwidth is expressed in terms of the difference between the highest-frequency signal component and the lowest-frequency signal component. Since the *frequency* of a signal is measured in *Hertz* (the number of cycles made per second), a given bandwidth is the difference in Hertz between the highest frequency the signal uses and the lowest frequency it uses. A typical voice signal has a bandwidth of approximately three kilohertz (3 kHz); an analog television (TV) broadcast video signal has a bandwidth of six megahertz (6 MHz) ~ some 2,000 times as wide as the voice signal.

Q.9. How is information related to bandwidth ?

Ans. Hartley's law specifies the relation between information and bandwidth of a communication system. Another factor that is involved with this law is the time taken by the transmitted signal to reach the receiver i.e. the transmission time. The law is mathematically expressed as

$$\boxed{\text{Information} \propto (\text{Bandwidth} \times \text{Transmission time})}$$

Q.10. What is stray capacitance ?

Ans. The stray capacitance appears between two closely placed wires in a circuit, i.e. these two wires together act as a *Capacitor*. At low frequency, this capacitance is very high and acts as open circuit without affecting the system but at high frequencies this capacitance is low and acts as short circuit affecting the system.

❋❋❋

2 NOISE

Topic	Theory imp	Oral imp
Types/Sources of Noise	☆☆	☆
Noise Voltage/Thermal Noise	☆☆	
SNR	☆☆☆	☆☆
Noise Power	☆☆	
Noise Factor	☆☆☆	☆
Equivalent Noise Temperature	☆☆☆	☆
Equivalent Noise Resistance	☆	☆
Lossy Components		
Cascaded Networks	☆	
FAQ's	☆☆	☆☆☆
Problems	☆☆☆	

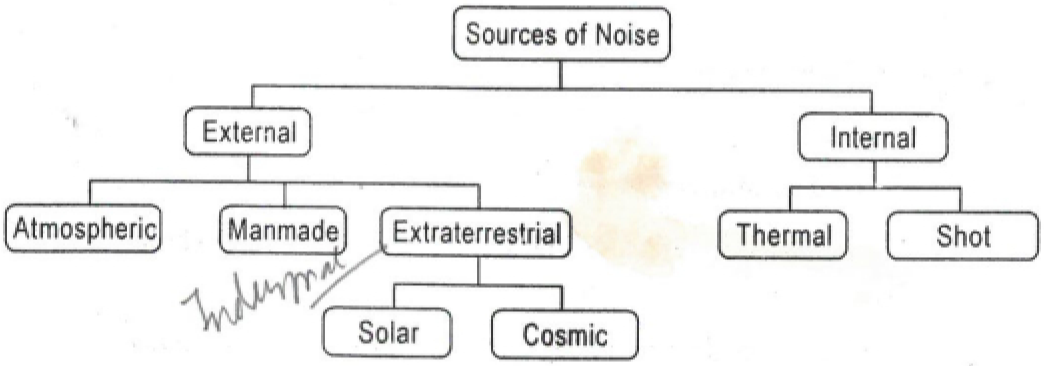
2.1 Noise

- It is unwanted signal of random and unpredictable nature.
- Does not contain any useful information
- It is not only added during the transmission of the signal, but also during the processing of the signal i.e. noise is added to the signal at every stage of the communication system.

2.2 Sources of Noise

Q.1. Classify and explain different types of noise affecting communication.

Q.2. Give sources/causes of all types of internal noise.



Tree 2.1

(a) External Sources

- Introduced by the transmission medium or the channel.
- Mainly divided into three types
  - Atmospheric source
    - Also called *static* or *natural noise.*
    - Main sources are lightning discharges and natural electrical disturbances.
  - Manmade noise
    - Also called *industrial noise.*
    - Main sources are automobiles, aircraft engines, motors, fluorescent lights.
  - Extraterrestrial noise
    - Also called as *space noise.*
    - It comes from outer space.
    - Solar noise comes from sun.
    - Cosmic noise comes from stars.

(b) Internal Sources

- Introduced by the internal circuits of the system like transmitter or receiver circuit.
- Mainly classified into two types
  - Thermal noise
    - Also called *Johnson's* or *white noise.*
    - Caused due to the random motion of electrons and molecules of the resistors used in the circuit.
    - Contributes to maximum portion of the total noise present in the system.

  (Covered in detail later on)

  - Shot noise
    - Also called as *transistor noise.*
    - It is generated by random motion of electrons and wholes inside the transistor.
    - It is directly proportional to output current and inversely proportional to transconductance.

2.3 Thermal Noise Voltage / Noise Voltage

Q. *What causes Thermal noise ? Why is it also called white noise ?*

Thermal noise tends to be dominant in most systems, so we will concentrate on this. Consider a resistor with resistance R at a temperature T (in Kelvin). The kinetic energy of the electrons is proportional to T. The random motion of the electrons create voltage fluctuations at the resistor terminals. The voltage has zero average, but the RMS value of the voltage is given by Planck's blackbody radiation equation,

$$\overline{v_n} = \sqrt{\frac{4hfBR}{e^{hf/k_BT} - 1}}$$

where, B = Bandwidth in Hertz

h = Planck's constant = 6.546×10^{-34} J-sec

k_B = Boltzmann's constant = 1.380×10^{-23} J/K

f = Frequency (Hz)

If the frequency is large, say $f = 100$ GHz, and the temperature is low, so that $T = 100\,°K$, then

$$hf = 6.5 \times 10^{-23} << k_BT = 1.38 \times 10^{-21}$$

This means that the exponent hf/k_BT is very small. The inequality gets even larger for higher frequencies at room temperature ($T = 273\,°K$).

Because of this, at higher frequencies the exponential can be approximated by the first two terms of the Taylor series,

$$e^{hf/k_BT} \approx 1 + \frac{hf}{k_BT}$$

This simplifies the RMS voltage to

$$\overline{v_n} \approx \sqrt{\frac{4hfBR}{1 + hf/k_BT - 1}} = \sqrt{4k_BTBR}$$

In this approximation, $\overline{v_n}$ is independent of frequency. For this reason, the thermal noise signal is called *white noise*.

Note : *Taylor's series for* e^x $e^x = 1 + x + x^2 + x^3 + \ldots\ldots\ldots$

2.4 Signal to Noise Ratio (SNR)

The SNR is the ratio of the signal power to the noise power at any point of time.

Noise is added to the signal as soon as it is transmitted and it keeps on getting added. Thus the actual signal at any point is

$$r(t) = \underbrace{x(t)}_{\text{Original signal}} + \underbrace{n(t)}_{\text{Noise}}$$

$$\therefore \boxed{\text{SNR} = \frac{\text{Power of } x(t)}{\text{Power of } n(t)} = \frac{\text{Signal power}}{\text{Noise power}}}$$

2.5 Noise Power

We can replace any noisy (warm) resistor with a Thevenin's equivalent noise source and an ideal noiseless resistor (*Figure 2.1*). If we connect this equivalent circuit to a bandpass filter with bandwidth B Hz and then to a second ideal resistor R (where the resistance of the load is chosen for maximum power transfer), the noise power delivered to the load is given by maximum power transfer theorem as :

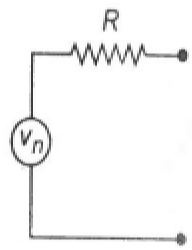


Fig. 2.1 : Equivalent circuit of a noise source.

$$P_n = \left(\frac{\overline{v_n}}{2}\right)^2 \frac{1}{R} = \frac{\overline{v_n}^2}{4R} \quad \text{where } \overline{v_n} = \sqrt{4k_B TBR}$$

> **Note :** *We do not have another of two in the denominator since the voltage is already an R.M.S. quantity.*

Using our expression for $\overline{v_n}$, we get

$$\boxed{P_n = \frac{4k_B TBR}{4R} = k_B TB}$$

2.6 Noise Factor

Q.1. Explain : Noise factor and noise figure.

Q.2. Define noise figure. Derive the Friss formula for calculation of total noise figure for a two amplifiers connected in cascade.

> **Note :** *For derivation of Friss formula, refer section 2.10.*

A key measure of system performance is its signal-to-noise ratio (SNR) :

$$\text{SNR} = \frac{S}{N} = \frac{\text{Signal power}}{\text{Noise power}}$$

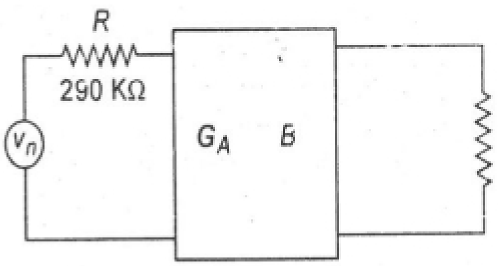
A high SNR means that it is easy to recognize the signal, and a low SNR means that the signal is obscured by noise. Ideal components do not add any noise, so the SNR at the output is the same as the SNR at the input. Non-ideal component in general will add some additional noise, so the output SNR is less than the input SNR.

Noise factor is a measure of the degradation in signal-to-noise ratio (SNR) as a signal passes through any component. The definition of noise factor (F) is *the ratio of the total available noise power at the amplifier output to the available noise power at the output due to the input noise.* It is given by :

$$F = \frac{\text{Output noise power}}{\text{Available noise power at the output due to input noise}}$$

$$F = \frac{\text{Output noise power}}{\text{Gain} \times \text{Input noise power}}$$

For an ideal component, $F = 1$.



Here,

$$G_A = \frac{S_o}{S_i}$$

S_o = Output signal power

S_i = Input signal power

Fig. 2.2 : Noisy amplifier.

It can be seen that noise factor is also equal to the ratio of the input SNR to the output SNR :

$$F = \frac{N_o}{N_i G_A} = \frac{N_o}{N_i S_o/S_i} = \frac{S_i/N_i}{S_o/N_o} = \frac{\text{SNR}_{in}}{\text{SNR}_{out}}$$

We can also write

$$F = \frac{G_A N_i + P_n}{G_A N_i} = 1 + \frac{P_n}{G_A N_i}$$

where, N_i = Input noise power to the circuit

P_n = Noise power added due to the component or circuit

G_A = Gain

Since noise factor is a dimensionless quantity, it is often expressed in dB.

> **Note :** *Noise Factor expressed in decibels is NOISE FIGURE*
>
> Noise figure, $F_{dB} = 10 \log_{10} F$

2.7 Equivalent Noise Temperature

Q.1. Explain : Noise temperature.

Q.2. Define Equivalent Noise Temperature. Derive the relation between noise factor and equivalent noise temperature.

We can also express the *noisiness* of a component in terms of an equivalent noise temperature using $P = k_B TB$

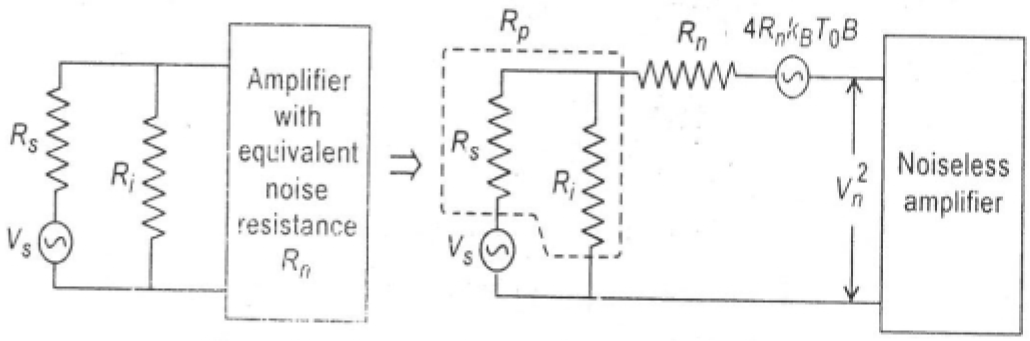
$$F = \frac{S_i/k_B T_0 B}{GS_i/Gk_B(T_0 + T_e)B} = 1 + \frac{T_e}{T_0}, \therefore T_e = (F-1)T_0$$

where, S_i = Input signal power

When specifying the equivalent temperature T_e of a component, we assume that the input noise power corresponds to room temperature, so that $T_0 = 290\ ^\circ K$. Equivalent temperature is most useful for low noise figure devices.

2.8 Equivalent Noise Resistance (R_n)

- The noise generated by the devices is represented by means of an imaginary resistance R_n.
- This resistance R_n is called *equivalent noise resistance.*
- The resistance R_n is assumed to generate noise at room temperature which is equal to the noise generated by the device.
- Therefore if R_n is the equivalent noise resistance then the device is assumed to be noiseless with a resistor R_n in series with it.



Here, R_s = Some resistance, V_s = Voltage source

Fig. 2.3

- The equivalent mean square thermal noise voltage at the input of the noiseless amplifier is given by :

$$v_n^2 = 4(R_n + R_i)\, k_B T_0 B$$

- For an ideal amplifier i.e. which does not produce noise, $R_n = 0$.
- Noise factor and noise resistance are related as

$$F = \frac{R_p + R_n}{R_p}$$

where, R_p = Parallel combination of R_s and R_i, or the antenna resistance.

2.9 Lossy Components

A lossy system component such as a length of lossy transmission line leads to a degradation in SNR. The basic principle for determining the noise figure of a lossy component is to realize that the noise power at the output of the component must be the same as the noise power at the input (thermal equilibrium), so that

$$GN_i + GN_{added} = N_i$$

where, G = Gain

N_i = Input noise

N_{added} = Added noise

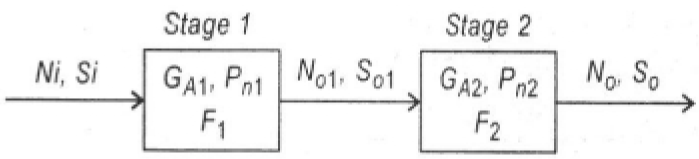
Solving for the equivalent additional power at the input gives $N_{added} = N_i(1 - G)/G$. The noise figure is then

$$F = 1 + \frac{P_n}{GN_i} = 1 + \frac{N_i(1-G)}{GN_i} = \frac{1}{G} = L$$

where L is the power loss of the device. Thus, the noise figure is the same as the loss.

2.10 Cascaded Networks

If we have two stages in a system,



$$N_o = G_{A2}N_{o1} + P_{n2} = G_{A2}(G_{A1}N_i + P_{n1}) + P_{n2}$$

$$F = \frac{G_{A2}(G_{A1}N_i + P_{n1}) + P_{n2}}{N_i G_{A1} G_{A2}} = 1 + \frac{P_{n1}}{N_i G_{A1}} + \frac{P_{n2}}{N_i G_{A1} G_{A2}}$$

Because the gain G_{A1} appears in the denominator of the second and third terms, the first stage in a system is most critical in obtaining a low noise system, if G_{A1} is large. In terms of the noise factors of the two stages,

$$F_1 = 1 + \frac{P_{n1}}{N_i G_{A1}}$$

$$F_2 = 1 + \frac{P_{n2}}{N_i G_{A2}}$$

the noise factor of the system is

$$F = F_1 + \frac{F_2 - 1}{G_{A1}}$$

The noise factor of the second state is divided by the gain of the first stage. Again, we can see that the first stage is most critical in deternining the noise factor of the system.

2.11 Frequently Asked Questions

Q.1. How noise figure and SNR are related ?

Ans. Refer section 2.6 for the definition of noise factor and just prove till

$$F = \frac{SNR_{in}}{SNR_{out}}$$

Then,

$$\text{Noise figure} = 10 \log_{10} F$$

$$\boxed{F_{dB} = 10 \log_{10} \frac{SNR_{in}}{SNR_{out}}}$$

Q.2. What is the significance of SNR ?

Ans. We know that

$$\boxed{SNR = \frac{\text{Signal power}}{\text{Noise power}}}$$

Thus depending on the value of SNR at any point of time we can estimate how much the signal is corrupted by noise at that point of time.

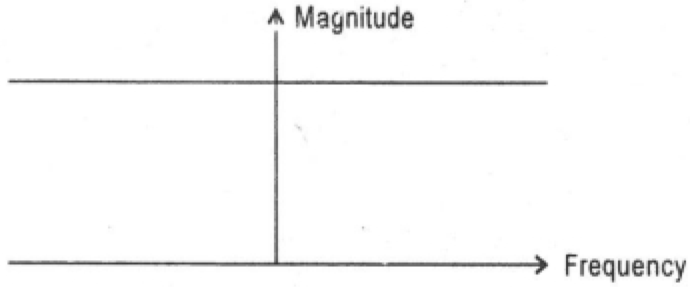
- If SNR is low it implies signal is corrupted to a greater extend.
- If SNR is high it implies less corruption.

Q.3. *Why the first stage in an amplifier is the most important in determining the noise figure for entire amplifier ?*

Ans. Write down the section "Cascaded network's" (Refer section 2.10).

Q.4. *Why is thermal noise called white noise ?*

Ans. The spectrum of thermal noise consists of all the frequencies with equal magnitude.

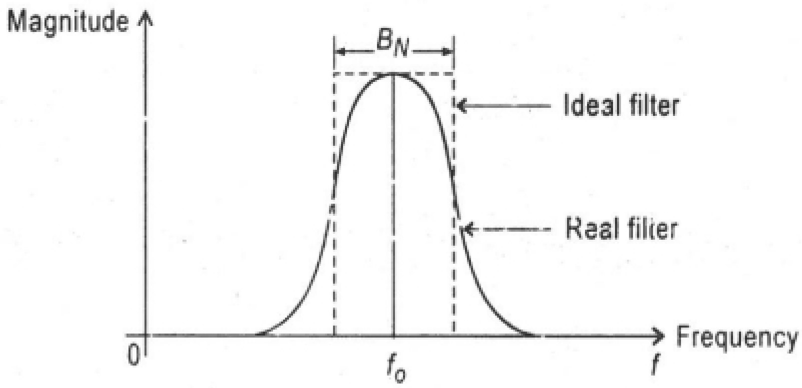


This is very much similar to the spectrum of white light. Hence, the name white Also, since thermal noise does not depend on frequency, it is called *white noise*.

$(\overline{v_n} = \sqrt{4k_B TBR})$

Q.5. *Explain Noise Bandwidth.*

Ans. Noise Bandwidth is defined as *bandwidth of an ideal rectangular filter which passes the same noise power as does the real filter and is denoted by B_N.*



f_o = Central frequency of ideal filter

Q.6. *Explain noise figure measurement using method 'diode noise generator'.*

Ans. One of the most satisfactory methods of determining noise figure of network involves the use of a noise generator and a power measuring device such as a thermocouple type meter as shown in *figure 2.4*.

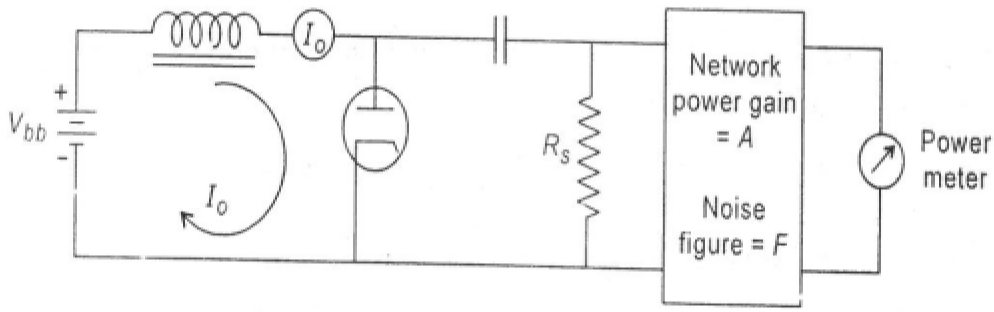
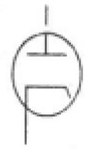


Fig. 2.4 : Experimental determination of noise figure.

Note : *The diode used in this circuit is not the ordinary semiconductor diode. It is the one which is developed by using vacuum tubes. The symbol used for this kind of diode is*



A noise generator, is a device for producing a known and preferably controllable amount of noise for use as a standard of comparison in making noise figure measurements. Most noise generators used in practice are based on the shot noise generated by a diode so operated that the plate current is limited by the cathode temperature. If the tube involved has a pure metal filament, and if the transient time of the electrons is negligibly small compared with the time represented by a cycle in the frequency range of interest then,

$$i_n^2 = 2qI_oB$$

where i_n^2 = Mean-square noise current in band width B

q = Charge on the electron i.e. 1.6×10^{-19} Coulomb

I_o = Direct current flowing between anode and cathode in amperes

B = Equivalent band width of noise involved.

If R_s is the source resistance in ohms, noise power of the noise generator will be given by

$$v_n^2 = (i_nR_s)^2 = 2qI_oR_s^2\,B$$

If internal resistence of the noise generator is chosen equal to the source resistance R_s with which the network under test is normally used P_o, the output noise power with $I_o = 0$, will be given by

$$P_o = (4k_BTR_sB)AF$$

Let P_o' the output noise power for a given value of I_o ; hence, by superposition theorem, we have

$$P_o' = (4k_BTR_sB)AF + (2qI_cR_s^2 B)A$$

Taking the ratio of P_o' and P_o, we get

$$\frac{P_o'}{P_o} = \frac{4k_BTR_sBAF + 2qI_oR_s^2 BA}{4k_BTR_sBAF}$$

$$\frac{P_o'}{P_o} = 1 + \frac{qI_oR_s}{2k_BTF}$$

If we make $P_o' = 2P_o$, by properly adjusting I_o, the above expression simplifies to

$$\frac{2P_o}{P_o} = 1 + \frac{qI_oR_s}{2k_BTF}$$

$$2 = 1 + \frac{qI_oR_s}{2k_BTF}$$

$$1 = \frac{qI_oR_s}{2k_BTF}$$

$$F = \frac{qI_oR_s}{2k_BT}$$

At standard room temperature i.e. at 17 °C, $T = 273 + 17 = 290$ °K, we have

$$F = \frac{1.6 \times 10^{-19} I_oR_s}{2 \times 290 \times 1.38 \times 10^{-23}} = 20I_oR_s$$

Thus knowing the diode current I_o and the source resistance R_s enables the determination of the noise figure F.

2.12 Formulae

(1) Noise voltage $\overline{v}_n = \sqrt{4k_BTBR}$

(2) $$\text{SNR} = \frac{\text{Signal power}}{\text{Noise power}}$$

(3) Noise power $P_n = k_BTB$

(4) Noise factor, $$F = \frac{\text{(SNR) at input}}{\text{(SNR) at output}}$$

(5) Noise figure, $F_{dB} = 10 \log_{10} F$

(6) Equivalent noise temperature.

$$T_e = (F-1)T_0 \quad \text{where } T_0 = \text{Room temperature.}$$

(7) For cascaded networks, the equivalent noise factor is

$$F = F_1 + \frac{F_2 - 1}{G_{A1}} + \frac{F_3 - 1}{G_{A1}G_{A2}} + \text{.........}$$

(8) Noise factor in terms of equivalent noise resistance is

$$F = \frac{R_p + R_n}{R_p}$$

where, R_p = Antenna resistance

R_n = Equivalent noise resistance.

(9) Noise figure using diode noise generator

$$F = 20 I_o R_s$$

where, I_o = Direct current flowing between anode and cathode in amperes

R_s = Source resistance.

2.13 Solved Problems

Problem 1 : A three stage amplifier has following power gains and noise figure (not in dB)

Stage	Power Gain	Noise Figure
1	$G_{A1} = 10$	$F_1 = 2$
2	$G_{A2} = 20$	$F_2 = 4$
3	$G_{A3} = 30$	$F_3 = 5$

Find overall noise figure, gain and equivalent noise temperature.

Solution :

> **Note :** *Here actually noise factors are given i.e. F. Since they have mentioned not in dB.*

Now for cascaded network's

(i) Gain : The gain just gets multiplied.

$\therefore$ Overall gain $= G_{A1} \times G_{A2} \times G_{A3}$

$$\boxed{\text{Overall gain} = 6000}$$

(ii) Noise Figure : The overall noise factor in cascaded network's is given by

$$F = F_1 + \frac{F_2 - 1}{G_{A1}} + \frac{F_3 - 1}{G_{A1} G_{A2}}$$

$$\therefore F = 2 + \frac{4 - 1}{10} + \frac{5 - 1}{10 \times 20}$$

$$\boxed{F = 2.32}$$

Noise figure i.e. Noise factor is dB is

$$F_{dB} = 10 \log_{10} (2.32)$$

$$\boxed{F_{dB} = 3.654 \text{ dB}}$$

(iii) Overall Equivalent Temperature :

$$T_e = (F - 1) T_0$$

$$T_e = (2.32 - 1) \times 300 \text{ °K. (Assuming room temperature } T_0)$$

$$\boxed{T_e = 396 \text{ °K}} \quad \textbf{\textit{Ans.}}$$

Problem 2 : A 300 Ω resistor is connected across the 300 Ω antenna input of a television receiver. The bandwidth of the receiver is 6 MHz and the resistor is at room temperature 27 °C. Find noise power and noise voltage applied to the receiver input.

Given :

$$B = BW = 6 \text{ MHz}$$

$$T = 27 \text{ °C} = 300 \text{ °K (Assume)}$$

$$R_r = 300 \, \Omega \text{ (Resistor)}$$

$$R_a = 300 \, \Omega \text{ (Antenna)}$$

To find : P_n, $\overline{v_n}$

Solution : **(i) Noise Power :**

$$P_n = k_B T B$$

where, $k_B = 1.38 \times 10^{-23}$ J/k

$$\therefore P_n = 1.38 \times 10^{-23} \times 300 \times 6 \times 10^6$$

$$\boxed{\therefore P_n = 2.484 \times 10^{-14} \text{ watts.}}$$

(ii) Noise Voltage :

$$\overline{v_n} = \sqrt{4 k_B T B R}$$

$$\therefore \overline{v_n} = \sqrt{4 \times 1.38 \times 10^{-23} \times 300 \times 6 \times 10^6 \times R} \quad \text{..... (i)}$$

but since R_r and R_a are connected in parallel (it is mentioned R_r across R_a):

$\therefore$ Equivalent resistance is

$$R = (R_r \,||\, R_a)$$

$$= (300 \,||\, 300) = \frac{300 \times 300}{300 + 300} = \frac{90000}{600}$$

$$\boxed{R = 150\ \Omega}$$

Put this in equation (i)

$$\therefore \boxed{\overline{v_n} = 3.8605 \times 10^{-6} \text{ volts}}$$ **Ans.**

Problem 3 : *A receiver connected to an antenna whose resistance is 50 Ω has an equivalent noise resistance of 30 Ω. Calculate the receiver's noise figure in decibles and its equivalent noise temperature.*

Given : Antenna resistance, $R_p = 50\ \Omega$

Equivalent noise resistance, $R_n = 30\ \Omega$

To find : (i) Noise figure in dB

(ii) Equivalent noise temperature.

Solution :

(i) Noise factor $F = \dfrac{R_p + R_n}{R_p}$

$$\therefore F = \frac{50 + 30}{50} = \frac{80}{50}$$

$$\boxed{F = 1.6}$$

$\therefore$ Noise Figure, $F_{dB} = 10 \log_{10} F = 10 \log 1.6$

$$\boxed{F_{dB} = 2.041 \text{ dB}}$$

(ii) Equivalent noise temperature,

$$T_e = (F - 1) T_0$$

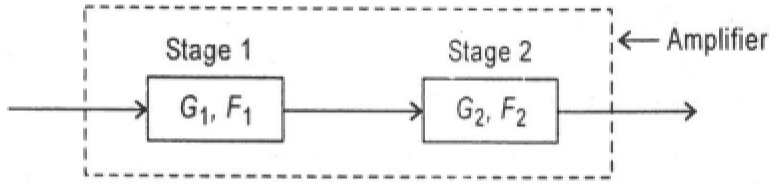
$$\text{Let } T_0 = 300\ ^\circ\text{K} = 27\ ^\circ\text{C}$$

$$\therefore T_e = (1.6 - 1) \times 300$$

$$\boxed{T_e = 180\ ^\circ\text{K}}$$

Problem 4 : In a two stage Cascaded Amplifier, if each stage has a gain of 10 dB, calculate the overall noise figure.

Solution :



$$\text{Gain in dB of stage 1} = 10 \log_{10} G_1 = \text{Gain in dB of stage 2}$$

$$\therefore\ 10 = 10 \log_{10} G_1 = 10 \log_{10} G_2$$

$$\therefore\ \boxed{G_1 = G_2 = 10}$$

Overall noise figure,

$$F = F_1 + \frac{F_2 - 1}{G_1}$$

But, F_1 and F_2 are not given, one option is to assume F_1 and F_2 or just write the final answer as

$$\boxed{F = F_1 + \frac{F_2 - 1}{10}}$$

Problem 5 : Explain the term thermal noise.
Prove that noise voltage $\overline{v_n} = \sqrt{4k_B TBR}$
For electronic device operating at a temperature of 17 °C with a band width of 10 kHz, determine : (i) thermal noise power in dBm. (ii) rms noise voltage for a 100 Ω internal resistance and a 100 Ω load resistance.

Solution : Refer section 2.3 for thermal noise and noise voltage $\overline{v_n} = \sqrt{4k_B TBR}$.

Given :

$$T = 17\,^\circ\text{C} = 290\,^\circ\text{K}$$

$$\text{Bandwidth } B = 10 \text{ kHz}$$

To find : (i) Thermal noise power in dBm

(ii) r.m.s. noise voltage for 100 Ω internal resistance and 100 Ω load resistance.

Solution : (i) Thermal Noise Power :

$$P_n = k_B TB$$

$$\therefore\ P_n = 1.38 \times 10^{-23} \times 290 \times 10 \times 10^3$$

$$\boxed{\therefore\ P_n = 4.002 \times 10^{-17} \text{ watts}}$$

$$\boxed{\therefore\ P_n = 4.002 \times 10^{-14}\ \text{m watts}}$$

Now, P_n in dBm is given by

$$P_{n(\text{dBm})} = 10 \log_{10} (P_n \text{ in m watts})$$

$$= 10 \log_{10} (4.002 \times 10^{-14})$$

$$\boxed{P_{n(\text{dBm})} = -133.772\ \text{dBm}}$$

(ii) r.m.s. Noise Voltage :

As we know, the load resistance and internal resistance are in series

$$\therefore\ R_{eq} = R_i + R_l$$

$$\therefore\ R_{eq} = 100 + 100 = 200\ \Omega$$

$$\therefore \text{ Noise voltage, } \overline{v_n} = \sqrt{4k_B TBR_{eq}}$$

$$= \sqrt{4 \times 1.38 \times 10^{-23} \times 290 \times 10 \times 10^6 \times 200}$$

$$= \sqrt{3.2016 \times 10^{-14}}$$

$$= 1.789 \times 10^{-7}\ \text{volts}$$

$$\boxed{\overline{v_n} = 0.1789\ \mu\ \text{volts}}$$

Note : *In the above problem, in dBm, m signifies milli, hence*

$$P_{n(\text{dBm})} = 10 \log_{10} (P_n \text{ in m watts})$$

if it would have been asked to calculate power in dBμ then,

$$P_{n(\text{dB}\mu)} = 10 \log_{10} (P_n \text{ in } \mu \text{ watts})$$

Problem 6 : *A 300 Ω resistor is connected across the 300 Ω antenna input of a television receiver. The bandwidth of the receiver is 6 MHz and the resistor is at a room temperature of 293 °K or 20 °C. Find noise power and noise voltage applied to the receiver input.*

Given :

$$B = BW = 6\ \text{MHz}$$

$$T = 20\ ^\circ\text{C} = 293\ ^\circ\text{K}$$

$$R_r = 300\ \Omega \text{ (Resistor)}$$

$$R_a = 300\ \Omega \text{ (Antenna)}$$

To find : P_n, $\overline{v_n}$

Solution : Same as problem 2, just values are different.

Problem 7 : *The signal power at the input to an amplifier is 100 μW and the noise power is 1 μW. At the output, the signal power is 1 W and the noise power is 30 mW. What is the noise factor for the amplifier ?*

Given : $S_i = 100\ \mu W,\ N_i = 1\ \mu W$

$S_o = 1\ W,\ N_o = 30\ mW$

To find : F

Solution : Now, the SNR at input is

$$(SNR)_{in} = \frac{S_i}{N_i} = \frac{100\ \mu W}{1\ \mu W} = 100$$

Similarly, SNR at output is

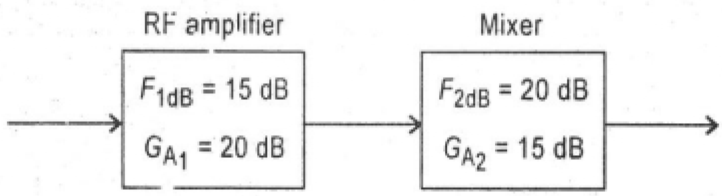
$$(SNR)_{out} = \frac{S_o}{N_o} = \frac{1\ W}{30 \times 10^{-3}\ W} = 33.333$$

$$\therefore \text{ Noise Factor } F = \frac{(SNR)_{in}}{(SNR)_{out}} = \frac{100}{33.333}$$

$$\boxed{F = 3}$$

Problem 8 : *In a radio receiver an RF amplifier and a mixer are connected in cascade. The amplifier has a noise figure of 15 dB and power gain of 20 dB. The noise figure of mixer stage is 20 dB and power gain is 15 dB. Calculate the overall noise figure referred to the input.*

Solution : The given information can be drawn as



Since everything is given in dB, first convert it to normal units

$$\because F_{1dB} = 10 \log_{10} F_1$$

$$\therefore F_1 = 10^{(F_{1dB}/10)}$$

$$\therefore \boxed{F_1 = 31.62}$$

Similarly $G_{A1} = 100$, $F_2 = 100$, $G_{A2} = 31.62$

Now, the overall noise factor is,

$$F = F_1 + \frac{F_2 - 1}{G_{A1}} = 31.62 + \frac{99}{100}$$

$$\therefore \boxed{F = 32.16}$$

$\therefore$ Overall noise figure is, $F_{dB} = 10 \log_{10} F$

$$\therefore \boxed{F_{dB} = 15.133}$$

Problem 9 : *Derive an expression for the total noise figure of a cascaded amplifier. A three stage amplifier has the following specifications :*

Stage	Power Gain	Noise Figure
1	$G_{A1} = 10$	$F_1 = 2$
2	$G_{A2} = 25$	$F_2 = 4$
3	$G_{A3} = 30$	$F_3 = 5$

Find overall noise figure, gain and equivalent noise temperature.

Solution : Same as problem 1, just values are different. Also refer section 2.10.

Problem 10 : *A mixer stage has a noise figure of 20 dB, and this is preceded by an amplifier that has a noise figure 9 dB and an available power gain of 15 dB. Calculate the overall noise figure referred the input.*

Solution : Same as problem 8, just values are different.

***

3 Amplitude Modulation and Generation

Topic	Theory imp	Oral imp
Introduction		
Equation of AM	☆☆☆	☆
Frequency Spectrum	☆	☆
Modulation Index	☆☆	☆☆☆☆☆
Power Distribution	☆	☆
Grid Modulated Class C Amp	☆☆☆	☆☆
Plate Modulated Class C Amp	☆	☆☆
Collector Modulator	☆☆☆☆	☆☆
Low Level Transmitter	☆☆☆☆	☆☆☆
High Level Transmitter	☆☆☆☆	☆☆☆
Comparison of HLM and LLM	☆☆☆☆	☆☆☆☆
FAQ's	☆☆☆	☆☆☆☆
Problems	☆☆☆☆	☆☆

3.0 Introduction

As mentioned in the earlier chapter there are three parameters of any wave

(1) Amplitude

(2) Frequency

(3) Phase

In order to transmit signals over common channel some type of modulation is essential.

We know that modulation is most important part of any common system. Amplitude modulation is the oldest technique of modulation.

3.1 Definition

Amplitude modulation is defined as *modulation process that varies the instantaneous amplitude of carrier signal in accordance with instantaneous amplitude of modulating signal.*

The modulating signal is baseband signal that is to be transmitted to the destination. This signal is low in frequency, random and unpredictable.

In amplitude modulation the instantaneous amplitude of the carrier will vary, but the frequency and phase of the carrier signal will remain constant.

3.1.1 Physical Appearance

Amplitude Modulation can be further explained by using physical appearance of AM wave and other signals evolved. For this we will consider following signals :

(1) Baseband Signal / Modulating Signal

- Pure sine wave with amplitude V_m and instantaneous amplitude v_m [refer *figure 3.1(a)*]

(2) Carrier Signal / High frequency Signal

- A pure sine wave with amplitude V_c and instantaneous amplitude v_c.

  Further amplitude of carrier signal V_c, is greater than that of modulating signal V_m ($V_c > V_m$) [refer *figure 3.1(b)*]

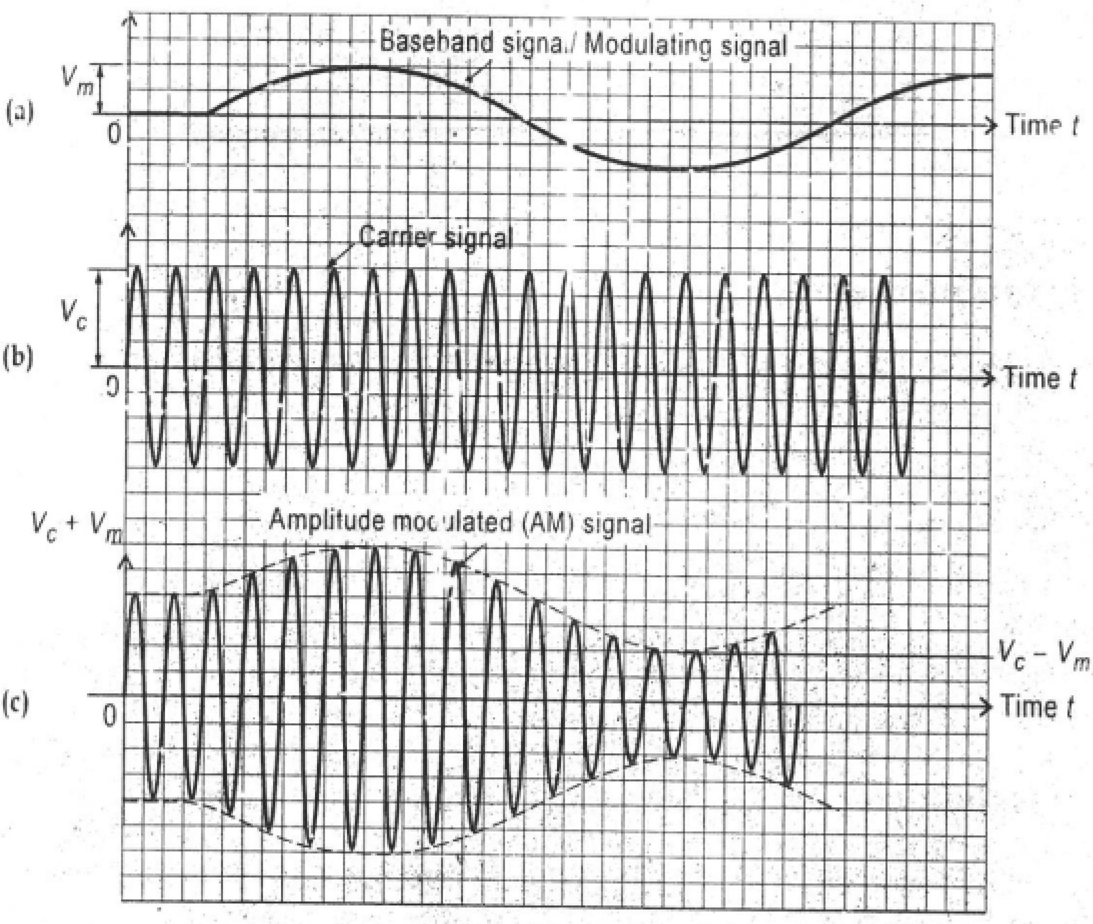


Fig. 3.1 : AM waveform for sinusoidal modulating signal.

- The modulating and carrier signals are shown in *figure 3.1(a)* and *(b)* respectively.
- The AM wave is shown by *figure 3.1(c)*.
- The modulating signal amplitude modulates the carrier signal.
- It can be seen in *figure 3.1(c)* that, after modulation, the amplitude of the carrier varies according to the instantaneous amplitude of modulating signal.

3.2 Equation of AM Wave

Let the equation of modulating signal be

$$v_m = V_m \sin \omega_m t$$

Let the equation of carrier signal be

$$v_c = V_c \sin \omega_c t$$

Now, when this carrier is amplitude modulated,

- Frequency remains same
- Phase remains same
- Amplitude changes in accordance with modulating signal

The general equation of the AM wave is

$$v_{AM} = V \sin \omega_c t \qquad \text{..... (i)}$$

where, V = Peak voltage and

$V \propto V_c + v_m$ $\therefore$ (Amplitude changes in accordance with modulation signal)

$v = V_c + V_m \sin \omega_m t$ (Assuming constant of proportionality to be one)

Substituting in (i)

$$v_{AM} = (V_c + V_m \sin \omega_m t) \sin \omega_c t$$

$$\therefore\ v_{AM} = V_c \sin \omega_c t + V_m(\sin \omega_c t \sin \omega_m t)$$

$$\therefore\ v_{AM} = V_c \sin \omega_c t + \frac{V_m}{2}\,[2 \sin \omega_c t \cdot \sin \omega_m t]$$

$$v_{AM} = V_c \sin \omega_c t + V_m\,[\cos (\omega_c - \omega_m)t - \cos (\omega_c + \omega_m)t]$$

$[\therefore\ 2 \sin A \sin B = \cos(A - B) \cos (A + B)]$

Taking V_c common we get,

$$v_{AM} = V_c\left\{\sin \omega_c t + \frac{V_m}{2V_c}[\cos (\omega_c - \omega_m)t - \cos (\omega_c + \omega_m)t]\right\} \qquad \text{..... (ii)}$$

does not have freq component of fm. as effect of modulation is to translate modulating signal in the freq domain so that it is reflected symmetrically about the carrier freq.

Now, we define a new term

$$V_m/V_c = m_a \text{, modulation index}$$

∴ Equation (ii) is rewritten as

$$\boxed{\underset{\text{(A)}}{v_{AM} = V_c \sin \omega_c t} + \underset{\text{(B)}}{\frac{m_a V_c}{2} \cos(\omega_c - \omega_m)t} - \underset{\text{(C)}}{\frac{m_a V_c}{2} \cos(\omega_c + \omega_m)t}}$$

BW = 2fm.

The equation contains three terms.

(A) $V_c \sin \omega_c t$ i.e. carrier itself

- Frequency = f_c
- Amplitude = V_c

(B) $\frac{m_a V_c}{2} \cos(\omega_c - \omega_m)t$ i.e. Lower Side Band (LSB)

- Frequency = $f_c - f_m$
- Amplitude = $\frac{m_a V_c}{2}$

(C) $\frac{m_a V_c}{2} \cos(\omega_c + \omega_m)t$ i.e. Upper Side Band (USB)

- Frequency = $f_c + f_m$
- Amplitude = $\frac{m_a V_c}{2}$

3.3 Frequency Spectrum of AM Wave

Q. *Explain frequency spectrum of AM wave.*

Frequency spectrum of any signal is just the graph of amplitude versus frequency.

Its significance is that it shows what all frequencies are present in the signal with their amplitudes.

Now, we know that the equation of an AM wave is

$$v_{AM} = V_c \sin \omega_c t + \frac{m_a V_c}{2} \cos(\omega_c - \omega_m)t + \frac{m_a V_c}{2} \cos(\omega_c + \omega_m)t$$

Thus the frequency spectrum of AM wave is as shown in *figure 3.2*.

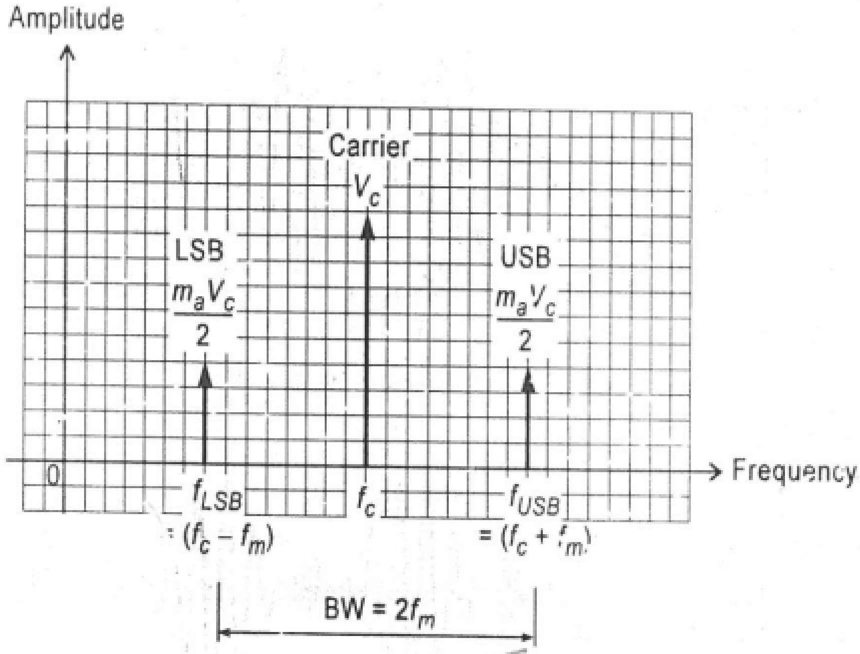


Fig. 3.2 : Single sided frequency spectrum of AM wave.

The following information can be inferred from the above graph :

(i) The amplitudes of LSB and USB are equal.

(ii) The LSB and USB are equidistant from the location of carrier signal. Thus, f_c is also called *central frequency*.

(iii) Bandwidth of an AM signal is $2f_m$, where f_m = Modulating frequency.

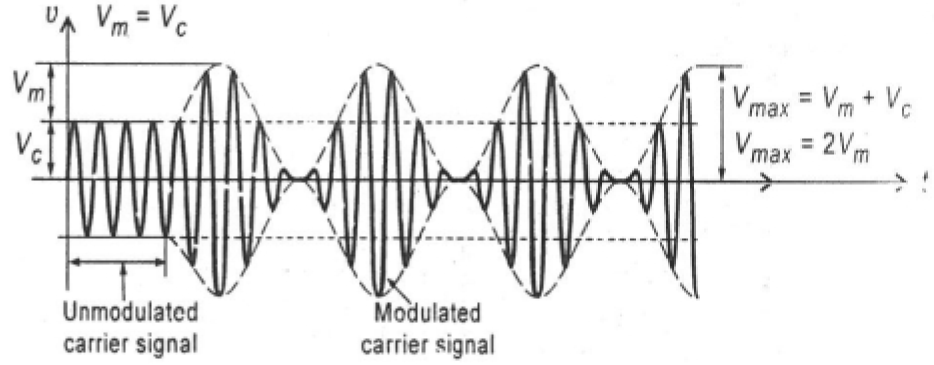
3.4 Modulation Index

Q. ***Derive an expression to calculate the modulation index from A.M. waveform.***

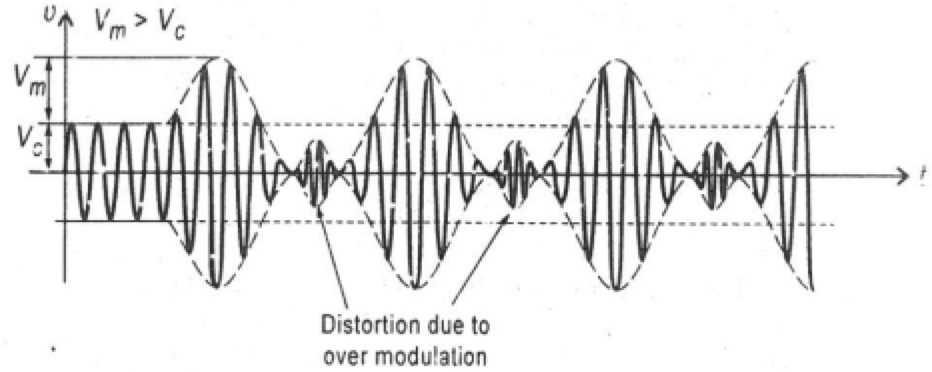
Definition : The modulation index of an AM wave is defined as *the ratio of amplitudes of modulating signal to the carrier signal*. The modulating index is denoted by m_a where a signifies AM. Thus m_a is given as

$$m_a = \frac{V_m}{V_c}$$

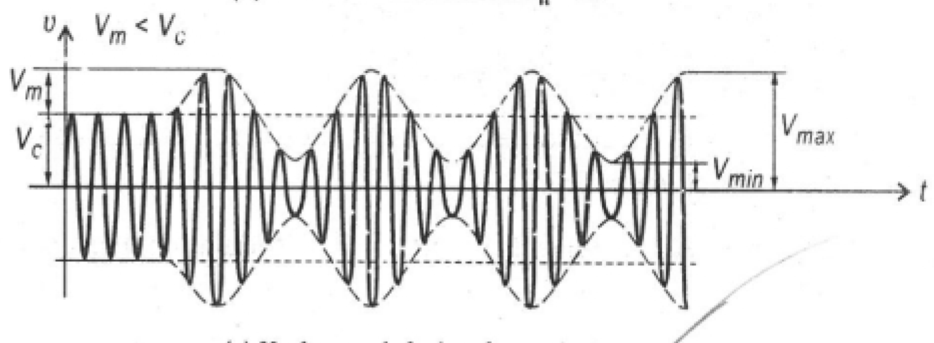
The modulation index is also referred as modulation factor, modulation coefficient, depth of modulation or degree of modulation.



(a) 100 % modulation for $m_a = 1$.



(b) Over modulation for $m_a > 1$.



(c) Under modulation for $m_a < 1$.

Fig. 3.3 : AM waveform for various values of m_a.

The modulation index decides the physical appearance of an AM wave (*figure 3.3*)

- $m_a = 1$, when $V_m = V_c$ implies 100% modulation
- $m_a > 1$, when $V_m > V_c$ implies over modulation
- $m_a < 1$, when $V_m < V_c$ implies under modulation

Practically feasible value of m_a is $0 < m_a < 1$

Calculation of Modulation Index

As we already know, the modulation index m_a is given by

$$m_a = \frac{V_m}{V_c} \qquad \text{..... (i)}$$

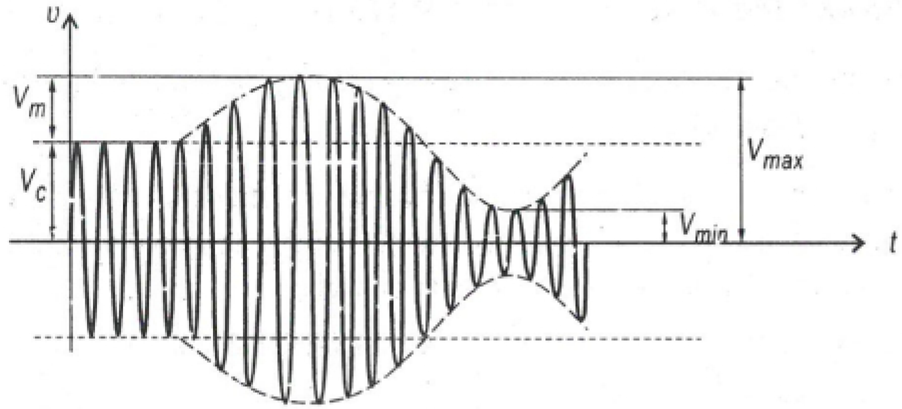


Fig. 3.4 : AM wave for calculation of modulation index m.

For the *figure 3.4* the maximum and minimum peak values of AM wave are obtained as :

$$V_{max} = V_c + V_m \quad \text{..... (ii)}$$

$$V_{min} = V_c - V_m \quad \text{..... (iii)}$$

Note : *Modulation index of AM,* m_a *is also denoted by* m.

Referring to *figure 3.4.*

Subtracting equation (iii) from (ii)

$$V_{max} - V_{min} = 2V_m$$

$$\therefore \; V_m = \frac{V_{max} - V_{min}}{2}$$

Adding equation (ii) and (iii)

$$V_{max} + V_{min} = 2V_c$$

$$\therefore \; V_c = \frac{V_{max} + V_{min}}{2}$$

Substituting in equation (i), we get

$$m_a = \frac{V_{max} - V_{min}}{V_{max} + V_{min}}$$

or the percentage modulation is given as

$$\% \, m_a = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} \times 100$$

Note : *This technique is used for calculating* m_a *by viewing an AM wave on a CRO.*

3.5 Power Distribution in an AM Wave

Q. *Derive the relation between output power of the AM transmitter and the depth of modulation.*

As the AM wave travels through the channel, the strength of the signal degrades due to two factors, i.e. power loss and channel noise. Therefore, it is highly desirable that the power of the signal should be good enough.

The AM wave consists of two side bands and the carrier signal.

∴ The total power P_T is given as

$$P_T = \text{Carrier power} + \text{Power in LSB} + \text{Power in USB}$$

$$\therefore\ P_T = P_c + P_{LSB} + P_{USB}$$

> **Note :** *If R is the resistance of antenna and the signal applied is V, then power dissipated in antenna is given as*
>
> $$\text{Power} = \frac{V_{rms}^2}{R}\ ; \text{ where } R = \text{Resistance of antenna}$$
> $$V_{rms} = \text{Root mean square voltage.}$$

Carrier Power is given by

$$P_c = \frac{V_{c\,rms}^2}{R}$$

$$P_c = \frac{\left[V_c/\sqrt{2}\right]^2}{R}$$

$$\boxed{P_c = \frac{V_c^2}{2R}}$$

Power in Sidebands

$$P_{USB} = P_{LSB} = \frac{\left[\frac{mV_c}{2}/\sqrt{2}\right]^2}{R}$$

as peak amplitude $= \dfrac{mV_c}{2}$

$$\boxed{\therefore\ P_{USB} = P_{LSB} = \frac{m^2V_c^2}{8R}}$$

∴ Total power in both sidebands is

$$\boxed{P_{SB} = \frac{m^2V_c^2}{4R}}$$

Total Power

The total power is given by

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

$$\therefore\ P_T = P_c + \frac{m^2}{4} P_c + \frac{m^2}{4} P_c$$

$$\boxed{\therefore\ P_T = \left[1 + \frac{m^2}{2}\right] P_c}$$

Transmission Efficiency

Transmission efficiency of an AM wave is the ratio of transmitted power which contains the information (i.e. total sideband power) to the total power transmitted.

$$\eta = \frac{P_{LSB} + P_{USB}}{P_T} = \frac{\left[\frac{m^2}{4} P_c + \frac{m^2}{4} P_c\right]}{\left[1 + \frac{m^2}{2}\right] P_c} = \frac{m^2}{2 + m^2}$$

AM Power in Terms of Current

$$\boxed{I_T = I_C\left[1 + \frac{m^2}{2}\right]^{\frac{1}{2}}}$$

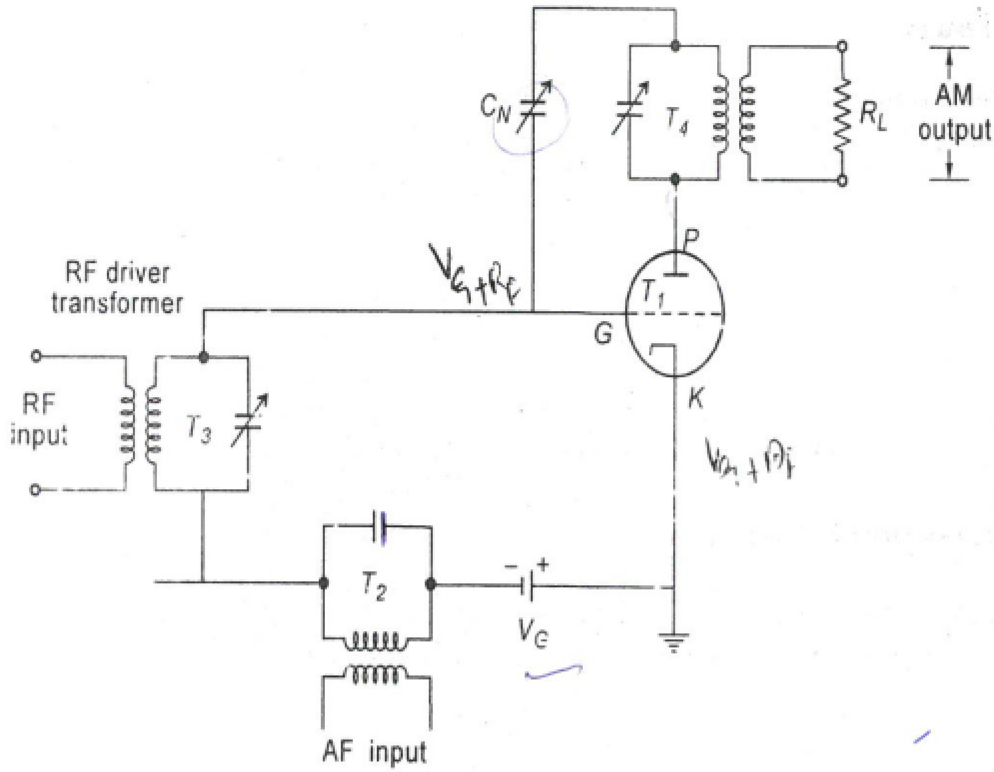
3.6 AM Generation Circuits

> **Note :** *Before studying AM generation circuits please read what is tank circuit and its use from the FAQ's section.*

3.6.1 GRID Modulated Class C Amplifier

Q. Explain with aid of waveforms how a grid modulated class C amplifier generates AM ?

The circuit diagram of grid modulated class C amplifier is as shown in *figure 3.5(a)*.



V_G : Grid biasing voltage

T_1 : Triode

T_2 : Tank circuit used for selection of AF

T_3 : Tank circuit used for selection of RF

T_4 : Tank circuit used for getting AM wave

C_N : Neutralization capacitor

Fig. 3.5(a) : Grid modulated class C amplifier. RLM

> **Note :** *AF signal, modulating signal and the base band signal means the same. And, RF signal and carrier signal means the same.*

Operation

- The voltage V_G is used to bias the triode T_1 below cut-off.
- The tank circuit T_2 is used for selecting only AF signal and T_3 is used for selecting only RF signal.
- The AF input is applied in series with V_G such that the total bias on triode T_1 is proportional to the modulating signal.
- Because of this, the amplitude of total bias of T_1 varies according to the amplitude of modulating signal.

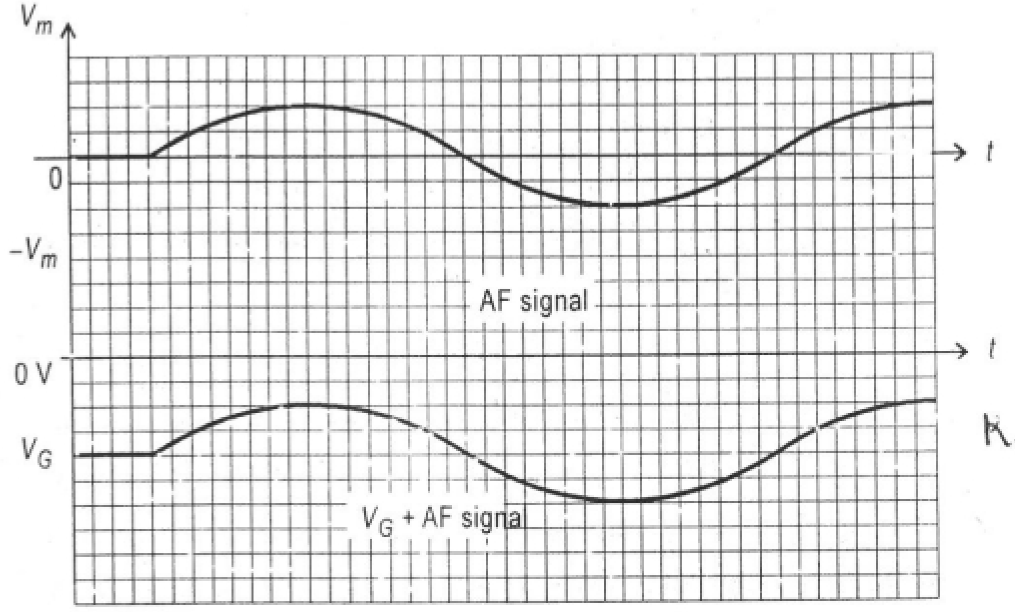


Fig. 3.5(b)

- The RF signal or carrier is superimposed on total bias.

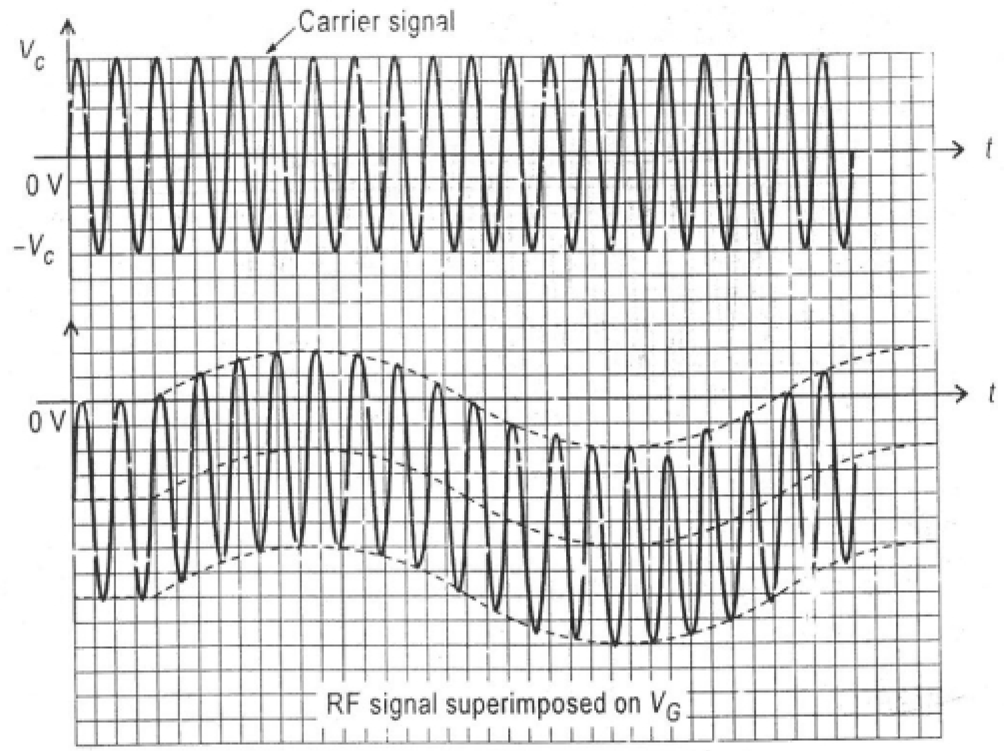


Fig. 3.6(a)

- In triode T_1, superimposed signal V_G + RF is applied at the gate and V_G + AF is applied at the cathode.
- The resulting plate current flows in pulses.

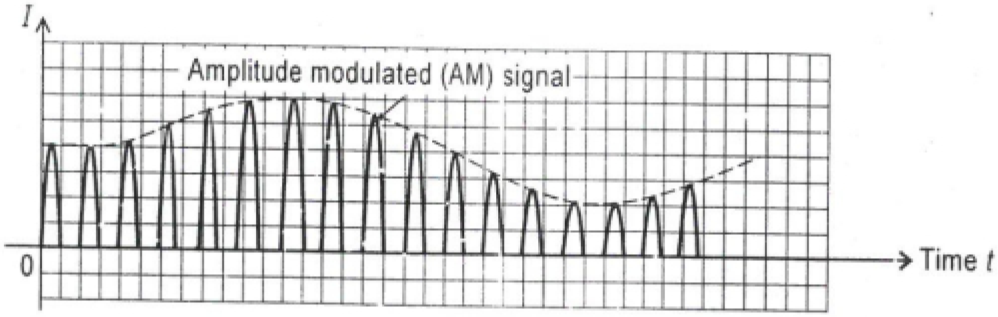
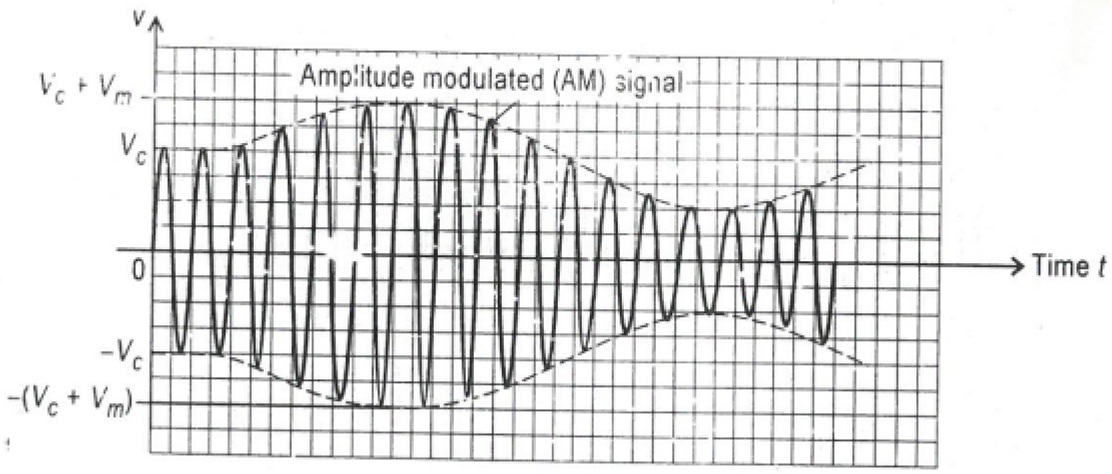


Fig. 3.6(b)

Note : *The figure 3.6(b) looks like a half AM wave, whenever such signal is given to a tank circuit it converts it into a full AM wave. This is a very important characteristic of tank circuit.*

- The amplitude of these pulses varies according to the instantaneous amplitude of modulating signal.
- This plate current in the form of pulses is given to the Tank Circuit T_4 and we get the amplitude modulated signal, AM wave across load resistor R_L.



LLM

Fig. 3.6(c)

Advantage

- The only advantage of this circuit is requirement of low AF power because AF is applied in grid.

Disadvantage

- There is distortion in AM output due to non linear transfer characteristics of the triode.
- Low efficiency compared to plate modulated class C amplifier as the grid is not driven to its limit.
- The output signal can not be directly transmitted as output power is low.
- Amplifiers are required after modulation, hence cost of system increases.

Applications

- It is normally used in TV transmitting station to modulate video signai which is normally of low power

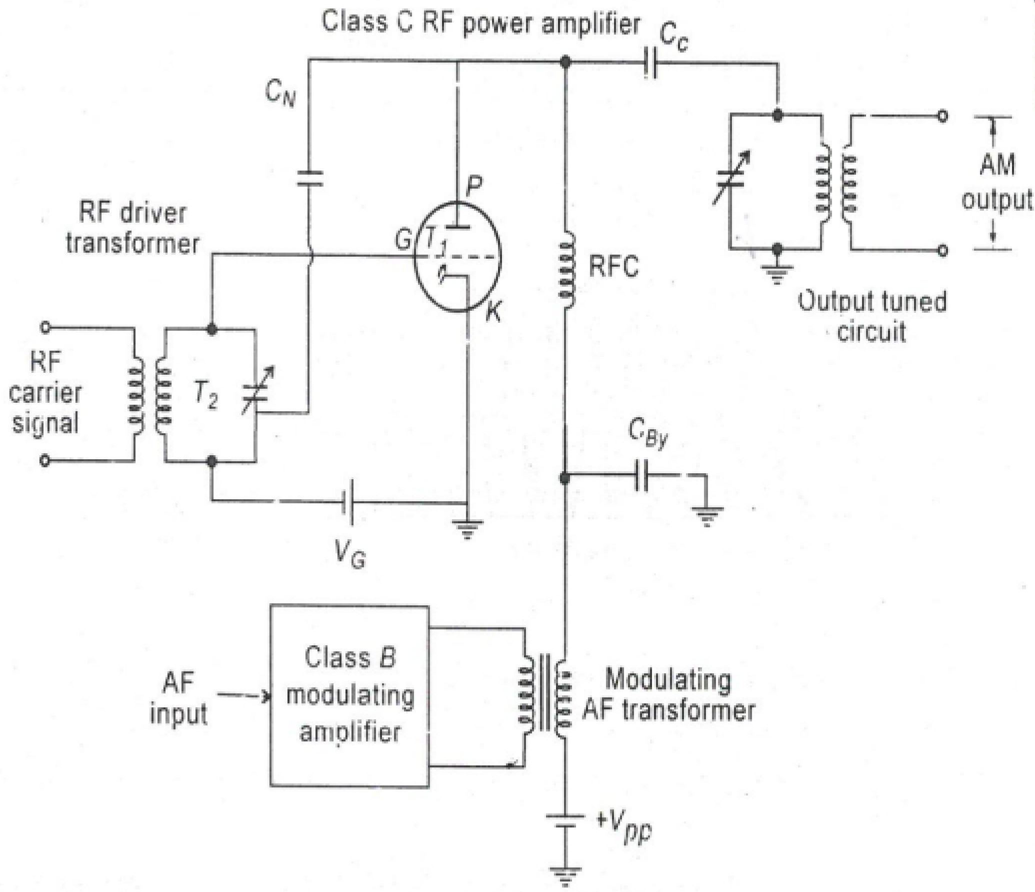
3.6.2 Plate Modulated Class C Amplifier

HLM

Here, the modulating signal is applied in series with plate supply voltage of class C power amplifier, hence it is called *plate modulated class C amplifier*.

The circuit diagram is as shown in *figure 3.7(a)*.

Grid
Plate
K - Cath



T_1 : Triode

T_2 : Tank circuit to select RF

Fig. 3.7(a) : Plate modulated class C amplifier.

Operation

- The AF signal is amplified with help of class *B* modulating amplifier
- The amplified AF signal is applied to the plate of triode T_1 in series with d.c. voltage V_{PP}.

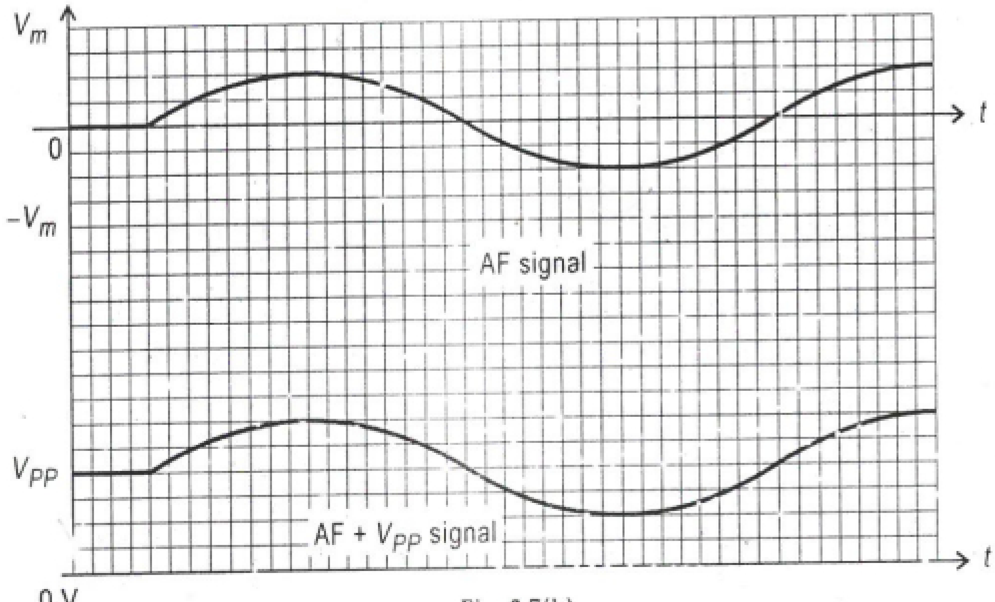


Fig. 3.7(b)

- The tank circuit T_2 is used to select the carrier signal.

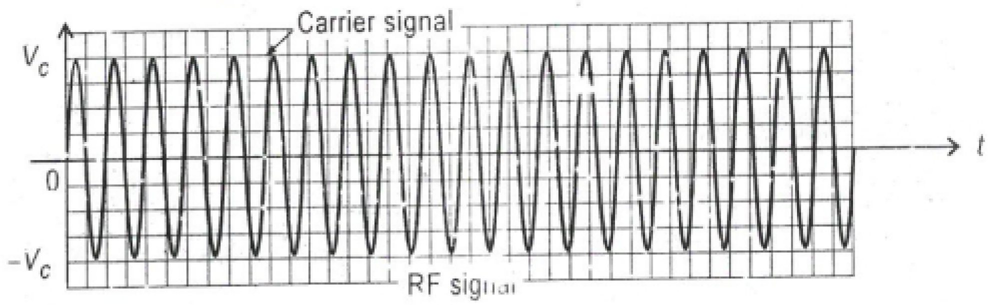


Fig. 3.7(c)

- Voltage V_G is applied such that the triode T_1 is biased below cut-off, hence it operates as class C amplifier.
- At the output of the Triode we get current pulses, the amplitude of which depends upon maximum power supply V_{PP}.
- As the power supply V_{PP} and AF are in series, the amplitude of the current pulses depend upon the instantaneous amplitude of modulating signal.

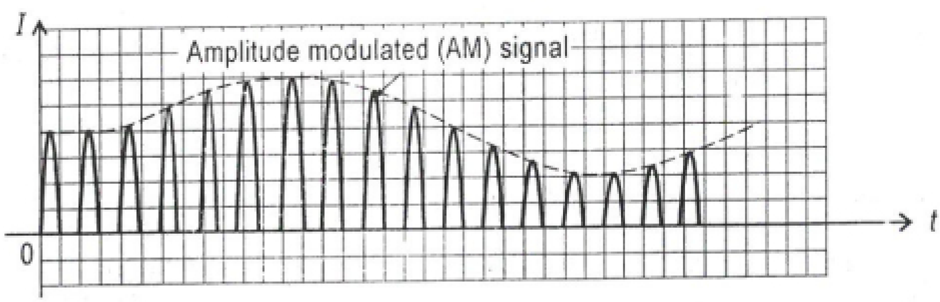


Fig. 3.8(a)

- These pulses are applied to the tank circuit which produces AM signal across R_L.

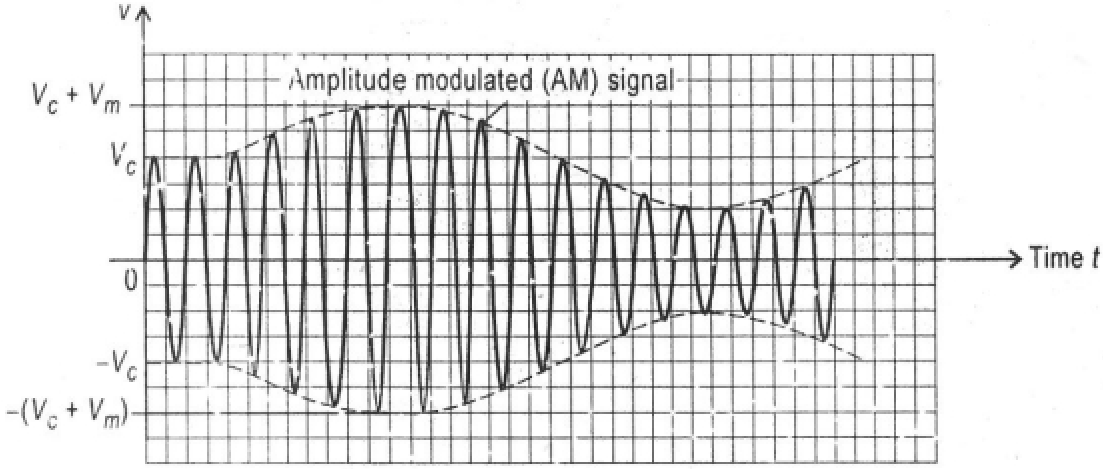


Fig. 3.8(b)

Advantage

(1) Efficiency is much higher than Grid Modulated Class C Amplifier (GMCCA) as grid is driven to its limit.

(2) Distortion in output is less compared to GMCCA.

(3) The modulated signal can be directly transmitted due to high power at output.

(4) Since AF is applied in plate, the distortion due to non linear characteristics of triode will be less.

(5) Use of class B amplifier to amplify AF provides good efficiency.

Disadvantage

(1) AF power required is high as AF is applied in plate.

Application

Plate Modulated Class C Amplifier (PMCCA) is used in radio broadcasting and other high frequency transmission.

3.6.3 Collector Modulator

Q. Explain collector modulation with a neat sketch.

The circuit diagram of collector modulator is as shown in *figure 3.9(a)*.

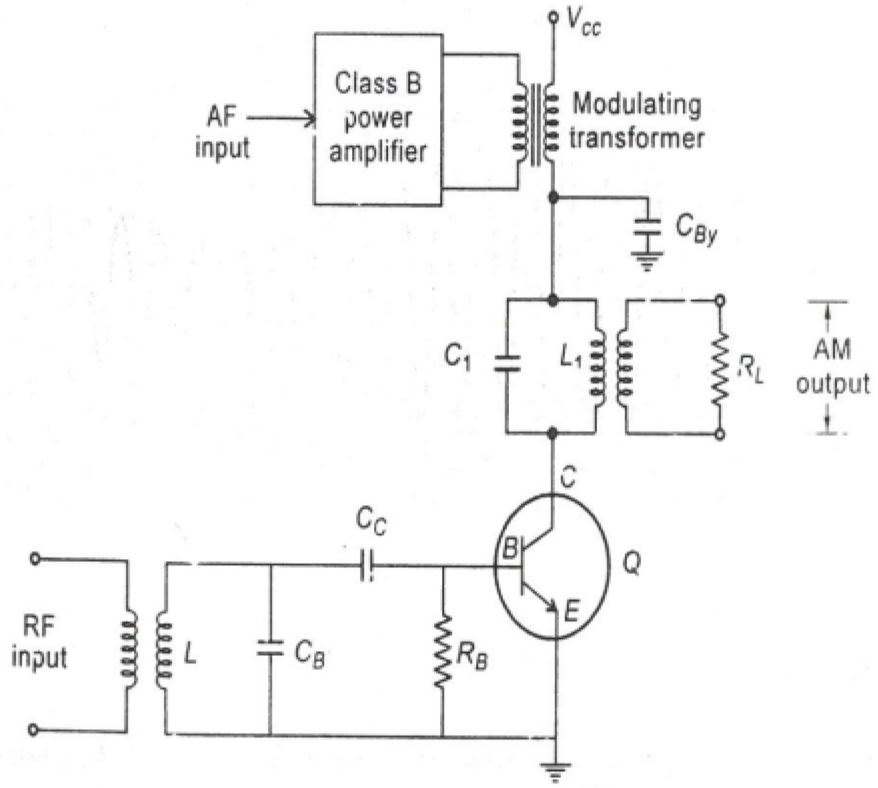


Fig. 3.9(a)

HLM

Operation

- RF carrier signal is applied at the base of transistor Q.

Base

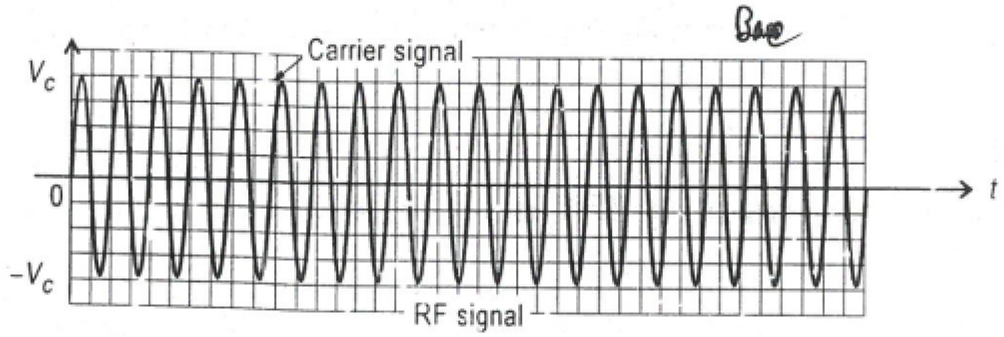


Fig. 3.9(b)

- The transistor will conduct only for positive half cycle of carrier signal.
- The AF input is given to the class B amplifier which increases power level of AF signal.
- This modulating signal is the applied in series with V_{CC}.

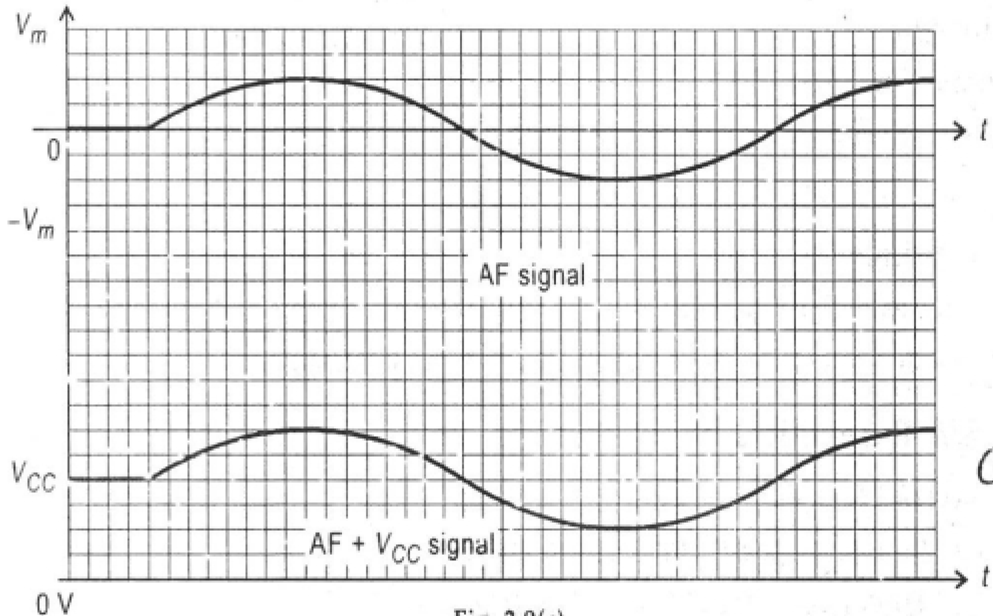


Fig. 3.9(c)

- Therefore the supply voltage varies according to the amplitude of modulating signal.
- Such varying voltage is applied to the collector which results in variation in amplitude of current pulses at the collector of transistor.

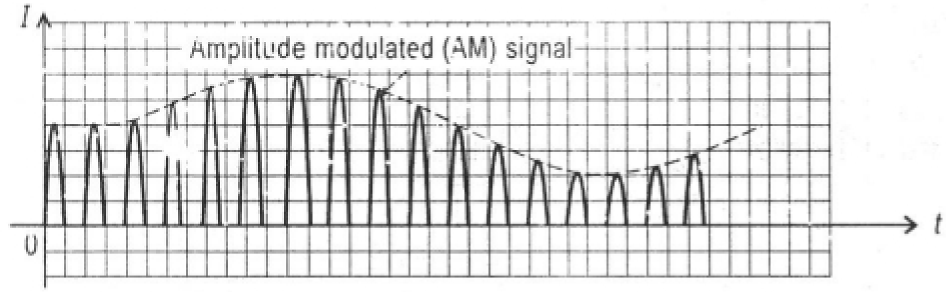


Fig. 3.9(d)

- These pulses are applied to the tank circuit and we get the AM signal across load resistor R_L.

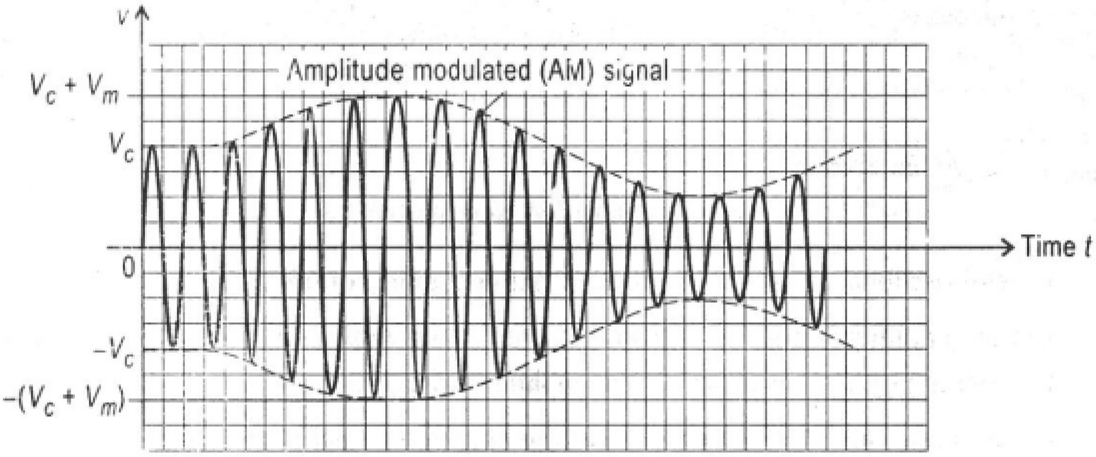


Fig. 3.9(e)

Advantage

(1) Better linearity.

(2) High efficiency.

(3) High output power.

Disadvantage

(1) Requires high AF power.

(2) Collector saturation does not allow 100% modulation.

3.7 AM Transmitters

Q. Sketch block diagrams of :

(i) Low level AM DSBFC transmitter.

(ii) High level AM DSBFC transmitter.

Explain the function of each block and distinguish between the working of both transmitters.

3.7.1 Low Level Modulated AM Transmitter

- The block diagram of low level AM transmitter is as shown in *figure 3.10*.

Diagram

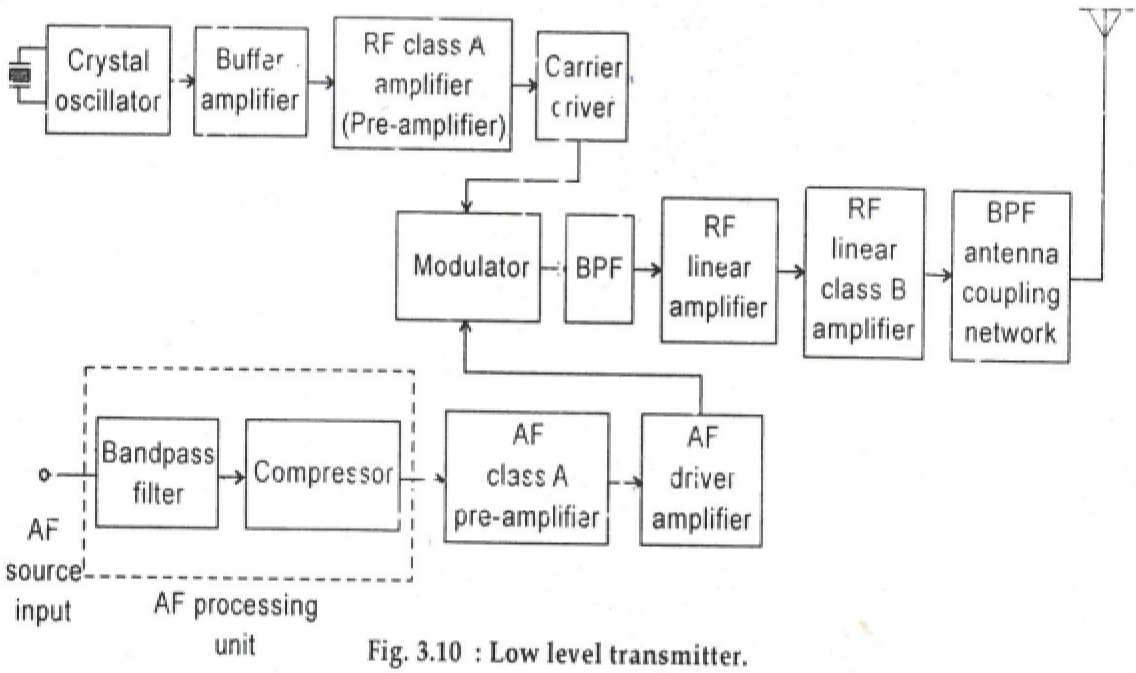


Fig. 3.10 : Low level transmitter.

- Crystal oscillator is used as it provides better stability of frequency.
- Buffer amplifier is used to isolate oscillator from high power amplifiers so that the loading of the oscillator by pre-amplifier is divided.
- Pre-amplifier and carrier driver raises the amplitude level of carrier.
- AF source can be microphone, magnetic tape or CD etc.

- AF processing unit filters AF so as to occupy the current bandwidth (generally 10 kHz).
- AF pre-amplifier and driver raises the amplitude level of AF signal.

Operation

- The carrier generated by the oscillator and the amplified modulating signal is applied to the modulator.
- At the output of modulator we get AM wave.
- This AM signal is then amplified using chain of linear amplifiers to raise its power level.
- The amplitude modulated signal is then transmitted using transmitting antenna.
- The low level transmitter does not require a large AF modulator power so its design is simplified.
- However, the efficiency is much lower compared to high level modulation.

3.7.2 High Level Modulated AM Transmitter

- The block diagram of high level AM transmitter is as shown in *figure 3.11*.

Diagram

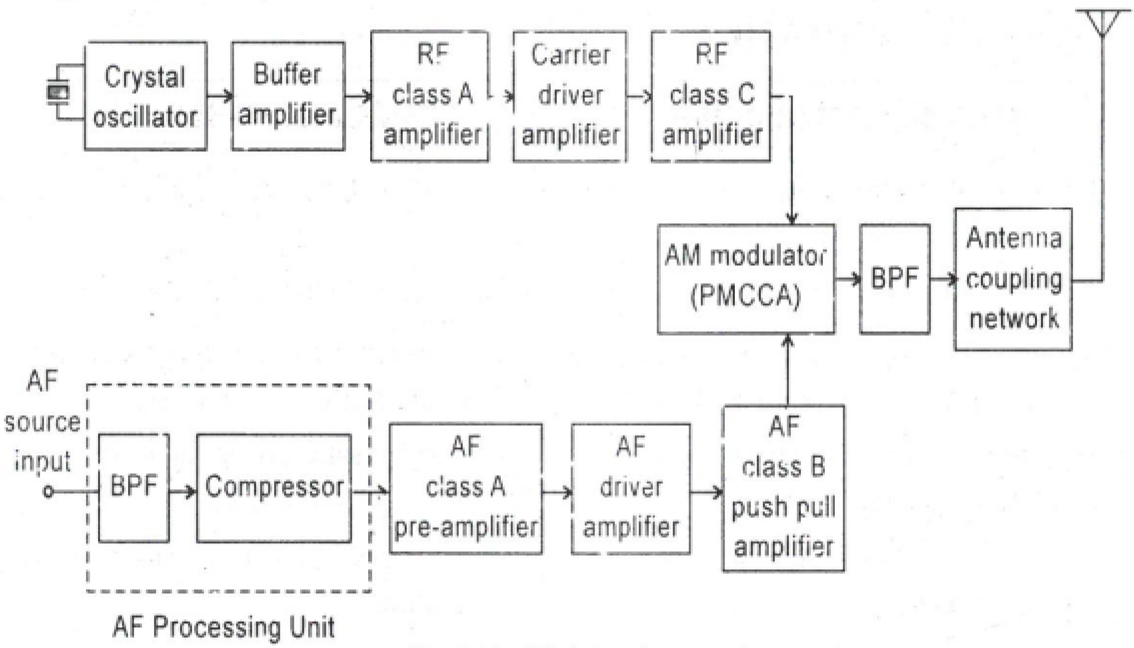


Fig. 3.11 : High level transmitter.

- Crystal oscillator is used as it provides better stability of frequency.
- Buffer amplifier is used to isolate oscillator from high power amplifiers so that the loading of the oscillator by pre-amplifier is divided.
- Pre-amplifier and carrier driver raises the amplitude level of carrier.
- AF source can be microphone, magnetic tape or CD etc.

- AF processing unit filters AF so as to occupy the current bandwidth (generally 10 kHz).
- AF pre-amplifier and driver raises the amplitude level of AF signal.
- The main thing to notice here is that after AF driver amplifier, AF power amplifier is used.

Operation

- The carrier generated by the oscillator and the amplified modulating signal is applied to the modulator.
- At the output of modulator we get AM wave.
- Compared to LLM transmitter, in this transmitter RF undergoes additional power amplification, i.e. after carrier driver, carrier power amplifier is used.
- The amplitude modulated signal is then transmitted using transmitting antenna.

Note : *In High Level Modulation (HLM) first the signals are amplified and then modulated and in Low Level Modulation (LLM) the signals are first modulated then amplified.*

Differentiate Between HLM and LLM

Sr.	High Level Modulation	Low Level Modulation
(1)	Modulation takes place at high power level.	Modulation takes place at low power level.
(2)	High efficiency.	Comparatively low efficiency.
(3)	Signal generated can be directly transmitted after modulation.	Signal generated can not be directly transmitted after modulation.
(4)	Modulating signal is applied at collector/plate.	Modulating signal is applied at grid or emitter.
(5)	No amplifiers are required after modulation.	Amplifiers are required after modulation.
(6)	Distortion in generated output is less.	Distortion in generated output is more.
(7)	Complex system.	Comparatively less complex.
(8)	**Applications** : High power broadcast transmitters.	**Applications** : TV transmitting station, wireless intercoms, short range walkie-talkies.

Table 3.1

3.8 Frequently Asked Questions

Q.1. What is tank circuit and what is its use ?

Ans. Tank circuit or Tuning circuit is just a capacitor and an inductor connected in parallel

There are many uses of tank circuit in communication systems. Some of them are listed below. The main uses of tank circuits are :

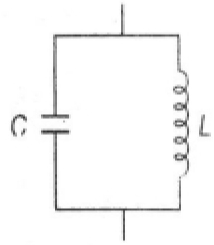


Fig. 3.12

(i) To select a particular frequency and reject other frequency

The above circuit will resonate at frequency

$$\boxed{f_r = \frac{1}{2\pi\sqrt{LC}}}$$

i.e. it will give maximum output when incoming frequency is

$$f_i = f_r = \frac{1}{2\pi\sqrt{LC}}$$

For rest all frequencies it will give low output. In short, if the incoming frequency is $f_i = f_r$ then it is selected and other frequencies are rejected. f_r can be varied by varying L or C.

(ii) To obtain full AM wave from half AM pulses

This answer is explained in *Q.8*.

(iii) To convert frequency variations to amplitude variations

This technique is widely used in FM detectors.

Refer chapter - 6 Radio Receivers.

Q.2. Amplitude modulation is a waste of power and bandwidth. Justify.

Ans. **(1) Power**

(a) $$P_T = P_C\left(1 + \frac{m^2}{2}\right)$$

If $m = 1$ then

$$P_T = \frac{3}{2} P_C \Rightarrow P_C = \frac{2}{3} P_T \approx 67\ \%\text{ of } P_T$$

i.e. $2/3^{rd}$ of total power is consumed by the carrier where carrier does not carry any information.

(b) $$P_T = P_C + P_{LSB} + P_{USB} = \frac{2}{3}P_T + P_{SB}$$

where P_{SB} = Power in both sidebands.

$$\therefore \quad P_{SB} = P_T\left(1 - \frac{2}{3}\right) = \frac{1}{3}P_T$$

$\therefore$ Power consumed by both sidebands $= \frac{1}{3}P_T$

$\therefore$ Power consumed by one sideband $= \frac{1}{6}P_T \approx 16\ \%$ of P_T

i.e. $1/6^{th}$ of total power is consumed by an extra sideband which contains same information as other.

(c) Therefore a total of 83 % of total power is wasted in carrying a carrier containing no information and a extra sideband.

(2) Bandwidth

As in AM we have to pack two sidebands of the same signal in a channel, there is a wastage of band width.

Q.3. Explain the difference between amplitude modulation and linear addition of carrier and information signals.

Ans. In **Amplitude Modulation** instantaneous amplitude of carrier is varied according to modulating signal.

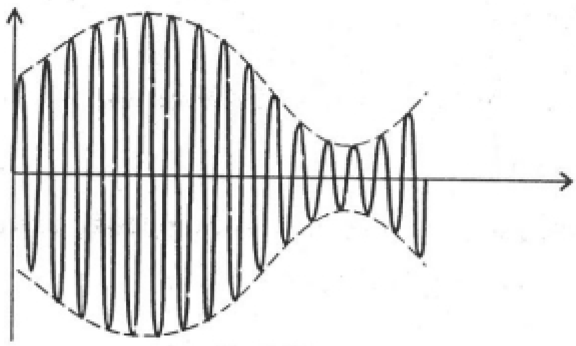


Fig. 3.13

In **Linear Addition** the carrier is superimposed on the modulating signal without any variation in phase or frequency.

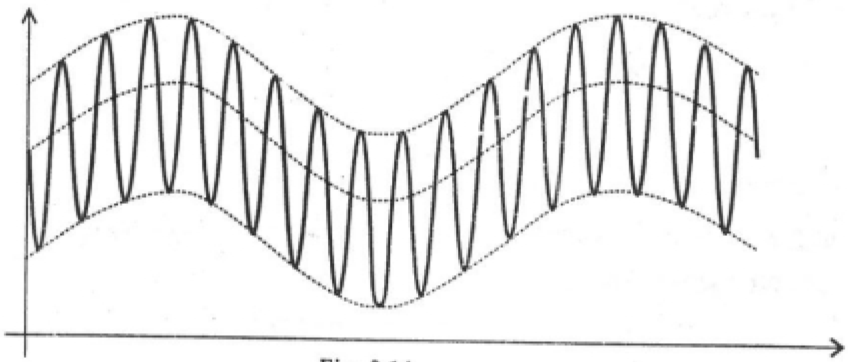


Fig. 3.14

Q.4. Describe different ways in which modulation index of an AM signal can be measured.

Ans. The following are the methods to measure the modulation index.

(i) Using AM wave

(ii) Using trapezoidal display of AM wave.

(i) Using AM Wave : Refer calculation of modulation index in section 3.3.

(ii) Using Trapezoidal Display of AM Wave : When we give the AM signal to the oscilloscope, we get trapezoidal shape on the screen as shown in *figure 3.15*.

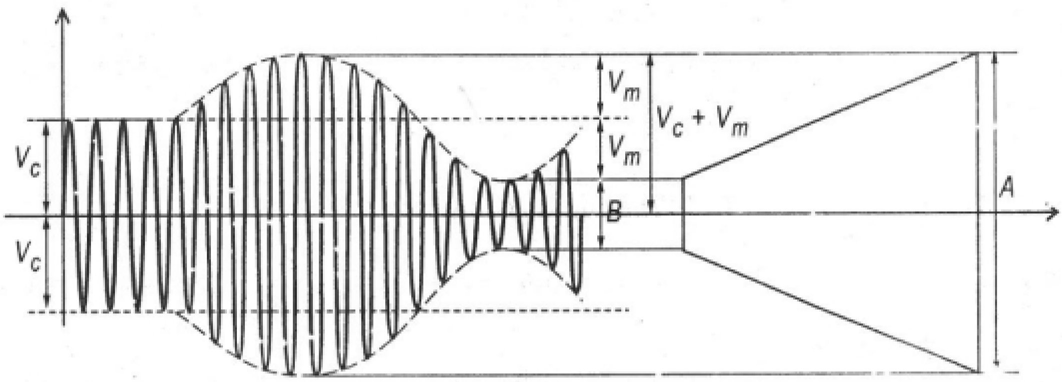


Fig. 3.15

We can see that

$$A = 2(V_c + V_m)$$

$$B = 2(V_c - V_m)$$

We know that

$$m = \frac{V_m}{V_c}$$

$\therefore$ we get the modulation index as

$$m = \frac{A - B}{A + B}$$

Q.5. Derive formulae for total carrier power when it is simultaneously modulated by more than one sine wave.

Ans. The equation of total transmitted power is

$$P_T = P_c\left(1 + \frac{m^2}{2}\right)$$

where carrier is modulated by only one sine wave.

If there are several signals modulating the carrier, then the effective modulation index is the square root of the quadratic sum of each modulation index.

$$m_e = \sqrt{m_1^2 + m_2^2 + m_3^2 + \text{........}}$$

And the power is given as

$$P_T = P_c\left(1 + \frac{m_e^2}{2}\right)$$

Q.6. *What do you mean by voice channel bandwidth ? If voice channel signal modulates a carrier then specify the bandwidth of the resultant.*

Ans. The human audible frequencies are in the range 20 Hz to 20 kHz.

But in specific applications such as telephony where only voice signal is the main concern, the range taken is different.

The range is 300 Hz to 3 kHz. This channel is voice channel bandwidth.

If voice channel signal amplitude modulates a carrier then the bandwidth is 6 kHz or 8 kHz.

Q.7. *What is carrier shift ? What does it indicate ?*

Ans. Carrier shift is a form of amplitude distortion which is introduced because of shift in d.c. voltage level.

Due to this the negative and positive alterations do not match. We have following types of shift.

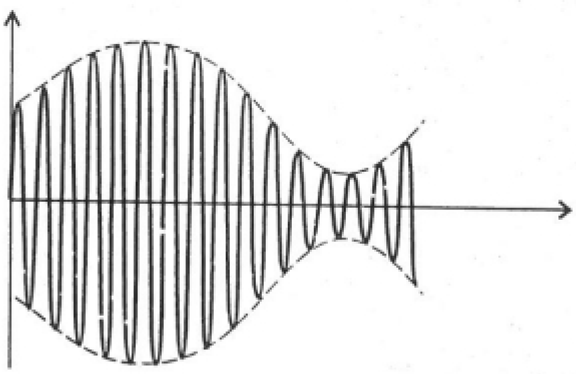


Fig. 3.16(a) : Amplitude modulation without carrier shift.

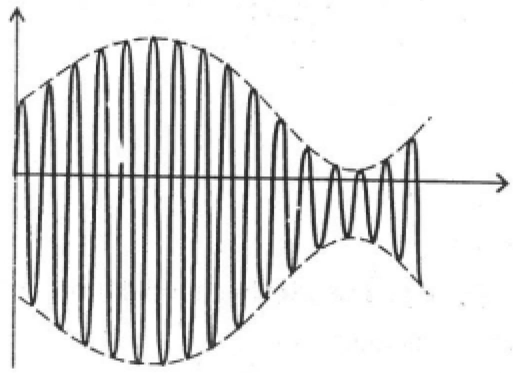


Fig. 3.16(b) : Negative carrier shift.

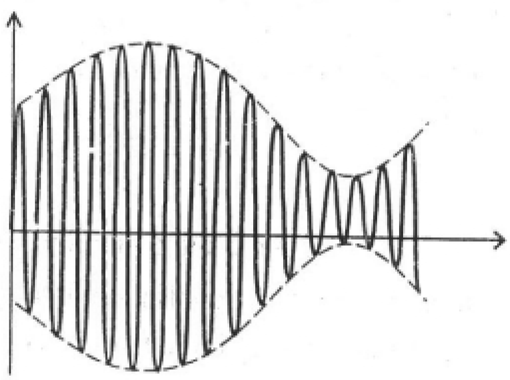


Fig. 3.16(c) : Positive carrier shift.

Q.8. Explain why LC tank circuit is used in AM generation. Also explain Flywheel effect.

Ans.

- LC tank circuit is used to get oscillations across capacitor or inductor.

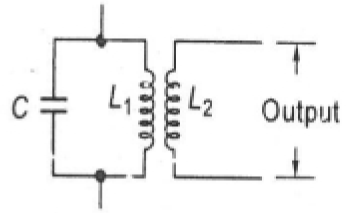


Fig. 3.17(a) : LC tank circuit.

- When a capacitor charges, the energy stored in it is $\frac{1}{2} CV^2$ and it is shown as
- The capacitor discharges and the energy is transferred to the inductor.
- Due to the energy in inductor, the current flows in same direction charging C in opposite direction.
- Again the capacitor discharges transferring energy to inductor.
- Due to repetition of charging and discharging of capacitor, oscillations are produced across tank.

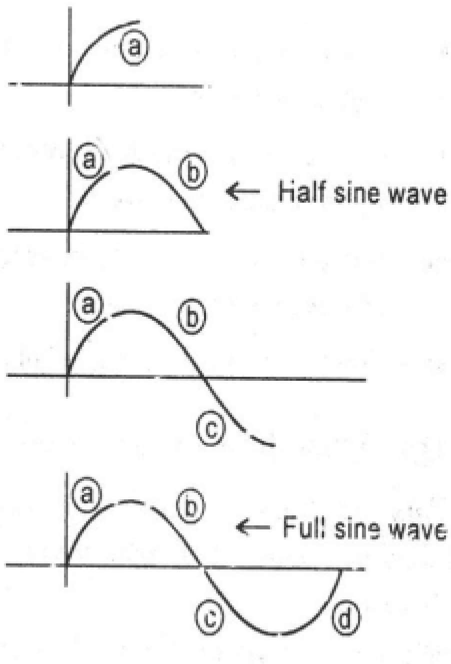


Fig. 3.17(b)

- If capacitor C and inductor L happens to be ideal then we would get oscillations as
- But due to loss of energy in the circuit, we get damped oscillations.

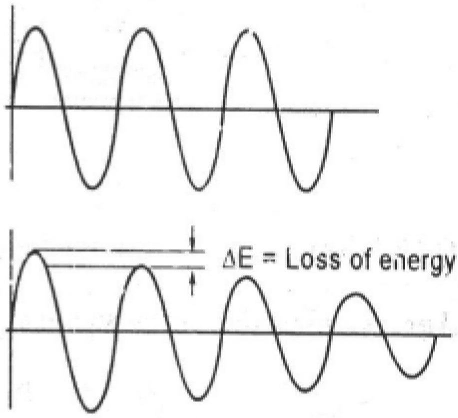


Fig. 3.17(c)

Flywheel Effect

- From the *figure 3.17(c)* we can see that in damped oscillations there is continuous energy loss in consecutive cycles.
- To avoid this, we can compensate the energy loss by providing the energy equal to the loss at the start of each cycle.

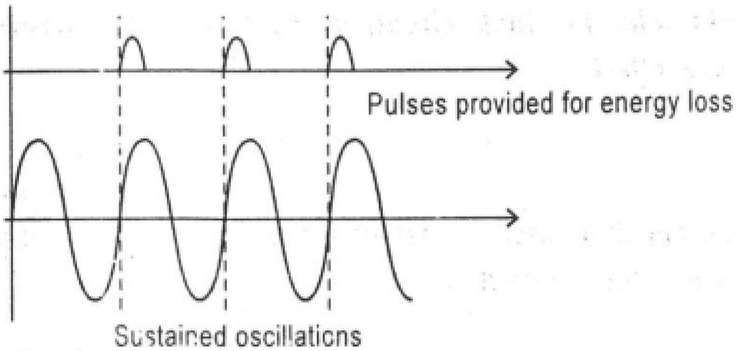


Fig. 3.18

- Therefore we can conclude that the amplitude of oscillations depend upon the energy provided.
- If we provide energy greater than the energy loss then resultant amplitude of the signal would increase.
- Similarly if we supply energy less than the energy loss then resultant amplitude of signal would decrease.
- This is known as *flywheel effect*.

Q.9. What do you mean by antenna coupling ?

Ans. Antenna coupling is nothing *but matching the output impedance of power amplifier with input impedance of antenna system*. It is done for maximum power transfer

We have antenna coupling network between RF amplifier and antenna.

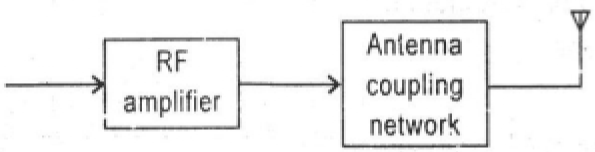


Fig. 3.19

Q.10. What happens when modulation index is 0 ?

Ans. We have the carrier signal $v_c = V_c \sin \omega_c t$ which is shown as below

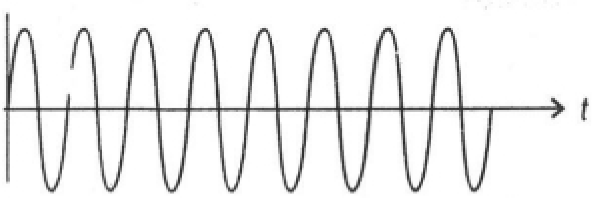


Fig. 3.20

The equation of AM is

$$v = (V_c + V_m \sin \omega_m t)$$

$$v = V_c\left(1 + \frac{V_m}{V_c}\sin\omega_m t\right)\sin\omega_c t$$

$$\text{We know that } m = \frac{V_m}{V_c}$$

$$\text{In this case } m = 0$$

$$\therefore\ v = V_c \sin\omega_c t$$

i.e. after modulation we get the same carrier signal

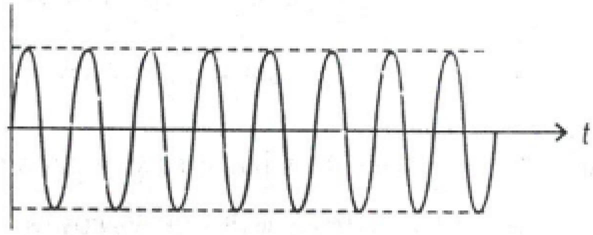


Fig. 3.21

$$\text{Modulation index} = 0$$

At the receiver, since only the peaks are detected, we get d.c. signal.

Q.11. What happens when two signals are added and passed through a non-linear device ? State application of this principle.

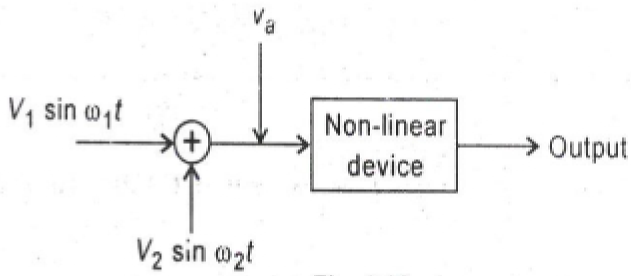


Fig. 3.22

Ans.

$$\text{Output of adder} = v_a = V_1\sin\omega_1 t + V_2\sin\omega_2 t \qquad \text{...... (i)}$$

The characteristic of a non-linear device i.e. output in terms of input is

$$\text{o/p} = a + b(\text{i/p}) + c(\text{i/p})^2$$

Fig. 3.23

Here a, b, c are constants and they depend on circuit hardware.

Now our input is v_a

$$\therefore \text{ Output } = a + bv_a + cv_a^2$$

$$= a + b(V_1 \sin \omega_1 t + V_2 \sin \omega_2 t) + c(V_1 \sin \omega_1 t + V_2 \sin \omega_2 t)^2$$

$$= a + bV_1 \sin \omega_1 t + bV_2 \sin \omega_2 t$$
$$+ c(V_1^2 \sin^2 \omega_1 t + 2V_1V_2 \sin \omega_1 t \sin \omega_2 t + V_2^2 \sin^2 \omega_2 t)$$

$$\therefore \text{ Output } = a + bV_1 \sin \omega_1 t + bV_2 \sin \omega_2 t + cV_1^2 \sin^2 \omega_1 t + cV_2^2 \sin^2 \omega_2 t$$
$$+ 2cV_1V_2\left(\frac{1}{2}\right)\Big(\cos(\omega_1 - \omega_2)t - \cos(\omega_1 + \omega_2)t\Big)$$
$$\left(\because\ 2 \sin A \sin B = \cos(A - B) - \cos(A + B)\right)$$

$$\therefore \text{ Output } = a + bV_1 \sin \omega_1 t + bV_2 \sin \omega_2 t + cV_1^2 \sin^2 \omega_1 t + cV_2^2 \sin^2 \omega_2 t$$
$$+ cV_1V_2 \cos(\omega_1 - \omega_2)t - cV_1V_2 \cos(\omega_1 + \omega_2)t$$

Also,

$$\because \sin^2 A = \frac{(1 - \cos 2A)}{2}$$

$$\therefore \text{ Output } = a + bV_1 \sin \omega_1 t + bV_2 \sin \omega_2 t$$
$$+ \frac{cV_1^2}{2}(1 - \cos 2\omega_1 t) + \frac{cV_2^2}{2}(1 - \cos 2\omega_2 t)$$
$$+ cV_1V_2 \cos(\omega_1 - \omega_2)t - cV_1V_2 \cos(\omega_1 + \omega_2)t$$

$$\boxed{\begin{aligned} \therefore \text{ Output } = & \overset{(1)}{\left(a + \frac{cV_1^2}{2} + \frac{cV_2^2}{2}\right)} + \overset{(2)}{\Big(bV_1 \sin \omega_1 t + bv_2 \sin \omega_2 t\Big)} \\ & - \overset{(3)}{\left(\frac{cV_1^2}{2} \cos 2\omega_1 t\right)} - \overset{(4)}{\left(\frac{cV_2^2}{2} \cos 2\omega_2 t\right)} \\ & + \overset{(5)}{\Big(cV_1V_2 \cos(\omega_1 - \omega_2)t\Big)} - \overset{(6)}{\Big(cV_1V_2 \cos(\omega_1 + \omega_2)t\Big)} \end{aligned}}$$

Thus from above equation we can conclude that after adding two signals and then passing them through nonlinear device, we can get the following results

① $\left(a + \frac{cV_1^2}{2} + \frac{cV_2^2}{2}\right)$ it is just a d.c. value.

② $\Big(bV_1 \sin \omega_1 t + bV_2 \sin \omega_2 t\Big)$ the original two signals.

③ $\left(\frac{cV_1^2}{2}\cos 2\omega_1 t\right)$ this is the frequency upscaled version of the original signal.
It is used by multiplier circuits.

④ $\left(\frac{cV_2^2}{2}\cos 2\omega_2 t\right)$ upscaled version of the original signal.

⑤ $\left(cV_1V_2\cos(\omega_1 - \omega_2)t\right)$ it gives the difference frequency.
This is used to implement balanced modulators and mixers.

⑥ $\left(cV_1V_2\cos(\omega_1 + \omega_2)t\right)$ it gives the sum of input frequencies.
Again used in mixers.

Note : *The italic part is the application of this principle.*

Q.12. Give a real life example of an amplitude modulated wave.

Ans. In real life, the actual voice signal looks like the signal given below.

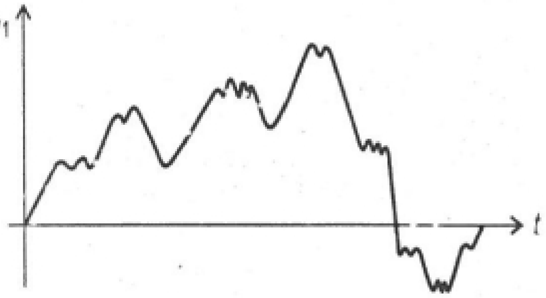


Fig. 3.24 (a) : Modulating signal v_{m_1}.

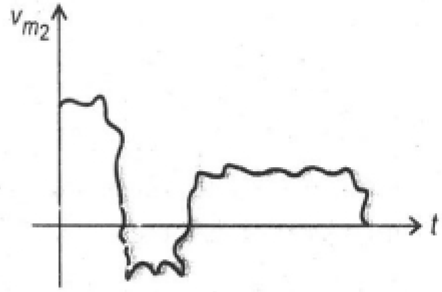


Fig. 3.24 (b) : Modulating signal v_{m_2}.

Thus, the corresponding AM wave obtained by taking this signal as the modulating signal is

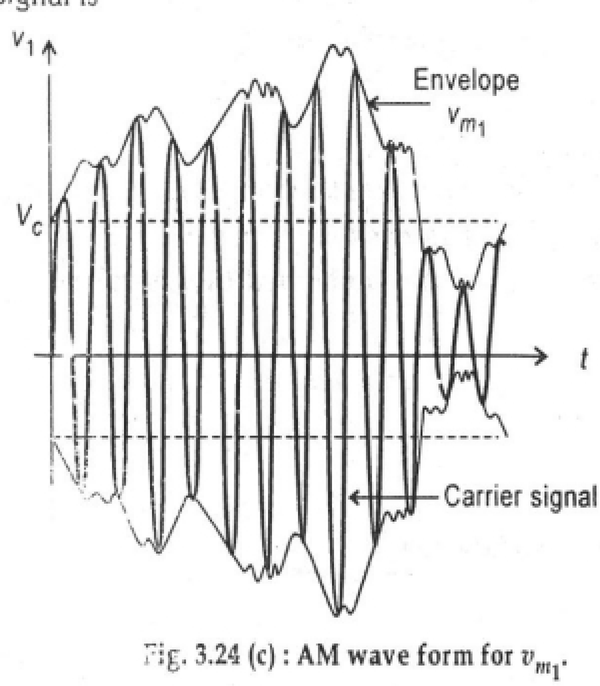


Fig. 3.24 (c) : AM wave form for v_{m_1}.

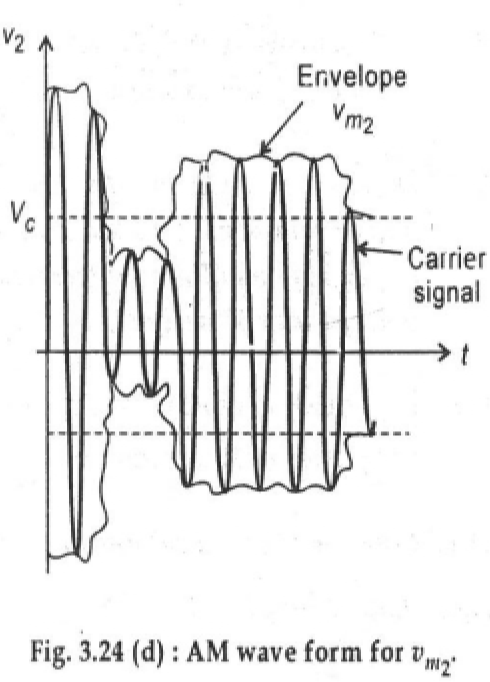


Fig. 3.24 (d) : AM wave form for v_{m_2}.

Q.13. What do you mean by flywheel effect ?

Ans. Refer Q.8.

Q.14. Why do any amplifier that follow the modulator circuit in an AM DSB-FC transmitter have to be linear ?

Ans. At the output of the modulator circuit, we get amplitude modulated signal. The amplitude of the amplitude modulated signal goes on changing.

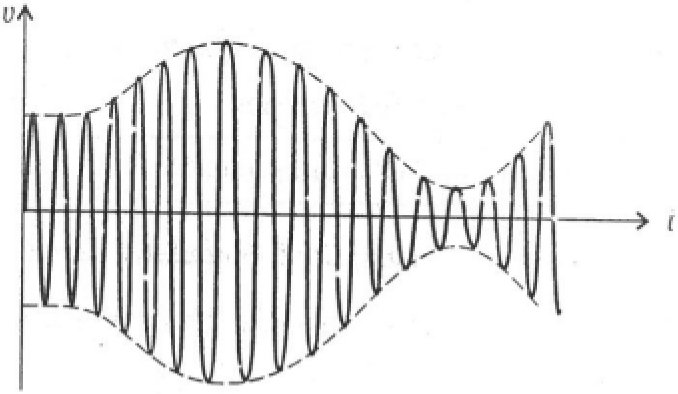


Fig. 3.25 : AM wave.

Therefore we are forced to use linear amplifiers which linearly amplifies the signal and no distortion is there.

Q.15 What is low level modulation and high level modulation ? Explain with neat circuit diagram the collector modulated class C transistor amplifier. What are the advantages of collector modulation over base modulation.

Ans. For comparison of high level and low level transmitters refer Q.13.

For the remaining question refer section 3.6.1.

Q.16. Derive the equation for an Amplitude modulated wave and explain the power relations. Calculate the percentage power saving when the carrier and upper side band are suppressed in an AM wave modulated to a depth of 100 percent and 50 percent.

Ans. For equation of AM wave refer section 3.2.

For power relations refer section 3.5.

Q.17. Compare Low Level and High Level modulation.

Ans. Refer section 3.7.

Q.18. Why is it desirable to have the modulation index of an AM signal as large as possible without overmodulatin ?.

OR

Modulation index for AM should be lesser than one. Explain.

Ans. The total power in the envelope of amplitude modulated signal increases with modulation index according to the following relation.

$$P_T = \left(1 + \frac{m^2}{2}\right) P_c$$

i.e. as modulation index increases, P_T also increases.

$$m = \frac{V_m}{V_c}$$

Modulation index should be such that V_m is less or equal to V_c, otherwise distortion will be there in the output due to overmodulation.

When $m = 0.5$

$$P_T = \left(1 + \frac{1}{8}\right) P_c = \frac{9}{8} P_c = 1.125\, P_c$$

When $m = 0.9$

$$P_T = \left(1 + \frac{0.9^2}{8}\right) P_c = 1.405\, P_c$$

It is clear from above examples that when the modulation index is more, the transmission power is more hence it is desirable to have modulation index near to 1 but not more than 1. If it is more than 1, then it will distort the signal.

$\therefore$ range of modulation index is $1 > m > 0$.

Q.19. What are the advantages of low level modulation ? Draw the circuit of single transistor emitter modulator and explain the operation with output waveforms.

***Ans.* Advantages of LLM :**

(i) It is simple to design.

(ii) Number of amplifiers are less compared to HLM.

(iii) Less in cost.

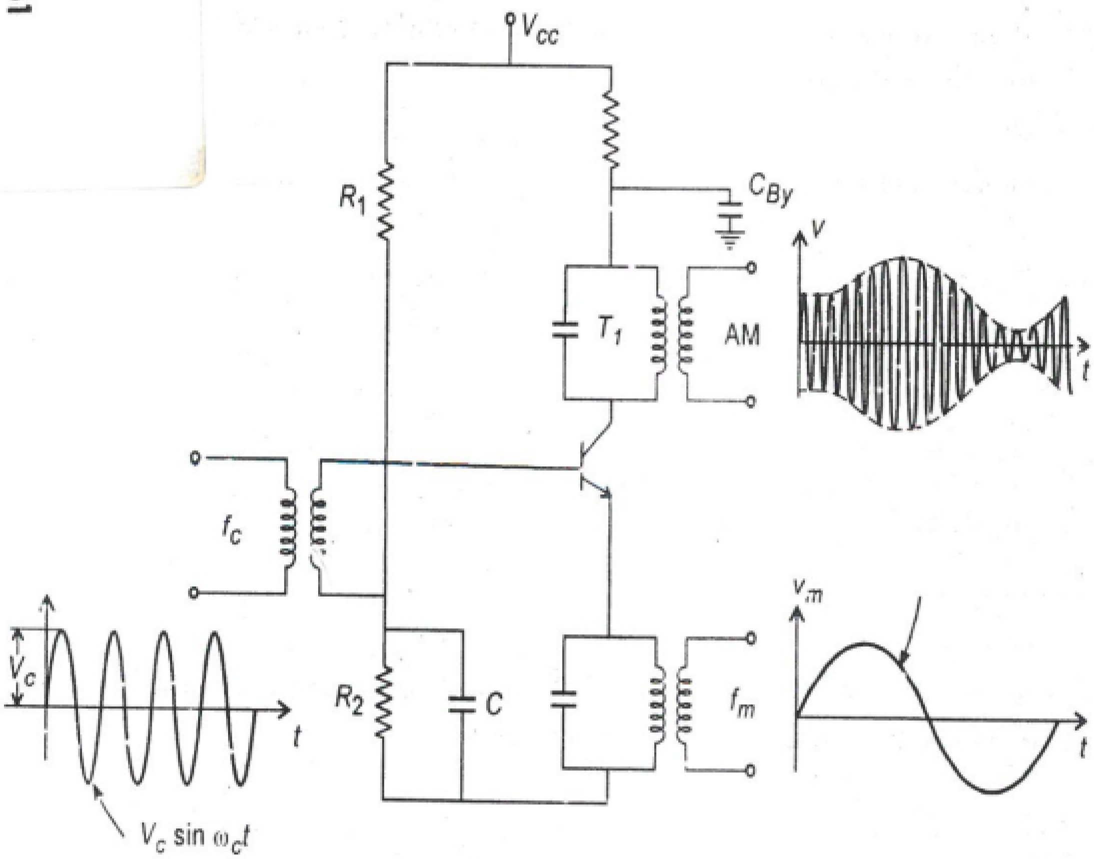


Fig. 3.26 : Transistor - Emitter modulation.

(1) If we apply two signals of frequencies f_c and f_m to a non-linear component, in this case to the transistor, at output we get $f_c + f_m$, $f_c - f_m$, f_c, harmonics of f_c and f_m and a d.c. component.

(2) $f_c + f_m$, $f_c - f_m$, f_c, forms the AM wave i.e. DSB-FC wave.

(3) The AM wave is selected using T_1 tank circuit.

(4) The modulating signal is applied at the emitter and carrier signal is applied at the base.

(5) The resultant AM wave is generated at the collector.

Waveforms

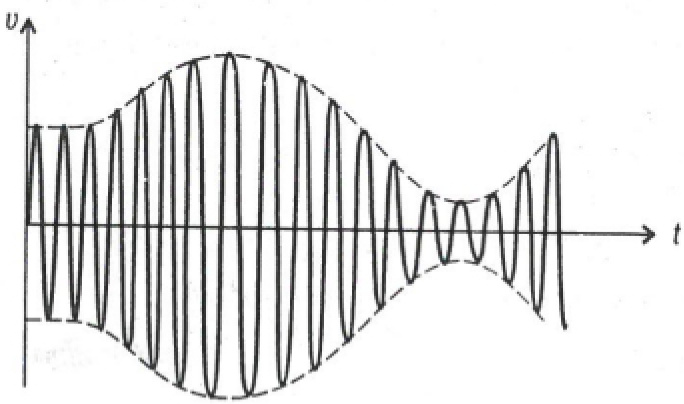


Fig. 3.27 : AM wave.

3.9 Formulae

(1) Equation $\quad v_{AM} = V_c \sin \omega_c t + \frac{mV_c}{2} \cos(\omega_c - \omega_m)t - \frac{mV_c}{2} \cos(\omega_c + \omega_m)t$

(2) Bandwidth $\quad BW = 2f_m$

(3) Modulation Index $\quad m = \frac{V_m}{V_c}$

(4) Upper Sideband Frequency $\quad f_c + f_m$

(5) Lower Sideband Frequency $\quad f_c - f_m$

(6) Modulation Index $\quad m = \frac{V_{max} - V_{min}}{V_{max} + V_{min}}$

(7) Carrier Power $\quad P_c = \frac{V_c^2}{2R}$

(8) Each Sideband Power $\quad P_{USB} = P_{LSB} = \frac{m^2}{4} P_c = \frac{m^2 V_c^2}{8R}$

(9) Total Sideband Power $\quad P_{SB} = \frac{m^2 V_c^2}{4R}$

(10) Total Transmitted Power $\quad P_T = \left(1 + \frac{m^2}{2}\right) P_c = \frac{V_{rms}^2}{R}$

where V_{rms} = r.m.s. voltage of AM signal

(11) Transmission Efficiency $\quad \eta = \frac{m^2}{2 + m^2} \times 100\ \%$

(12) Power in Terms of Current $\quad I_T = I_C \left[1 + \frac{m^2}{2}\right]^{\frac{1}{2}}$

(13) Peak Voltage of AM $= V_c + V_m$

3.10 Solved Problems

Problem 1 : *A sinusoidal carrier has an amplitude of 10 V and frequency 30 kHz. It's amplitude modulated by a sinusoidal voltage of amplitude 3 V and frequency 1 kHz. Modulated voltage is developed across 50 Ω resistance.*

(i) *Write equation for modulated wave.*

(ii) *Plot the modulated wave showing maxima and minima of the waveform.*

(iii) *Determine modulation index.*

(iv) *Draw spectrum of modulated wave.*

(v) *Calculate total average power.*

(vi) *Calculate power carried by sidebands.*

Given :

$$V_c = 10 \text{ V}$$
$$f_c = 30 \text{ kHz}$$
$$V_m = 3 \text{ V}$$
$$f_m = 1 \text{ kHz}$$
$$R = 50\ \Omega$$

Solution :

(i) Equation of modulated wave

$$v = (V_c + V_m \sin \omega_m t) \sin \omega_c t$$
$$= [10 + 3 \sin (2\pi \times 10^3 t)] \sin (6\pi \times 10^4 t)$$
$$v = 10[1 + 0.3 \sin (2\pi \times 10^3 t)] \sin (6\pi \times 10^4 t)$$

(ii) Plot the modulated wave

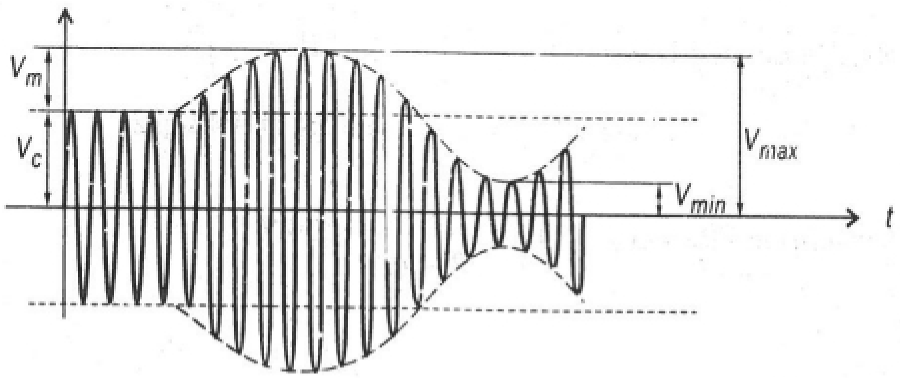


Fig. 3.26

$$V_{max} = V_c + V_m = 13 \text{ V}$$
$$V_{min} = V_c - V_m = 7 \text{ V}$$

(iii) Modulated index

$$m = \frac{V_m}{V_c} = \frac{3}{10} = 0.3$$

$$m = 0.3$$

(iv) Spectrum of modulated wave

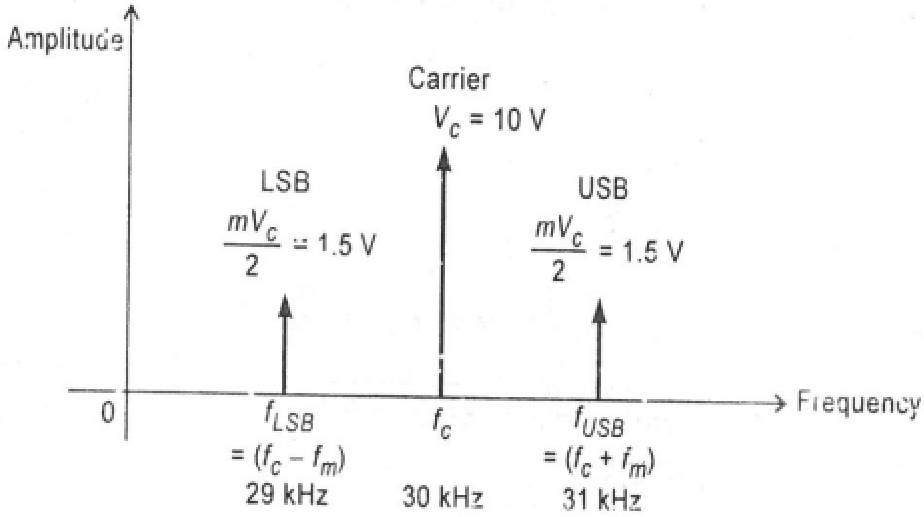


Fig. 3.27

$$f_{USB} = f_c + f_m = 30 + 1 = 31 \text{ kHz}$$

$$f_{LSB} = f_c - f_m = 30 - 1 = 29 \text{ kHz}$$

$$f_{USB} = 31 \text{ kHz}, \quad f_{LSB} = 29 \text{ kHz}$$

$$\text{Amplitude of each sideband} = \frac{mV_c}{2} = \frac{0.3 \times 10}{2}$$

$$\boxed{\text{Amplitude of each sideband} = 1.5 \text{ V}}$$

(v) Total average power

$$\text{Carrier power, } P_C = \frac{V_c^2}{2R} = \frac{10^2}{2 \times 50} = 1 \text{ W}$$

$$\therefore \text{ Total power, } P_T = \left(1 + \frac{m^2}{2}\right) P_c = 1.045 \text{ W}$$

(vi) Power carried by each sideband

$$P_{USB} = P_{LSB} = \frac{m^2 V_c^2}{8R} = \frac{0.3^2 \times 10^2}{8 \times 50}$$

$$\boxed{P_{USB} = P_{LSB} = 0.0225 \text{ W}}$$

∴ Power carried by sidebands

$$P_{SB} = 2P_{USB} = 2P_{LSB} = 0.045 \text{ W}$$

$$\boxed{\therefore P_{SB} = 0.045 \text{ W}}$$

Problem 2 : *A 400 W carrier is simultaneously modulated by two audio waves with modulation percentage of 50 and 60 respectively. What is total sideband power radiated ?*

Given :

$$m_1 = 0.5$$
$$m_2 = 0.6$$
$$P_c = 400 \text{ W}$$

To find : P_{SB}

Solution :

$$P_T = \left(1 + \frac{m_e^2}{2}\right) P_c \text{ and } P_{SB} = \frac{m_e^2}{4} P_c$$

$$m_e = \sqrt{m_1^2 + m_2^2} = \sqrt{0.5^2 + 0.6^2}$$

$$\boxed{m_e = 0.781}$$

$$P_{SB} = \frac{m_e^2}{4} P_c = 61 \text{ W}$$

$$P_{SB} = 61 \text{ W}$$

Problem 3 : *AM transmitter supplies 10 kW of carrier power to a 50 Ω load. It operates at a carrier frequency of 1.2 MHz and is 80 % modulated by a 3 kHz sine wave.*

(i) *Sketch the signal in frequency domain, with frequency and power scales. Show the power in dBW.*

(ii) *Calculate the total average power in signal in watts and dBW.*

(iii) *Calculate RMS voltage of AM signal.*

(iv) *Calculate peak voltage of AM signal.*

Given :

$$P_c = 10 \text{ kW}$$
$$R = 50\ \Omega$$
$$f_c = 1.2 \text{ MHz}$$
$$m = 0.8$$
$$f_m = 3 \text{ kHz}$$

To find : (1) Total power P_T in watts and dBW

(2) RMS voltage V_{rms}

(3) Peak voltage V_c

Solution :

$$P_T = \left(1 + \frac{m^2}{2}\right) P_c = \left(1 + \frac{0.8^2}{2}\right) 10 \times 10^3$$

$$= 13.2 \times 10^3 \text{ W}$$

$$\boxed{\therefore P_T = 13.2 \text{ kW}}$$

Now power in dBW is

$$P_{T(dB)} = 20 \log_{10} P_T$$

$$\boxed{P_{T(dB)} = 82.41 \text{ dBW}}$$

$$P_T = \frac{V_{rms}^2}{R}$$

$$\therefore\ 13.2 \times 10^3 = \frac{V_{rms}^2}{50}$$

$$\therefore\ V_{rms}^2 = 660000$$

$$\boxed{\therefore\ V_{rms} = 812.40 \text{ V}}$$

Similarly

$$P_c = \frac{V_c^2}{2R} = \frac{V_c^2}{2 \times 50}$$

$$\therefore\ V_c^2 = 1000000$$

$$\boxed{\therefore\ V_c = 1000 \text{ V}}$$

Now,

$$m = \frac{V_m}{V_c}$$

$$\boxed{\therefore\ V_m = 800 \text{ V}}$$

$$\text{Peak voltage} = V_c + V_m$$

$$\boxed{\text{Peak voltage} = 1800 \text{ V}}$$

. **Frequency domain representation**

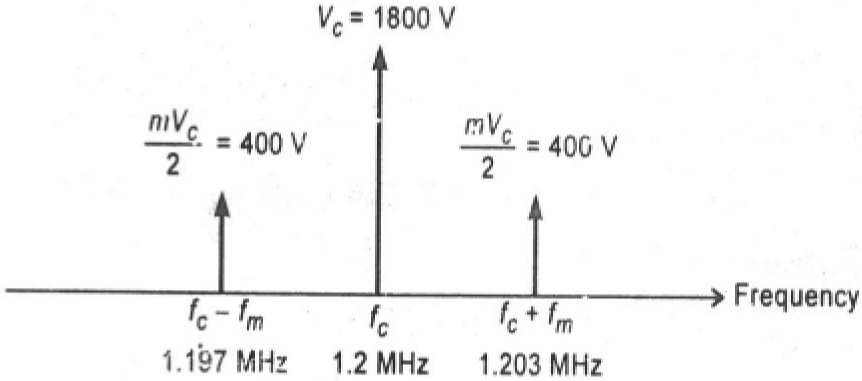


Fig. 3.28

$$f_{USB} = f_c + f_m = 1.2 \text{ MHz} + 3 \text{ kHz} = 1.203 \text{ MHz}$$

$$f_{LSB} = f_c - f_m = 1.2 \text{ MHz} - 3 \text{ kHz} = 1.197 \text{ MHz}$$

$$\text{Amplitude of sideband} = \frac{mV_c}{2} = \frac{0.8 \times 1000}{2}$$

$$\therefore \text{ Amplitude of sideband} = 400 \text{ V}$$

Power spectrum

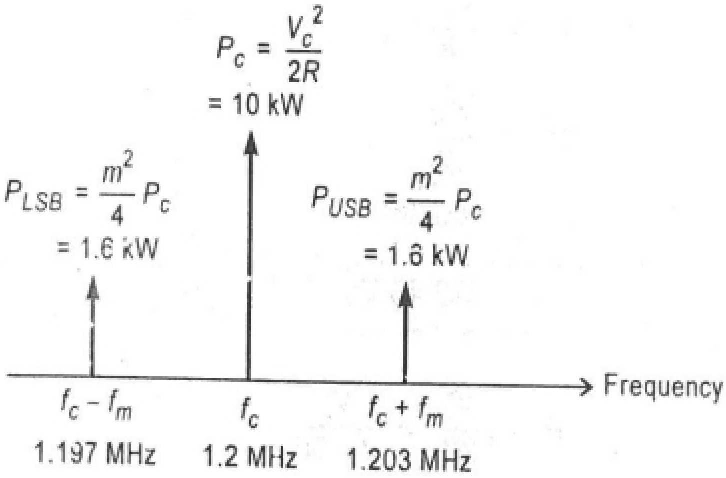


Fig. 3.29

Problem 4 : *An AM signal appears across a 50 Ω load and has following equation.*

(i) *Sketch the envelope of this signal in time domain.*

(ii) *Calculate modulation index, sideband frequencies, total power and bandwidth.*

Given :

$$v(t) = (12 + 12 \sin 12.566 \times 10^3 t) \sin (18.85 \times 10^6 t)$$

$$R = 50\ \Omega$$

To find : (1) Modulation index

(2) f_{USB} and f_{LSB}

(3) P_T

(4) BW

Solution : Given equation is

$$v(t) = (12 + 12 \sin 12.566 \times 10^3 t) \sin (18.85 \times 10^6 t)$$

Comparing given equation with

$$v(t) = (V_c + V_m \sin \omega_m t) \sin \omega_c t$$

We get,

$$V_c = 12 \text{ V},\ V_m = 12 \text{ V}$$

$$\omega_m = 12.566 \times 10^3$$

$$2\pi f_m = 12.566 \times 10^3$$

$$\therefore f_m = 1.999 \times 10^3 \text{ Hz}$$

$$\therefore f_m = 1.99 \text{ kHz}$$

$$\omega_c = 18.85 \times 10^6$$

$$2\pi f_c = 18.85 \times 10^6$$

$$\therefore f_c = 3 \times 10^6 \text{ Hz}$$

$$\therefore f_c = 3 \text{ MHz}$$

(1) Modulation index

$$m = \frac{V_m}{V_c} = 1$$

(2)
$$f_{USB} = f_c + f_m = 3001.99 \text{ kHz}$$

$$f_{LSB} = f_c - f_m = 2998.01 \text{ kHz}$$

(3)
$$P_c = \frac{V_c^2}{2R} = \frac{12^2}{2 \times 50}$$

$$\therefore P_c = 1.44 \text{ W}$$

idth.

$$\therefore \text{Total power } P_T = \left(1 + \frac{m^2}{2}\right) P_c$$

$$\boxed{P_T = 2.16 \text{ W}}$$

(4) Bandwidth $\text{BW} = 2f_m = 2 \times 1.99 \text{ kHz}$

$$= 3.98 \text{ kHz}$$

$$\boxed{\therefore \text{BW} \approx 4 \text{ kHz}}$$

Representation of waveform in time domain

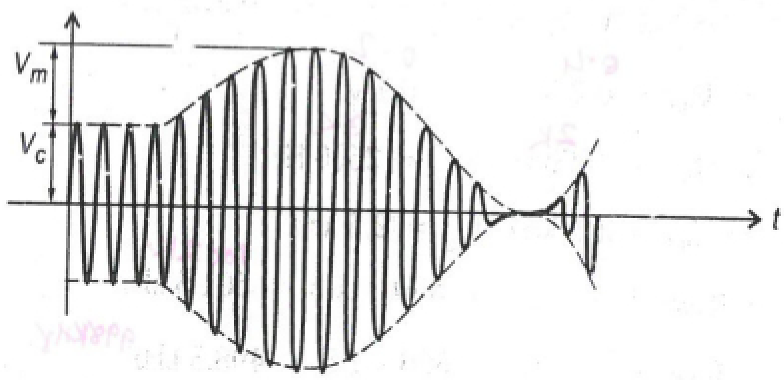


Fig. 3.30

Problem 5 : *A carrier wave $v_c = 4 \sin [2\pi \times 500 \times 10^3 t]$ is amplitude modulated by an audio wave $v_m = 0.2 \sin 3[2\pi \times 500t] + 0.1 \sin 5[2\pi \times 500t]$*

Determine the upper and lower sidebands and sketch spectrum of modulated wave, and find total power in sidebands. Also calculate bandwidth.

Given :
$$v_c = 4 \sin [2\pi \times 500 \times 10^3 t]$$
$$v_m = 0.2 \sin 3[2\pi \times 500t] + 0.1 \sin 5[2\pi \times 500t]$$

To find :
(1) Upper sideband
(2) Lower sideband
(3) Total power in sidebands
(4) Sketch spectrum of modulated wave
(5) Bandwidth

Solution **:** Given equation is
$$v_c = 4 \sin [2\pi \times 500 \times 10^3 t]$$

Comparing given equation with
$$v_c = V_c \sin \omega_c t$$

We get,
$$V_c = 4 \text{ V}$$ Vc= 2V

0.4/2

$$\omega_c = 2\pi \times 500 \times 10^3$$
$$2\pi f_c = 2\pi \times 500 \times 10^3$$
$$\therefore f_c = 500 \times 10^3 = 500 \text{ kHz}$$ 1000KHz
$$\boxed{f_c = 500 \text{ kHz}}$$

0.4 4 0.2 6
$$v_m = 0.2 \sin 3[2\pi \times 500t] + 0.1 \sin 5[2\pi \times 500t]$$

The v_m signal consists of two sine waves.
$$v_1 = 0.2 \sin 3[2\pi \times 500t]$$
$$v_2 = 0.1 \sin 5[2\pi \times 500t]$$

Comparing above two equations with
$$v_m = V_m \sin \omega_m t$$

We get

0.4 0.2
$$V_{m1} = 0.2 \text{ V},\ V_{m2} = 0.1 \text{ V}$$
2K 3K
$$f_{m1} = 1500 \text{ Hz},\ f_{m2} = 2500 \text{ Hz}$$
$$\Rightarrow f_{m1} = 1.5 \text{ kHz},\ f_{m2} = 2.5 \text{ kHz}$$
1002K
$$f_{USB1} = (f_c + f_{m1}) = 500 + 1.5 = 501.5 \text{ kHz}$$
$$f_{LSB1} = (f_c - f_{m1}) = 500 - 1.5 = 498.5 \text{ kHz}$$ 998KHz
$$f_{USB2} = (f_c + f_{m2}) = 500 + 2.5 = 502.5 \text{ kHz}$$ 1003

Syn detector Coh DSB-SC

$$f_{LSB_2} = (f_c - f_{m2}) = 500 - 2.5 = 497.5 \text{ kHz}$$ 997

Modulation index of first signal

$$m_1 = \frac{V_{m1}}{V_c} = \frac{0.2}{4} = 0.05 \text{ V}$$ 0.2

Modulation index of second signal

$$m_2 = \frac{V_{m2}}{V_c} = \frac{0.1}{4} = 0.025 \text{ V}$$ 0.1

Amplitude of USB_1 = Amplitude of LSB_1 $\frac{0.2\times 2}{4}=$ $\frac{m_1 E_c}{2}$

$$= \frac{m_1 V_c}{2} = \frac{0.05 \times 4}{2} = 0.1 \text{ V}$$ 0.2

Amplitude of USB_2 = Amplitude of LSB_2

$$= \frac{m_2 V_c}{2} = \frac{0.025 \times 4}{2} = 0.05 \text{ V}$$ $\frac{0.1\times 2}{2} = \frac{0.2}{2}$

The effective modulation index

$$m_e = \sqrt{m_1^2 + m_2^2} = \sqrt{0.05^2 + 0.025^2}$$

$$\boxed{m_e = 0.0559}$$

Total power in sideband $P_{SB} = \frac{m_e^2 V_c^2}{4R} = \frac{0.0559^2 \times 4^2}{4R}$

$$P_{SB} = \frac{0.0125}{R} \text{ W}$$

Spectrum of AM signal

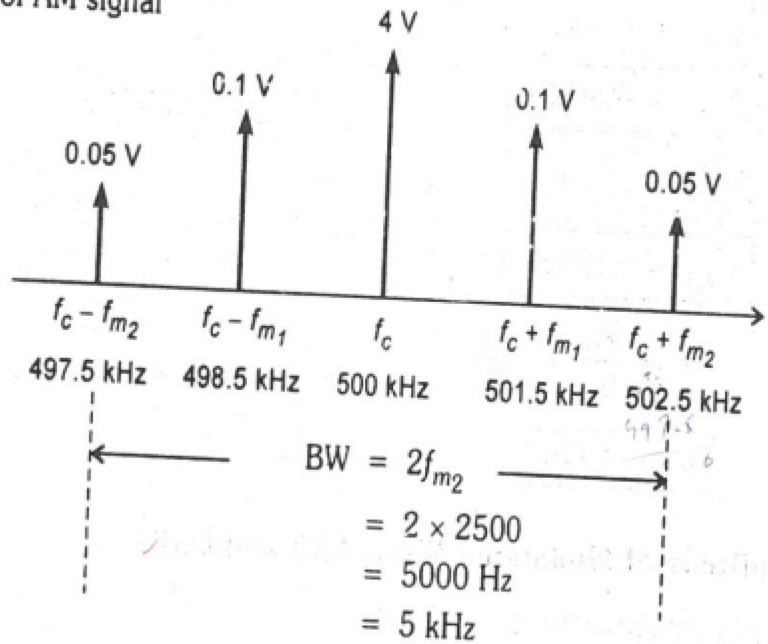


Fig. 3.31

Problem 6 : One input to a conventional AM modulator is a 500 kHz carrier with an amplitude of 20 V_p. The second input is a 10 kHz modulating signal that is of sufficient amplitude to cause a change in the output wave of ±7.5 V_p. Determine :

(i) Side frequencies and modulation index.

(ii) Peak amplitude of the modulated carrier and the upper and lower side frequency voltages.

(iii) Maximum and minimum amplitudes of the envelope.

(iv) Expression for the modulated wave.

(v) Draw the output spectrum and output envelope.

Given :

$$V_c = 20 \text{ V}$$
$$V_m = 7.5 \text{ V}$$
$$f_c = 500 \text{ kHz}$$
$$f_m = 10 \text{ kHz}$$

To find : (i) f_{LSB}, f_{USB} and m

(ii) Peak amplitude of modulated wave, LSB and USB

(iii) V_{max} and V_{min}

(iv) Expression of v_{AM}

(v) Spectrum and envelope.

Solution :

(i) f_{LSB}, f_{USB} and m

$$f_{LSB} = f_c - f_m$$
$$\boxed{f_{LSB} = 490 \text{ kHz}}$$
$$f_{USB} = f_c + f_m$$
$$\boxed{f_{USB} = 510 \text{ kHz}}$$
$$m = \frac{V_m}{V_c} = \frac{7.5}{20}$$
$$\boxed{m = 0.375}$$

(ii) Peak Amplitude of Modulated Wave, LSB and USB

Peak amplitude of modulated wave $= V_c + V_m$

$$= 27.5 \text{ V}$$

Peak amplitude of LSB and USB $= \dfrac{mV_c}{2} = 3.75$ V

(iii) V_{max} and V_{min}

$$V_{max} = V_c + V_m = 27.5 \text{ V}$$
$$V_{min} = V_c - V_m = 12.5 \text{ V}$$

(iv) Expression of υ_{AM}

$$\upsilon_{AM} = V_c \sin \omega_c t + \frac{mV_c}{2} \cos(\omega_c - \omega_m)t - \frac{mV_c}{2} \cos(\omega_c + \omega_m)t$$
$$= V_c \sin 2\pi f_c t + \frac{mV_c}{2} \cos 2\pi(f_c - f_c)t - \frac{mV_c}{2} \cos 2\pi(f_c + f_m)t$$

Substituting V_c, f_c, f_m and m we get,

$$\boxed{\upsilon_{AM} = 20 \sin 1000\,\pi t + 3.75 \cos 980\,\pi t - 3.75 \cos 1020\,\pi t}$$

(v) Spectrum and Envelope

(a) Envelope of AM wave

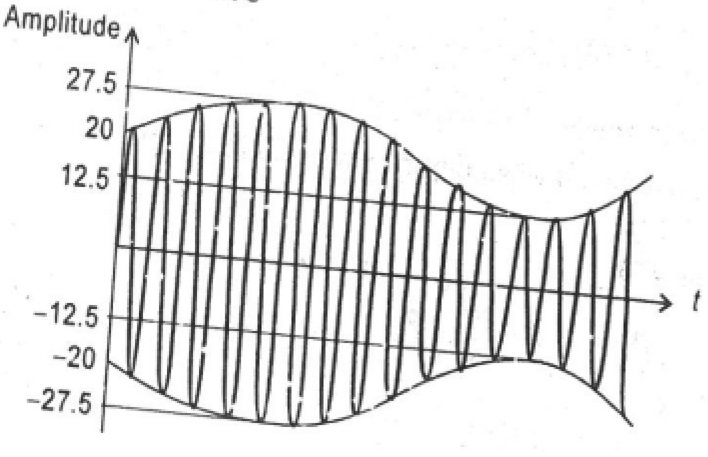


Fig. 3.32

(b) Spectrum of AM wave

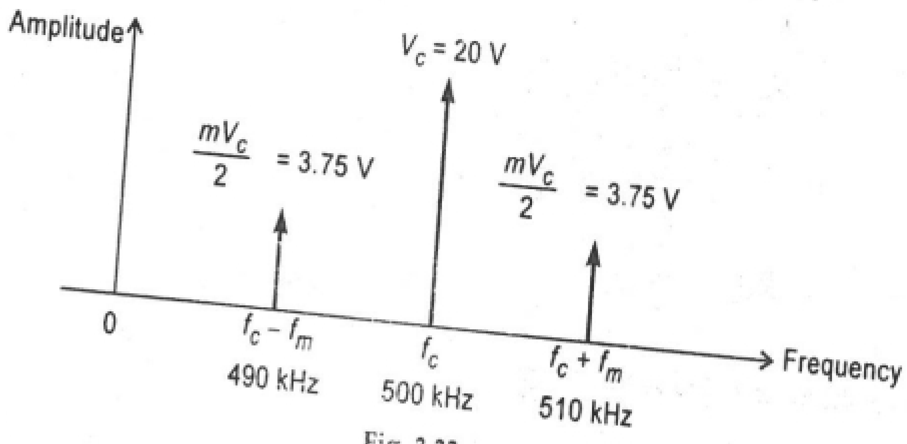


Fig. 3.33

Problem 7 : *A sinusoidal carrier has an amplitude of 10 V and frequency 100 kHz. It is amplitude modulated by a sinusoidal voltage of amplitude 3 V and frequency 500 Hz. Modulated voltage is developed across 75 Ω resistance.*

(i) *Write equation for modulated wave.*

(ii) *Determine modulation index.*

(iii) *Draw spectrum of modulated wave.*

(iv) *Calculate total average power.*

(v) *Calculate power carried by sidebands.*

Solution : Refer problem 1 from this section with

$$f_c = 100 \text{ kHz}$$

$$f_m = 500 \text{ Hz}$$

$$R = 75\,\Omega$$

Problem 8 : *Calculate the percentage power saving when the carrier and upper side band are suppressed in an AM wave modulated to a depth of 100 percent and 50 percent.*

Given : Carrier and USB suppressed

To find : (1) Power saved when depth of modulation = 100

(2) Power saved when depth of modulation = 50.

Solution :(1) m = 1

$$P_T = \left(1 + \frac{m^2}{2}\right) P_c$$

$$\therefore\ P_T = \frac{3}{2} P_c$$

$$\therefore\ P_c = \frac{2}{3} P_T = 0.67\, P_T$$

$$P_{SB} = \frac{m^2}{2}\, P_c$$

$$= \frac{1}{2} P_c = \frac{1}{2} \times \frac{2}{3} P_T$$

$$\therefore\ P_{SB} = \frac{1}{3} P_T$$

$\therefore$ Power saved in one sideband $= \dfrac{1}{6} P_T$

$$\approx 0.16\, P_T$$

∴ Total power saved in suppressing carrier and USB

$$= \frac{2}{3}P_T + \frac{1}{6}P_T$$

$$= 0.83\ P_T$$

∴ Power saved is approximately equal to 83 % of the total power of DSB-FC.

(2) m = 0.5

$$P_T = \left[1 + \left(\frac{1}{2}\right)^2 \times \frac{1}{2}\right] P_c$$

$$\therefore\ P_T = \frac{9}{8}\ P_c$$

$$\therefore\ P_c = \frac{8}{9}\ P_T$$

$$P_{SB} = \frac{m^2}{2}\ P_c$$

$$= \frac{1}{8}\ P_c = \frac{1}{8} \times \frac{8}{9}\ P_T$$

$$\therefore\ P_{SB} = \frac{1}{9}\ P_T$$

∴ Power saved in one sideband $= \frac{1}{18} P_T$

∴ Total power saved in suppressing carrier and USB

$$= \frac{8}{9}\ P_T + \frac{1}{18} P_T$$

$$= 0.9444\ P_T$$

∴ Power saved is approximately equal to 94.4 % of the total power of DSB-FC.

Problem 9 : *The AM transmitter develops an unmodulated power output of 400 watts, across a 50 Ω load. The carrier is modulated by sinusoidal signal with a modulation index 0.8. Assuming F_m = 5 kHz and F_c = 1 MHz.*

(i) Find the V_c and write equation of AM.

(ii) Find the total power of the modulated output.

Solution : Refer problem 3 from this section with change in values as follows :

$$P_c = 400\text{ W}$$

$$R = 50\ \Omega$$

$$m = 0.8$$

$$f_m = 5\text{ kHz}$$

$$f_c = 1\text{ MHz}$$

Problem 10 : *A 5 kW unmodulated carrier is simultaneously modulated by two audio signals with modulations index 80 % and 50 %. Find the transmitted power and effective modulation index. Also find the antenna current with ans without modulating signal assuming antenna resistance of 50 Ωs.*

Given :

$$P_c = 5 \text{ kW}$$
$$m_1 = 0.8$$
$$m_2 = 0.5$$

Antenna Resistance $R = 50\ \Omega$

To find : (1) m_e (3) I_C

(2) Total power P_T (4) I_T

Solution :

The effective modulation index

$$m_e = \sqrt{m_1^2 + m_2^2} = \sqrt{0.8^2 + 0.5^2}$$

$$\boxed{m_e = 0.9434}$$

$$P_T = \left(1 + \frac{m_e^2}{2}\right) P_c = \left(1 + \frac{0.89}{2}\right) (5 \text{ kW})$$

$$\boxed{P_T = 7.225 \text{ kW}}$$

$$P_c = I_C^2 R$$

$$\therefore \ I_C = \sqrt{\frac{P_c}{R}} = \sqrt{\frac{5000}{50}}$$

$$\therefore \ \boxed{I_C = 10 \text{ A}}$$

$$I_T = I_C \sqrt{1 + \frac{m^2}{2}}$$

$$= 10 \sqrt{1 + \frac{0.89}{2}}$$

$$\boxed{I_T = 12.02 \text{ A}}$$

Problem 11 : An AM transmitter supplies 10 kW of carrier power to a 50 ohm load. It operates at a carrier frequency of 1.2 MHz and is 80% modulated by a 3 kHz sine wave.

(i) *Sketch the signal in the frequency domain, with frequency and power scales. Show the power in dBW.*

(ii) *Calculate the total average power in the signal, in watts and dBW.*

(iii) *Calculate the RMS voltage of the signal.*

(iv) *Calculate the peak voltage of the signal.*

Solution : Refer problem 3 from this section.

Problem 12 : An AM transmitter radiates 5 MHz carrier with 60 kW power. Carrier is modulated with 300 Hz and 2 kHz signals.

(i) *What will be the total modulation index if each signal modulates at 70 % modulation ?.*

(ii) *What is the total power transmitted ?*

(iii) *Draw the frequency spectrum of modulated signal.*

(iv) *What is the power content of each spectral component ?*

Given :

$$f_c = 5 \text{ MHz}$$
$$P_c = 60 \text{ kW}$$
$$f_{m_1} = 300 \text{ Hz}$$
$$f_{m_2} = 2 \text{ kHz}$$

To find : (1) m_e if $m_1 = m_2 = 0.7$

(2) P_T

(3) Draw frequency spectrum

(4) Power content of each spectral component.

Solution :

$$m_e = \sqrt{m_1^2 + m_2^2} = \sqrt{0.7^2 + 0.7^2}$$

$$\boxed{m_e = 0.9899}$$

$$P_T = \left(1 + \frac{m_e^2}{2}\right) P_c$$

$$\boxed{P_T = 89.4 \text{ kW}}$$

$$f_{USB_1} = (f_c + f_{m1}) = 5.000300 \text{ MHz}$$

$$f_{LSB_1} = (f_c - f_{m1}) = 4.9997 \text{ MHz}$$

$$f_{USB_2} = (f_c + f_{m2}) = 5.002 \text{ MHz}$$

$$f_{LSB_2} = (f_c - f_{m2}) = 4.998 \text{ MHz}$$

$$P_{LSB_1} = P_{USB_1} = \frac{m_1^2}{4} P_c$$

$$= 0.1225\, P_c \quad = 7.35 \text{ kW}$$

$$P_{LSB_2} = P_{USB_2} = \frac{m_2^2}{4} P_c$$

$$= 0.1225\, P_c \quad = 7.35 \text{ kW}$$

$$P_c = \frac{V_c^2}{2R}$$

$$\Rightarrow V_c = \sqrt{P_c \times 2R}$$

$$= 346.4\sqrt{R}$$

Spectrum of AM signal

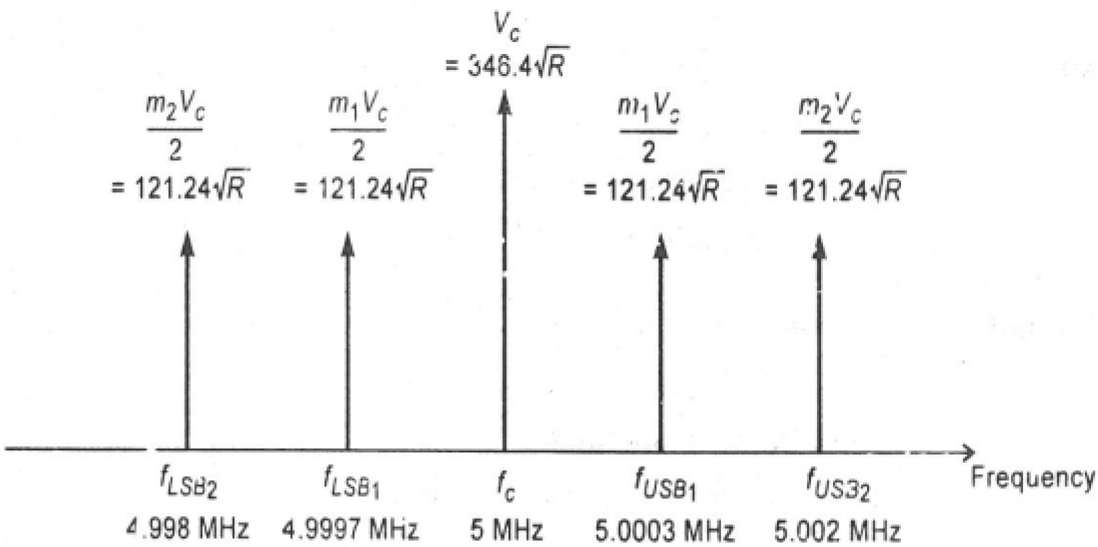


Fig. 3.34

Problem 13 : *The unmodulated carrier power of an AM transmitter is 10 kW and carrier frequency is 2 MHz. The carrier is modulated to a depth of 50 % by an audio signal of 3 kHz.*

(i) *Determine the total power transmitted.*

(ii) *Determine the SSB power.*

(iii) *Percentage of power saving if SSB is transmitted.*

(iv) *Draw the frequency spectrum and find the bandwidth.*

Solution : Refer problem 3 from this section with change in values

$$P_c = 10 \text{ kW}$$
$$f_c = 2 \text{ MHz}$$
$$m = 0.5$$
$$f_m = 3 \text{ kHz}$$

The answer will be in terms of load resistance R as it is not given.

Problem 14 : *The output voltage of a transmitter is given by*

$400(1 + 0.4 \sin 6230\, t) \sin 3.14 \times 10^7\, t$. This voltage is fed to a load of 600 Ω resistance. Determine :

(i) Carrier frequency

(ii) Modulating frequency

(iii) Carrier power

(iv) Total power output

Solution : Refer problem 4 from this section.

Problem 15 : *Draw spectrum of an AM waveform if the modulating signal is*

$$m(t) = (\cos 2000\pi t + 0.5 \cos 4000\, \pi t) \text{ and carrier is}$$

$$c(t) = 1.5 \cos (1000\, \pi t).$$

Also calculate (i) the total and sideband power and (ii) bandwidth.

Solution : Refer problem 5 from this section.

Problem 16 : *An audio frequency signal $10 \sin 2\pi \times 500t$ is used to amplitude modulate carrier of $50 \sin 2\pi \times 10^7 t$. Calculate :*

(i) Modulation index

(ii) Sideband frequencies

(iii) Amplitude of each sideband frequencies

(iv) Bandwidth required

(v) Total power delivered to the load of 600 Ω.

Given :

$$v_m = 10 \sin [2\pi \times 500t] \quad .....(I)$$
$$v_c = 50 \sin [2\pi \times 10^5 t] \quad .....(II)$$
$$R = 600\, \Omega$$

To find : (i) m

(ii) f_{LSB} and f_{USB}

(iii) Amplitudes of sidebands

(iv) Bandwidth

(v) P_T

Solution **:** We have

$$v_m = V_m \sin \omega_m t$$

∴ by comparing with given equation (I)

$$\boxed{V_m = 10 \text{ V}}$$

$$\omega_m = 2\pi \times 500$$

$$2\pi f_m = 2\pi \times 500$$

$$\boxed{\therefore f_m = 500 \text{ Hz}}$$

$$v_c = V_c \sin \omega_c t$$

∴ by comparing with given equation (I)

$$\boxed{V_c = 50 \text{ V}}$$

$$\therefore \omega_c = 2\pi \times 10^5$$

$$2\pi f_c = 2\pi \times 10^5$$

$$\boxed{\therefore f_c = 10^5 \text{ Hz}}$$

Modulation index

$$m = \frac{V_m}{V_c} = \frac{10}{50}$$

$$\boxed{\therefore m = 0.2}$$

$$\therefore f_{USB} = f_c + f_m = 10^5 + 500 = 1.005 \times 10^5 \text{ Hz}$$

$$\boxed{\therefore f_{USB} = 1.005 \times 10^5 \text{ Hz}}$$

$$f_{LSB} = f_c - f_m = 10^5 - 500 = 9.95 \times 10^4 \text{ Hz}$$

$$\boxed{f_{LSB} = 9.95 \times 10^4 \text{ Hz}}$$

$$\text{Amplitudes of sidebands} = \frac{mV_c}{2} = 5 \text{ V}$$

4 TYPES OF AM

Topic	Theory imp	Oral imp
Introduction		
Balanced Modulator	☆☆☆	☆
Ring Modulator	☆	☆
Filter Method	☆	☆
Phase Shift Method	☆☆☆	
Third Method/Weaver Method	☆	☆☆
Product Detector	☆	
Balanced Demodulator	☆☆	☆
Pilot Carrier System	☆☆☆☆	☆☆☆
VSB	☆☆	☆☆☆
ISB	☆	☆☆☆
FAQ's	☆☆	☆☆☆☆
Problems	--	--

4.0 Introduction

In the previous chapter, we have seen the basic form of an AM signal in detail. This basic form is called the *AM signal* or *Double Side Band-Full Carrier(DSB-FC) system* because it contains two side bands and the carrier signal.

But this is not the only type of AM system. The different types of AM systems are :

- DSB-FC - Double Side Band - Full Carrier System
- DSB-SC - Double Side Band - Suppressed Carrier System
- SSB-SC - Single Side Band - Suppressed Carrier System
- ISB - Independent Side Band System
- VSB - Vestigial Side Band System

Power Wastage or Disadvantages of DSB-FC

- The transmission efficiency of DSB-FC AM transmitter is very poor because the maximum efficiency that can be achieved is only 33.33 %.

- The remaining power is lost in the carrier signal, which does not carry any useful information.
- Also, we know that both the sidebands carry the same information as the two are equal in magnitude and equidistant from the location of the carrier signal on a frequency spectrum.
- The system is susceptible to noise.

This is the reason, we go for other AM systems.

4.1 DSB-SC System

Equation of DSB-SC Signal

Referring to the equation of a normal DSB-FC signal given in previous chapter,

$$v_{\text{DSB-FC}} = V_c \sin \omega_c t + \frac{m_a V_c}{2} \cos (\omega_c - \omega_m)t - \frac{m_a V_c}{2} \cos (\omega_c + \omega_m)t$$

In DSB-SC signal, the carrier signal is suppressed and is not transmitted. The AM wave consists of only LSB and USB.

∴ The equation of DSB-SC can be given as :

$$v_{\text{DSB-SC}} = \underbrace{\frac{m_a V_c}{2} \cos (\omega_c - \omega_m)t}_{\text{LSB}} - \underbrace{\frac{m_a V_c}{2} \cos (\omega_c + \omega_m)t}_{\text{USB}}$$

Power Saving in DSB-SC

$$P_T = P_c\left(1 + \frac{m^2}{2}\right)$$

If $m = 1$

$$P_T = P_c\left(\frac{3}{2}\right)$$

$$P_c = \left(\frac{2}{3}\right) P_T = 0.6666\, P_T$$

Thus 67% of total power is consumed by the carrier.

With this system, we save about 67% of the power when $m = 1$.

Frequency Spectrum of DSB-SC

Figure 4.1 shows the frequency domain representation of a DSB-SC signal.

- The frequency spectrum of a DSB-SC signal contains only two sidebands.
- The suppressed carrier is shown by dotted line.
- The LSB is located at $(f_c - f_m)$.
- The USB is located at $(f_c + f_m)$.

- The amplitude of the two sidebands is $\dfrac{m_a V_c}{2}$.

As it can be clearly seen from the *figure 4.1*, the bandwidth of the DSB-SC signal is :

$$BW_{DSB\text{-}SC} = (f_c + f_m) - (f_c - f_m)$$

$$BW_{DSB\text{-}SC} = 2f_m.$$

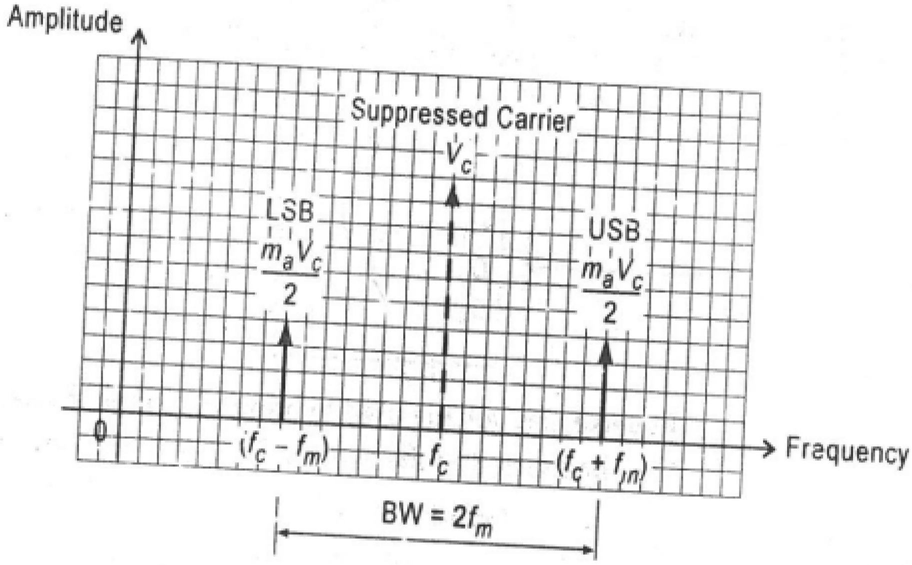


Fig. 4.1

The circuits used for generation of DSB-SC signals are :

(1) Balanced Modulator

(2) Balanced Ring Modulator

4.1.1 Balanced Modulator

Q.1. Draw a neat sketch and explain the working of balance modulator.

Q.2. Prove that a balance modulator suppresses the carrier.

The balanced modulator uses non linear device like diode, JFET in its circuit whose transfer function is given by

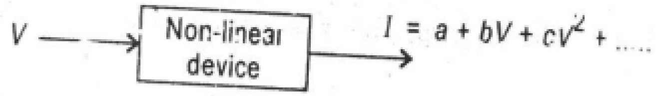


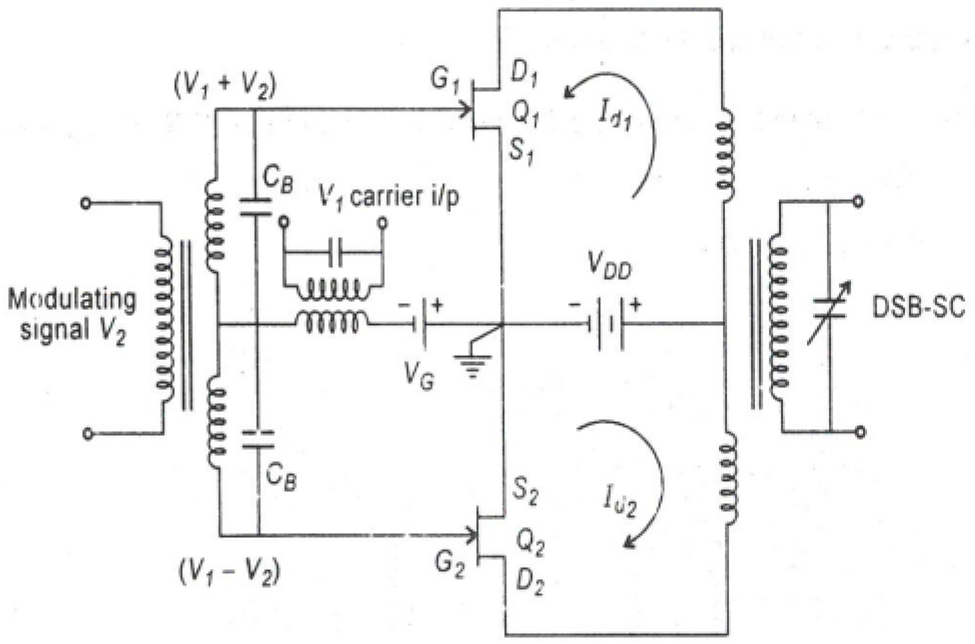
Fig. 4.2

$$I = a + bV + cV^2 + ... \qquad ..... (1)$$

where V = Input voltage

I = Output current.

Balanced Modulator using two JFET's is shown in the *figure 4.3*. It takes audio signal and carrier as two inputs.



V_G : Reverse bias voltage used to bring Q point at the centre of load line such that the two JFET's operate in class A mode

Fig. 4.3

Operation

- The carrier voltage is applied to two gates in phase.
- Due to centre tapped transformer at the input, the audio voltage applied to the two gates is 180° out of phase with each other.
- Let V_1 = Carrier signal
  V_2 = Audio signal, be the two inputs.

  ∴ Input to JFET $Q_1 = V_1 + V_2 = V_{G1}$

  ∴ Input to JFET $Q_2 = V_1 - V_2 = V_{G2}$

- The two currents I_{d1} and I_{d2} flowing in primary winding of output transformer are in opposite directions.
- If the two FETs are perfectly balanced, the carrier frequency will be completely cancelled.
- The capacitor used C_B in secondary of input transformer is RF filter and capacitor used in output transformer is used to filter out unwanted frequencies.
- Practically it is difficult to set perfectly balanced FETs hence the carrier signal is heavily suppressed.
- The output of Balance Modulator contains two sidebands and an unwanted component that can be cancelled out by tuning output transformer.
- Thus, the output of Balanced Modulator contains only two sidebands i.e. DSB-

Mathematical Analysis

$$\text{Let } V_1 = \text{Carrier signal} = V_c \sin \omega_c t$$

$$V_2 = \text{Audio signal} = V_m \sin \omega_m t$$

Now, according to equation (1)

$$I_{d_1} = a + bV_{G_1} + cV_{G_1}^2 \quad \ldots\ldots (2)$$

$$I_{d_2} = a + bV_{G_2} + cV_{G_2}^2 \quad \ldots\ldots (3)$$

$$\text{Let } V_{G_1} = V_1 + V_2 \rightarrow \text{Input to JFET } Q_1$$

$$V_{G_2} = V_1 - V_2 \rightarrow \text{Input to JFET } Q_2$$

$$\therefore I_{d_1} = a + b(V_1 + V_2) + c(V_1 + V_2)^2 \text{ and}$$

$$I_{d_2} = a + b(V_1 - V_2) + c(V_1 - V_2)^2$$

Lets assume that the two JFETs matched.

$$\therefore I_{d_1} = a + bV_1 + bV_2 + cV_1^2 + 2cV_1V_2 + cV_2^2 \text{ and}$$

$$I_{d_2} = a + bV_1 - bV_2 + cV_1^2 - 2cV_1V_2 + cV_2^2$$

$\therefore$ The resultant current in primary winding of output transformer is,

$$I = I_{d_1} - I_{d_2}$$

$$= 2bV_2 + 4cV_1V_2$$

$$= 2b\,(V_m \sin \omega_m t) + 4c\,(V_c \sin \omega_c t \cdot V_m \sin \omega_m t)$$

$$= 2bV_m\,(\sin \omega_m t) + 4cV_mV_c\,(\sin \omega_c t \cdot \sin \omega_m t)$$

$$= 2bV_m\,(\sin \omega_m t) + 4cV_mV_c \left\{\frac{1}{2}[\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t]\right\}$$

$$I = 2bV_m\,(\sin \omega_m t) + 2cV_mV_c\,[\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t] \quad \ldots\ldots (4)$$

Now, the output voltage at secondary winding is proportional to primary current

$$\therefore V_o \propto I$$

$$= kI$$

$$\therefore V_o = k\left\{2bV_m \sin \omega_m t + 2cV_mV_c\,[\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t]\right\}$$

$$\text{Let } A = 2kb\,V_m$$

$$\text{and } B = 2kc\,V_mV_c$$

$$\therefore V_o = \underbrace{A \sin \omega_m t}_{\text{Modulating signal}} + \underbrace{B \cos(\omega_c - \omega_m)t}_{\text{LSB}} - \underbrace{B \cos(\omega_c + \omega_m)t}_{\text{USB}} \quad \ldots\ldots (5)$$

If the secondary winding of output transformer is tuned then modulating frequency can be filtered out from above equation, so the resultant contains only two sidebands.

4.1.2 Balanced Ring Modulator

Q. Sketch the circuit and explain the working of a balanced ring modulator. Give suitable waveforms.

We have studied that Balanced Modulator is used to suppress the carrier signal and the modulating signal is eliminated by tuning the secondary winding of output transformer.

If we want to suppress both modulating signal and carrier signal such that output contains only the two sidebands then we use *Double Balanced Modulator* or "*Balanced Ring Modulator*.

The circuit diagram is shown in *figure 4.4*.

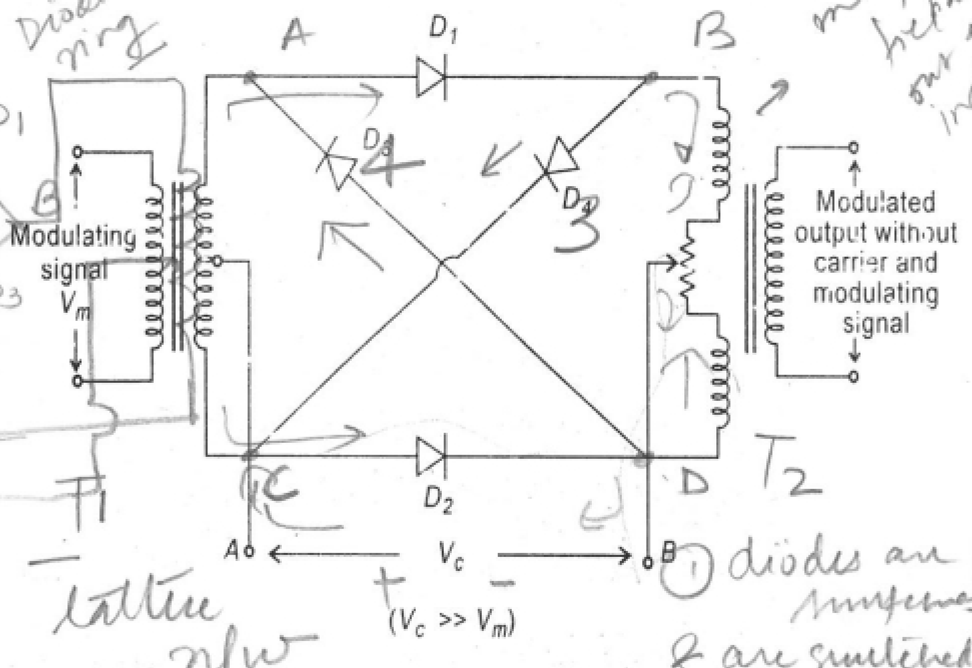


Fig. 4.4

diodes are perfect switches & are switched on & off by the RF carrier signal $V_c >> V_m$

Operation

- In the above circuit the carrier signal V_c is much larger than V_m. Hence it works as switching signal.
- During positive half cycle of carrier signal :
  (1) Point *A* becomes positive w.r.t. *B*.
  (2) Hence, diodes D_1 and D_2 conduct and acts as short circuit.
  (3) Diodes D_3 and D_4 are reverse biased, hence acts as open circuit.

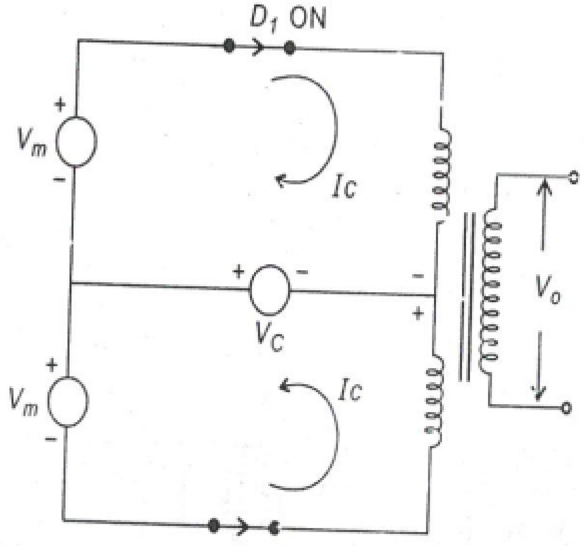


Fig. 4.5

(4) The modulating signal appears at the output with some polarity only during positive half cycle of carrier signal.

- During negative half cycle of carrier signal.

(1) Point B becomes positive with respect to point A.

(2) Hence diodes D_3 and D_4 conduct and acts as short circuit.

(3) Diodes D_1 and D_2 are reverse biased. Hence acts as open circuit.

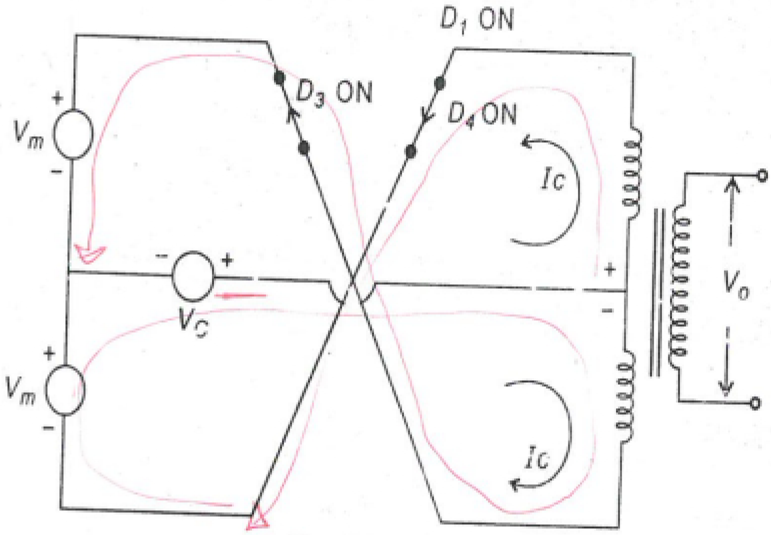


Fig. 4.6

(4) The modulating signal appears at output with reversed polarity during negative half cycle of carrier signal.

- The output waveforms are shown in *figure* 4.7.

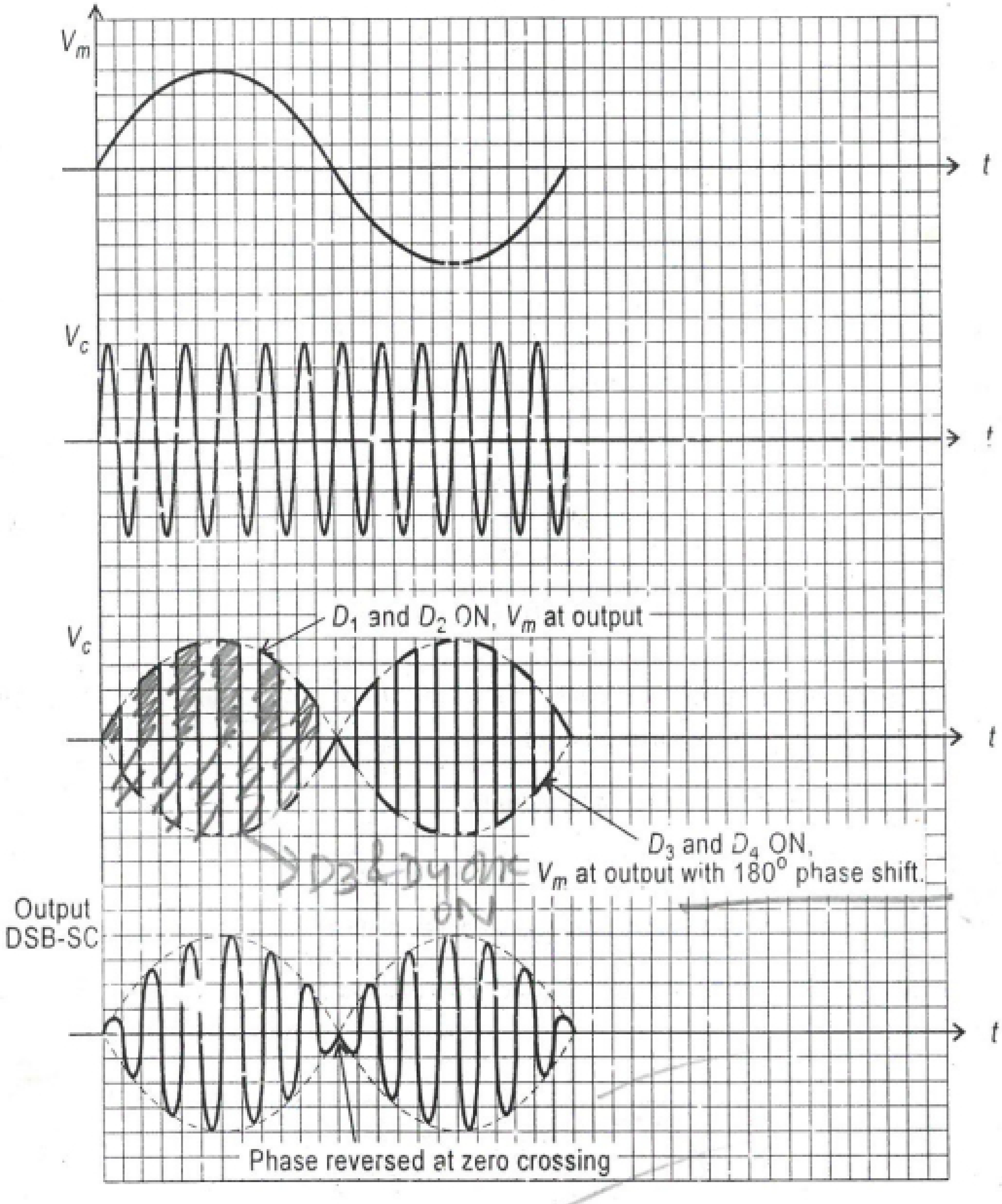


Fig. 4.7

- The output waveform shows that only sidebands appear at output.
- Both the modulating signal and carrier signal are suppressed.

4.2 SSB-SC

Q. *Explain any one method of SSB signal generation.*

Equation of SSB-SC Signal

Equation of SSB-SC signal can be obtained from equation of DSB-FC signal.

$$v_{SSB-L} = \frac{m_a V_c}{2} \cos(\omega_c - \omega_m)t \quad \text{and} \quad v_{SSB-U} = \frac{m_a V_c}{2} \cos(\omega_c + \omega_m)t$$

Power Saving in SSB-SC

Power saved in SSB signal is the total power saved in suppressing carrier and one sideband. The total power saved is calculated as :

- The power saved in suppressed carrier = 66.66%.
- The power saved in one sideband = 16.66%.

% Power saved in SSB = 83.32%.

Frequency Spectrum of SSB-SC

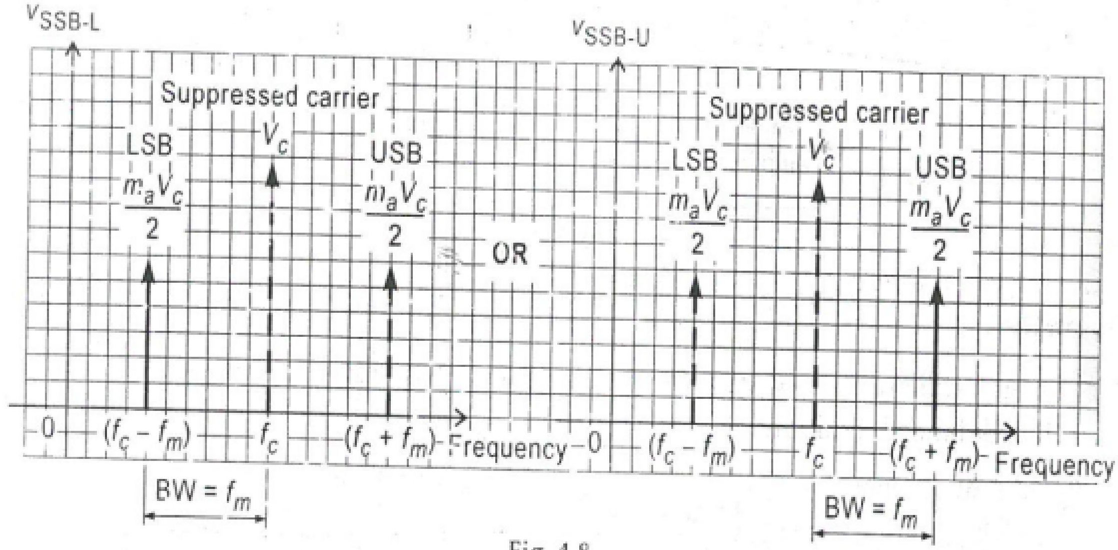


Fig. 4.8

Advantages

- Power requirement is low.
- Bandwidth requirement is low.
- Carrier Interference is low.

SSB-SC Generation

The generation involves following stages :

(1) Generation of DSB-SC.

(2) Generation of SSB-SC : Once DSB-SC signal is generated, the SSB-SC signal can be generated by removing one of the sidebands. This can be done by following ways :
- Filter Method
- Phase Shift Method.
- Third Method / Weavers Method

4.2.1 Filter Method

Q. *Write short note on Single Side Band transmitter using Filter method.*

This method is the simplest method to remove the unwanted sidebands, so as to produce the SSB-SC Signal. The Filters can be of the various types LC, crystal, ceramic or mechanical depending upon the carrier frequency.

The block diagram is shown in *figure 4.9*.

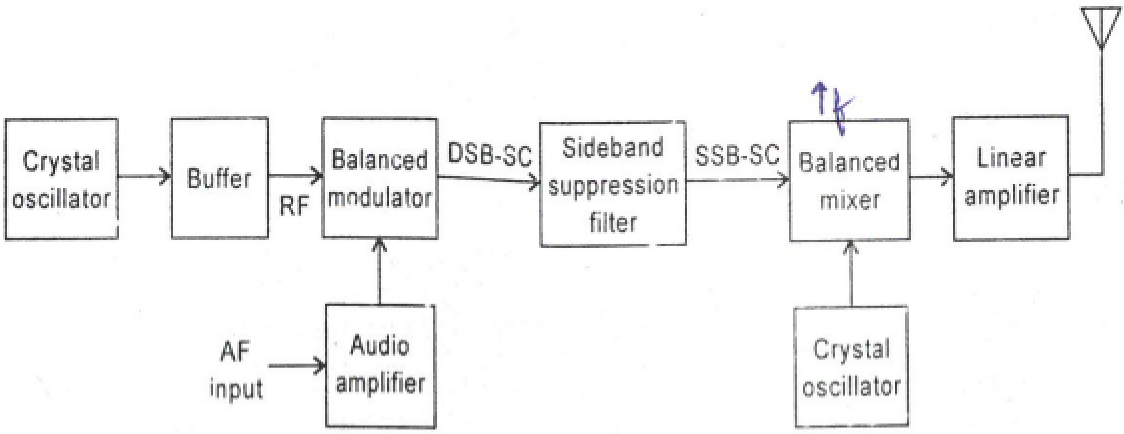


Fig. 4.9

Operation

- Crystal oscillator is used to generate the carrier frequency which is given as input to the Balanced modulator.
- AF signal is given to the Balanced Modulator as second input through Audio Amplifier.
- The Balanced Modulator then generates the DSB-SC signal.
- The output of the Balanced Modulator i.e. DSB-SC signal is given to the Sideband Suppression filter to suppress one sideband and generate the SSB-SC.
- To increase the frequency of the generated SSB-SC signal, Balanced Mixers are used.
- The balanced mixer is followed by linear amplifier to amplify the SSB signal and increase its power level before the final transmission.

Advantages

- The system is simple in working.

Disadvantages

- Very low audio frequencies can not be used because it is difficult to design filter having very small bandwidth .
- Cost increases as expensive filters are used to filter out one of the sidebands.

Uses/Applications

- In spite of the above disadvantages, the system is good for generating SSB signal of good communication quality. Hence it is used in majority of commercial systems.

4.2.2 Phase Shift Method/ Phase Cancellation Method

Q.1. Explain phase cancellation method of SSB generation. Also specify how USB and LSB generation can be done.

Q.2. Draw the block diagram of a phase cancellation SSB generator and explain how the carrier and the unwanted sideband are suppressed. What changes are necessary to supress the other sideband?

Q.3. Draw the block diagram of a phase cancellation SSB generator and explain how the carrier and the unwanted sideband are suppressed.

Q.4. Explain phase shift method of SSB generation.

The block diagram of phase shift method is as shown in *figure 4.10*.

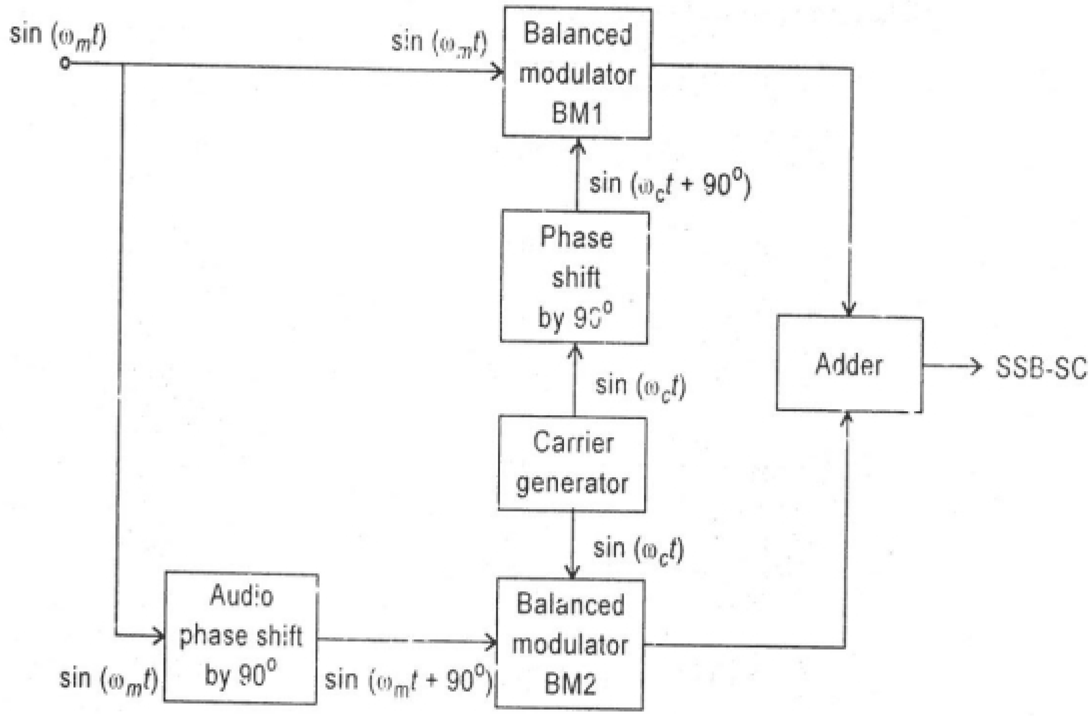


Fig. 4.10

Circuit Description

- Referring to the *figure 4.10*, it is clearly seen that the system consists of following components : Two Balanced modulator, Two phase shifters, one Carrier generator and an Adder.
- The inputs to the first modulator, lets say BM1, are modulating signal and carrier signal shifted by 90°.
- The inputs to the second modulator, lets say BM2, are carrier signal and modulating signal shifted by 90°.
- The output of both the Balanced Modulators are added to cancel out one of the sidebands so that the SSB signal is obtained at the adder output.

Let us see the mathematical equations that will help you to understand the working, of the system.

The input to BM1 is $\sin \omega_m t$ and $\sin (\omega_c t + 90^\circ)$

The output of BM1 is the product given by,

$$v_1 = \sin (\omega_c t + 90^\circ) \cdot \sin \omega_m t$$

$$v_1 = \frac{1}{2} [\cos (\omega_c t + 90^\circ - \omega_m t) - \cos (\omega_c t + 90^\circ + \omega_m t)]$$

$$v_1 = \frac{1}{2} \left\{ \cos [(\omega_c t - \omega_m t) + 90^\circ] - \cos [(\omega_c t + \omega_m t) + 90^\circ] \right\}$$

The input to BM2 is $\sin \omega_c t$ and $\sin (\omega_m t + 90^\circ)$

The output of BM2 is the product given by,

$$v_2 = \sin \omega_c t \cdot \sin (\omega_m t + 90^\circ)$$

$$v_2 = \frac{1}{2} \left\{ \cos [\omega_c t - (\omega_m t + 90^\circ)] - \cos [\omega_c t + (\omega_m t + 90^\circ)] \right\}$$

$$v_2 = \frac{1}{2} \left\{ \cos [(\omega_c t - \omega_m t) - 90^\circ] - \cos [(\omega_c t + \omega_m t) + 90^\circ] \right\}$$

The output of adder is

$$\begin{aligned} v_o &= v_1 + v_2 \\ &= \frac{1}{2} \left\{ \cos [(\omega_c t - \omega_m t) + 90^\circ] - \cos [(\omega_c t + \omega_m t) + 90^\circ] \right\} \\ &\quad + \frac{1}{2} \left\{ \cos [(\omega_c t - \omega_m t) - 90^\circ] - \cos [(\omega_c t + \omega_m t) + 90^\circ] \right\} \\ &= \frac{1}{2} \left\{ \cos [(\omega_c t - \omega_m t) + 90^\circ] + \cos [(\omega_c t - \omega_m t) - 90^\circ] \right\} \\ &\quad - \frac{1}{2} \left\{ \cos [(\omega_c t + \omega_m t) + 90^\circ] + \cos [(\omega_c t + \omega_m t) + 90^\circ] \right\} \end{aligned}$$

The terms $\cos [(\omega_c t - \omega_m t) + 90^\circ]$ and $\cos [(\omega_c t - \omega_m t) - 90^\circ]$ are at 180° phase shift $\therefore$ they cancel each other.

$$\therefore v_o = -\frac{1}{2} \left\{ 2 \cos [(\omega_c t + \omega_m t) + 90^\circ] \right\}$$

$$\therefore v_o = - \cos [(\omega_c t + \omega_m t) + 90^\circ]$$

$$\therefore v_o = \sin (\omega_c t + \omega_m t) \qquad \text{.... } [\because \cos(\theta + 90^\circ) = -\sin \theta]$$

$$\therefore v_o = \text{USB}$$

We can see that one of the sidebands is filtered out such that output contains only upper sideband.

Advantages

- It is very easy to switch from one sideband to other i.e. between LSB and USB.
- It is useful to generate SSB at any frequency even at low frequencies.

Disadvantages

- If the Phase Shifter provides a phase change other than 90°, then particular frequency will not be removed completely from the unwanted sidebands.
- The audio- phase shifter is more complex device since it has to work over a large frequency range.
- The two Balanced Modulators used also should match to avoid incomplete cancellation of frequencies.

To Obtain Lower Side Band

To obtain Lower Side Band at the adder, we should make following changes in the circuit.

(1) The audio input $\sin \omega_m t$ is to be given to the second Balanced Modulator (BM_2).

(2) The phase shifted audio input $\sin (\omega_m t + 90^{\circ})$ should be given to first Balanced Modulator (BM1).

4.2.3 Third Method / Weavers Method

Q. *Explain the THIRD method to generate SSB AM wave.*

This method was developed by D.K. Weaver, hence the name

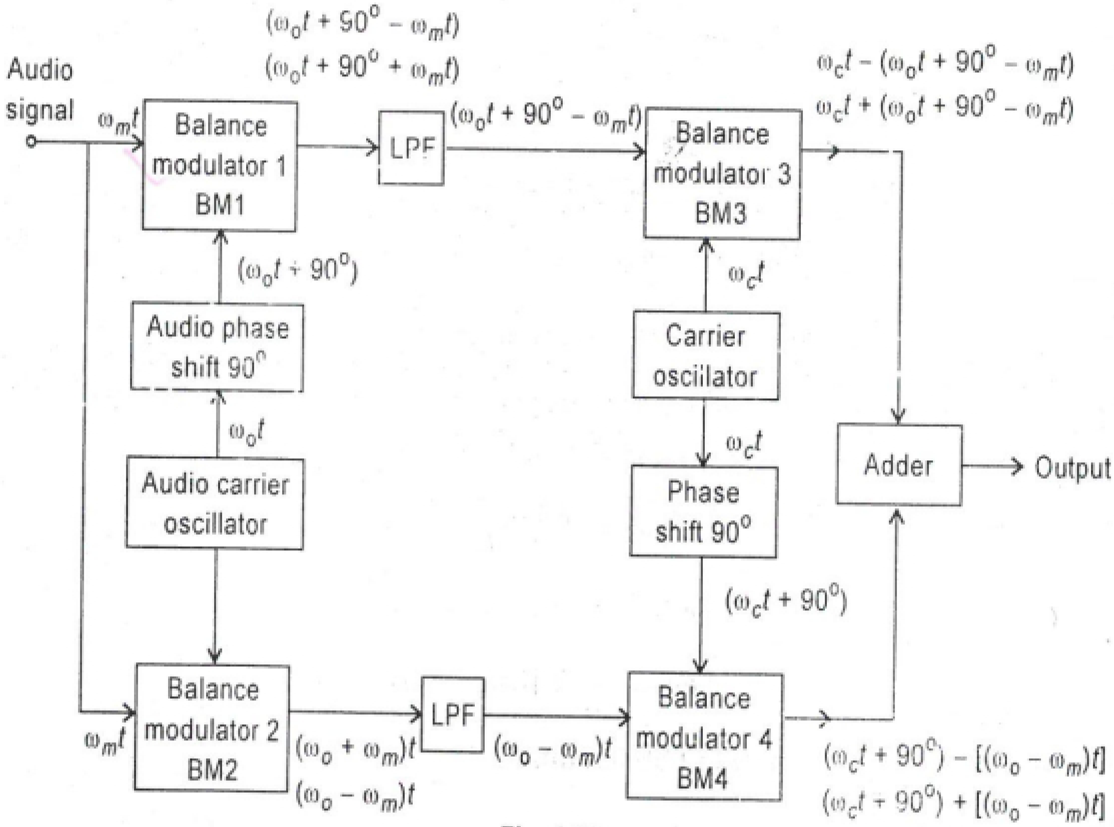


Fig. 4.11

Circuit Description

- The above method is very much similar to phase shift method.
- This method uses four balanced modulators, two carrier generators, two phase shifters and an adder.
- This method has all the advantages of the phase shift method.
- Unlike phase shift method, in this method the modulating signal does not undergo any phase shift.
- An intermediate frequency (ω_o) is generated and applied to BM1 and 90^o phase shifted signal ($\omega_o + 90^o$) is applied to BM2.
- Modulating signal is applied to both BM1and BM2.
- Now, let us see the mathematical proof

Output of BM1

$$\omega_o t + 90^o \pm \omega_m t$$

Output of BM2

$$(\omega_o \pm \omega_m)t$$

Output of BM3

$$\omega_c t - \omega_o t - 90^o + \omega_m t = [(\omega_c - \omega_o) + \omega_m]t - 90^o \quad \text{..... (1)}$$

$$\omega_c t + \omega_o t + 90^o - \omega_m t = [(\omega_c + \omega_c) - \omega_m]t + 90^o \quad \text{..... (2)}$$

Output of BM4

$$\omega_c t + 90^o - \omega_o t + \omega_m t = [(\omega_c - \omega_o) + \omega_m]t + 90^o \quad \text{..... (3)}$$

$$\omega_c t + 90^o + \omega_o t - \omega_m t = [(\omega_c + \omega_o) - \omega_m]t + 90^o \quad \text{..... (4)}$$

In output of BM3 and BM4, terms (1) and (3) are at 180^o phase shift

∴ The two terms cancel each other

∴ At the output of adder we get addition of remaining two terms as :

$$\text{Output} = 2\left\{[(\omega_c + \omega_o) - \omega_m]t + 90^o\right\}$$

$$\text{Let } \omega_c' = \omega_c + \omega_o$$

$$\therefore \text{Output} = 2\,[(\omega_c' - \omega_m)t + 90^o] = \text{LSB}$$

Hence output is SSB signal with LSB in output.

Advantages

- It can generate SSB signal at any frequency, even at low frequencies.
- After modulation, amplifiers are not required for up conversion.

Disadvantages

- Difficult to match all balanced modulators.
- Very complex, not practically used.

4.3 SSB Demodulator

The detectors used for DSB-FC can not be used for detection of SSB signal because the amplitude of SSB signal is constant.

We use following detectors.

4.3.1 Product Detector

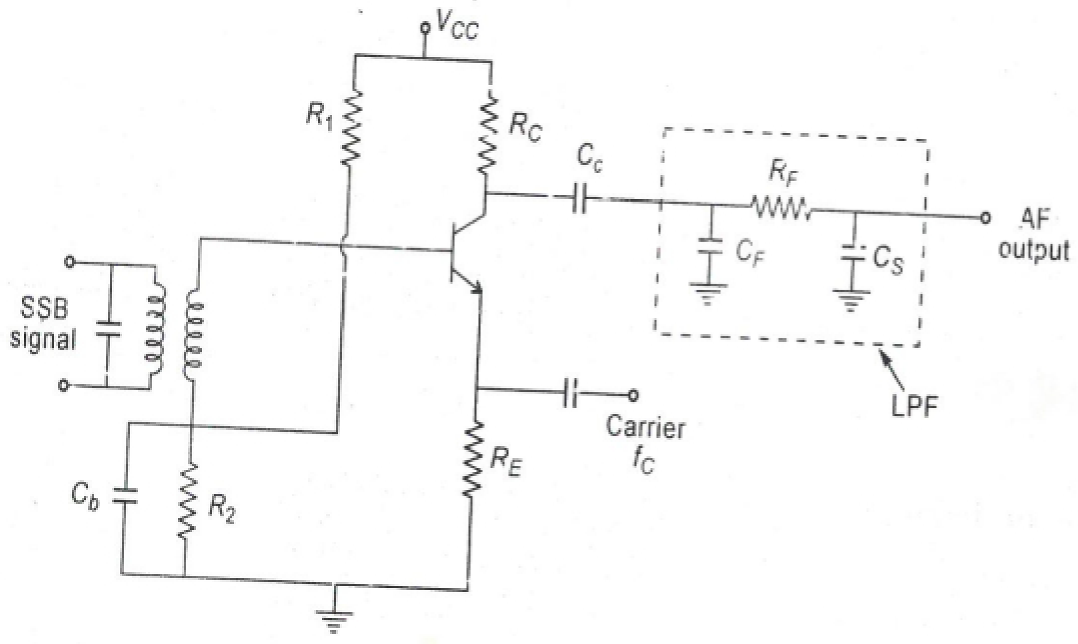


Fig. 4.12

Operatoin

- The SSB signal is applied to the base of transistor through transformer.
- The carrier signal is given to the emitter of transistor.
- The circuit acts as a mixer and transistor is operated in linear region.
- We get sum and difference frequency component of two input signals at the output of collector along with modulating signal.
- The modulating signal is eliminated by passing through LPF formed by R_F, C_F and C_S. Thus at the output, we get modulating signal.
- This filter rejects other high frequencies.

4.3.2 Balanced Demodulator

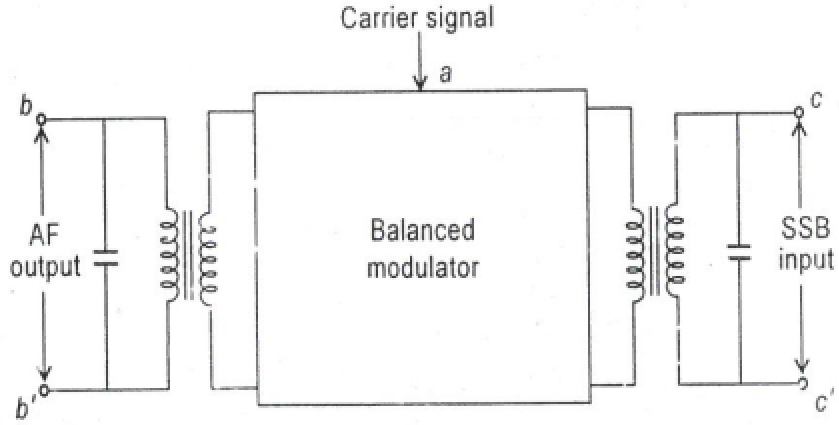


Fig. 4.13

- A balanced modulator which is used to generate DSB-SC can also be used as SSB-SC detector.
- According to the concept of balanced modulator if we apply ω_m and ω_c to balanced modulator then at the output we get $\omega_c + \omega_m$, $\omega_c - \omega_m$ and ω_m.
- Similarly, we apply SSB input, say USB i.e. $\omega_c + \omega_m$ at c-c' and the carrier signal ω_c at a.
- At b-b' we get the output as $2\omega_c + \omega_m$, $\omega_c + \omega_m$, ω_m and multiples of these frequency components.
- By tuning the capacitor to modulating frequency ω_m, we can get the modulating signal across b-b'.
- Hence, at b-b' we get the required modulating signal or the AF output.

Applications

Since balanced modulator using diode is used in generation and reception of SSB-SC, it is used in SSB-SC tranceivers (i.e. transmitter + receiver) to reduce number of components.

4.4 SSB Extensions - Pilot Carrier System Requirement

Q.1. What is the advantage of transmitting a pilot carrier in SSB modulation ?

Q.2. Explain in brief : Pilot Carrier System.

In transmission of SSB signal, we do not transmit the carrier. Hence the demodulation of the signal becomes quite difficult. The receiver uses very stable carrier frequency. Therefore if the transmitted signal drifts in frequency even by small amount, then the demodulated output also drifts by the same amount. Hence, the demodulation becomes difficult.

Therefore to avoid this either the carrier transmitted should not drift or some part of the carrier should be transmitted with the SSB signal.

Hence, we transmit the reduced carrier along with the SSB signal. This system is called *Pilot Carrier System*.

Pilot Carrier SSB Transmitter

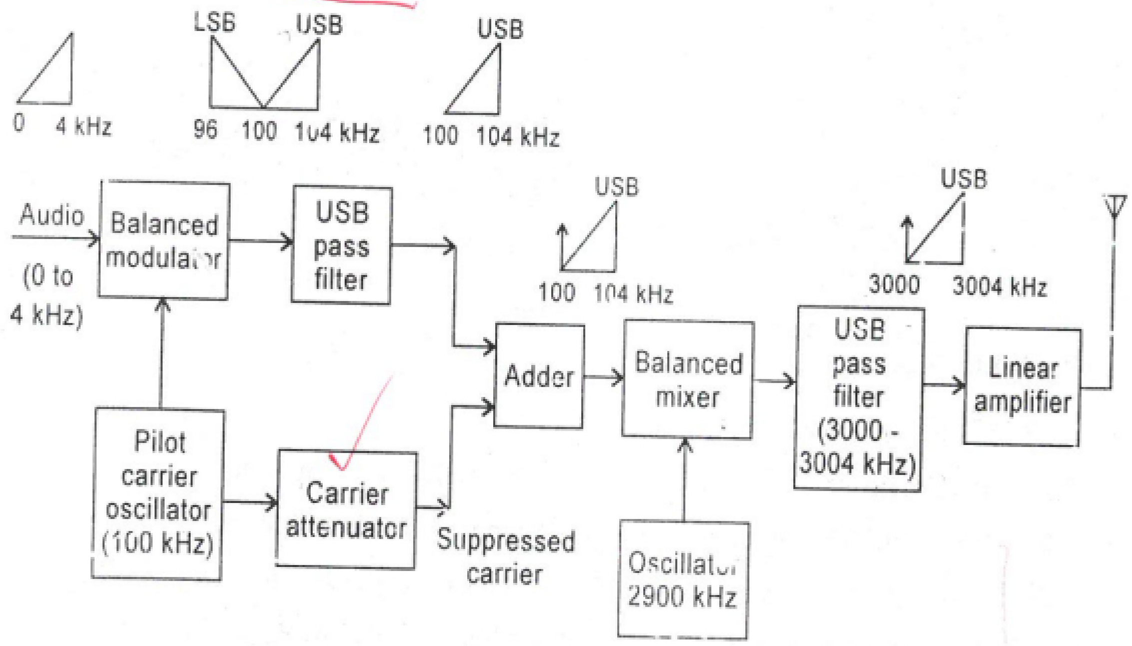


Fig. 4.14

Operation

- As shown in *figure 4.14* transmitter system, the inputs to the balanced modulator are audio signal and carrier signal.
- The balanced modulator produces two side bands where USB = 100 to 104 kHz and LSB = 100 to 96 kHz.
- At the output of USB pass filter, we get only the USB.
- According to Pilot Carrier System, the reduced pilot carrier is inserted and at the output of Adder we get SSB signals along with reduced carrier.
- Balanced mixer along with the 2900 kHz frequency oscillator is used to boost the signal given by the adder.
- The output of balanced mixer consists of sum and difference frequency components.
- The USB pass filter allows to pass only the upper side band frequency components.
- The linear amplifiers are used to increase the power level of the signal before transmission.
- Hence the transmission of SSB signal is with large power and at high frequency.

Pilot Carrier Receiver

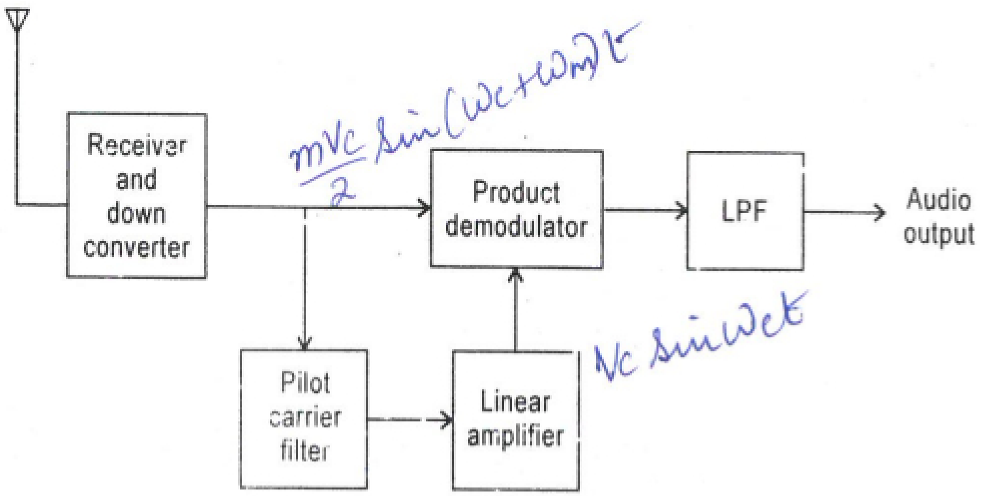


Fig. 4.15

Operation

- The SSB signal is received at the receiver.
- The Pilot Carrier is filtered out and given to the Linear Amplifier.
- The Linear Amplifier amplifies the carrier signal and filters out the noise.
- The received signal and amplified carrier is given to the Product Demodulator.
- At the output of the Product Demodulator we get two frequency components as follows :

$$\left(\frac{mV_c}{2}\sin(\omega_c+\omega_m)t\right)\cdot V_c\sin\omega_c t$$

$$= \frac{mV_c^2}{2}\sin(\omega_c+\omega_m)t\sin\omega_c t$$

$$= \frac{mV_c^2}{4}\left[\cos\omega_m t-\cos(2\omega_c+\omega_m)t\right]$$

- We get (ω_m) and $(2\omega_c+\omega_m)$ frequency components.
- The LPF rejects higher frequency component i.e. $(2\omega_c+\omega_m)$.
- The ω_m component is passed through LPF which is nothing but the audio signal.

4.5 Vestigial Sideband Transmission (VSB)

Q. *What is Vestigial Sideband Transmission ? Where is it used and why ?*

In VSB system, along with one full sideband and carrier we transmit some part of other sideband i.e. vestige of other sideband.

- The inputs to the Balanced modulators are audio signal and carrier frequency.
- The output of Balanced modulator is DSB-FC.
- LSB of output of Balanced modulator 1 is removed by USB pass filter and USB of output of Balanced modulator 2 is removed by LSB pass filter.
- Then both SSB signals are added together along with reduced carrier to get ISB signal.

4.7 Frequently Asked Questions

Q.1. Draw the block diagram of transmitter operating at 22.275 MHz without a pilot carrier and with two 3 kHz side bands.

Ans.

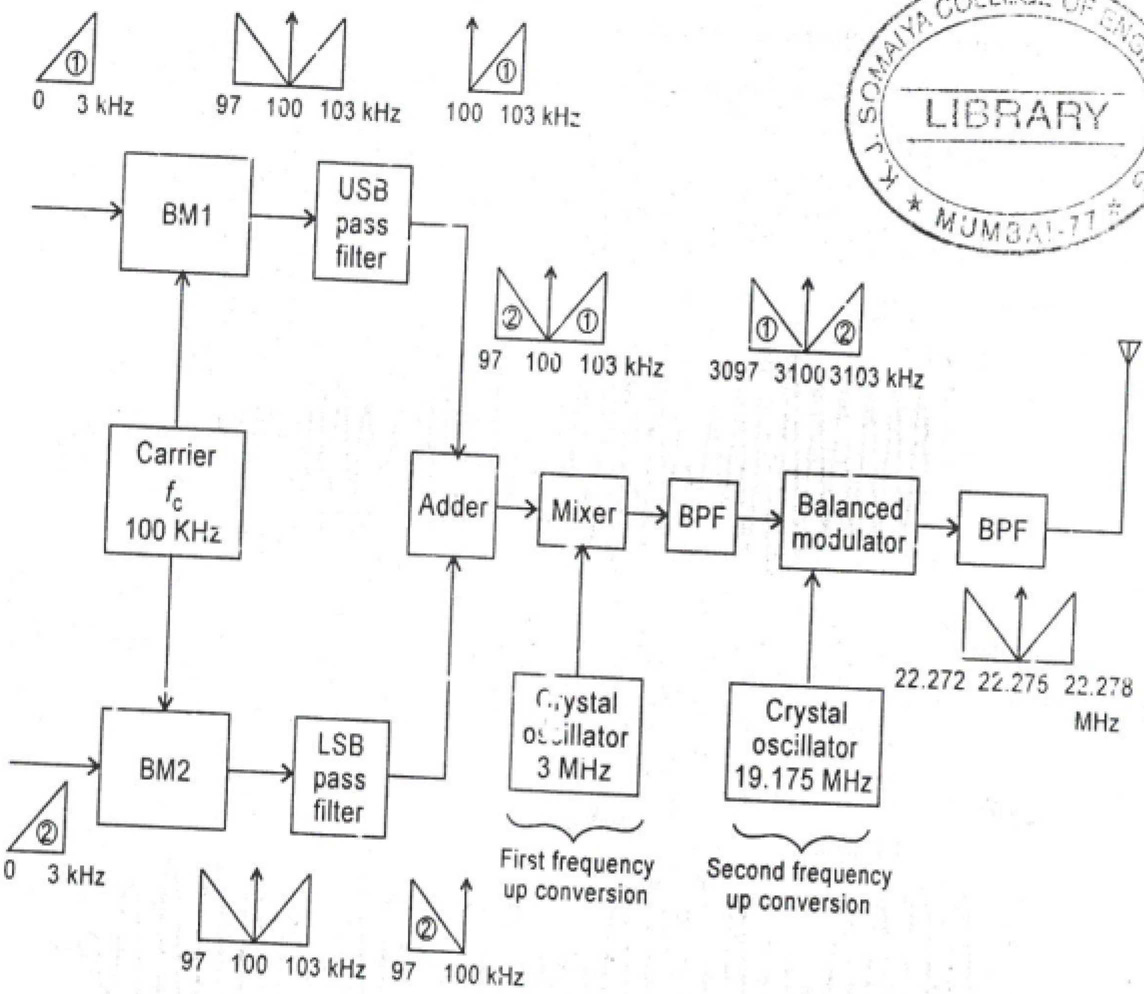


Fig. 4.18

Q.2. Compare the three main systems of SSB generation.

Ans.

Sr.	Parameter	Filter Method	Phase Shift Method	Third Method
(1)	Low modulating frequency	Cannot be used	Can be used	Can be used
(2)	SSB generation at high frequency	Not possible	Possible	Possible
(3)	Design	Using Filters	Using phase shifters by 90°	Using phase shifters by 90°
(4)	Complexity	Less	Average	Very complex
(5)	Frequency upconversion	Required	Not required	Not required

Table 4.1

Q.3. Give physical appearance of following :

(1) AM DSB-FC (2) AM DSB-SC (3) AM SSB-SC

Ans. (1) DSB-FC

Equation :

$$v_{DSB\text{-}FC} = V_c \sin \omega_c t + \frac{mV_c}{2} \cos (\omega_c - \omega_m)t - \frac{mV_c}{2} \cos (\omega_c + \omega_m)t$$

Waveform :

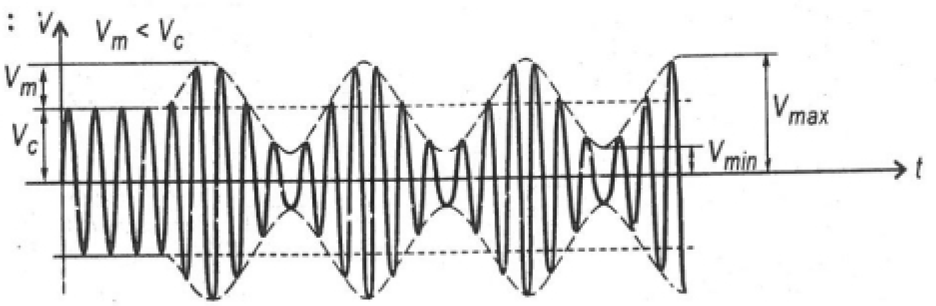


Fig. 4.19

(2) DSB-SC

Equation :

$$v_{DSB\text{-}SC} = \frac{mV_c}{2} \cos (\omega_c - \omega_m)t - \frac{mV_c}{2} \cos (\omega_c + \omega_m)t$$

Waveform :

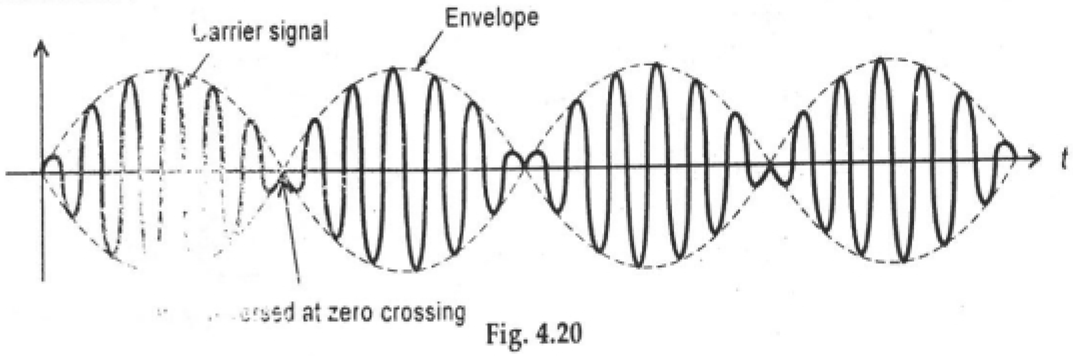


Fig. 4.20

(3) SSB-SC

Equation :

$$v_{SSB\text{-}L} = \frac{mV_c}{2} \cos(\omega_c - \omega_m)t \quad \text{.....(Lower sideband)}$$

$$v_{SSB\text{-}U} = \frac{mV_c}{2} \cos(\omega_c + \omega_m)t \quad \text{.....(Upper sideband)}$$

Waveform :

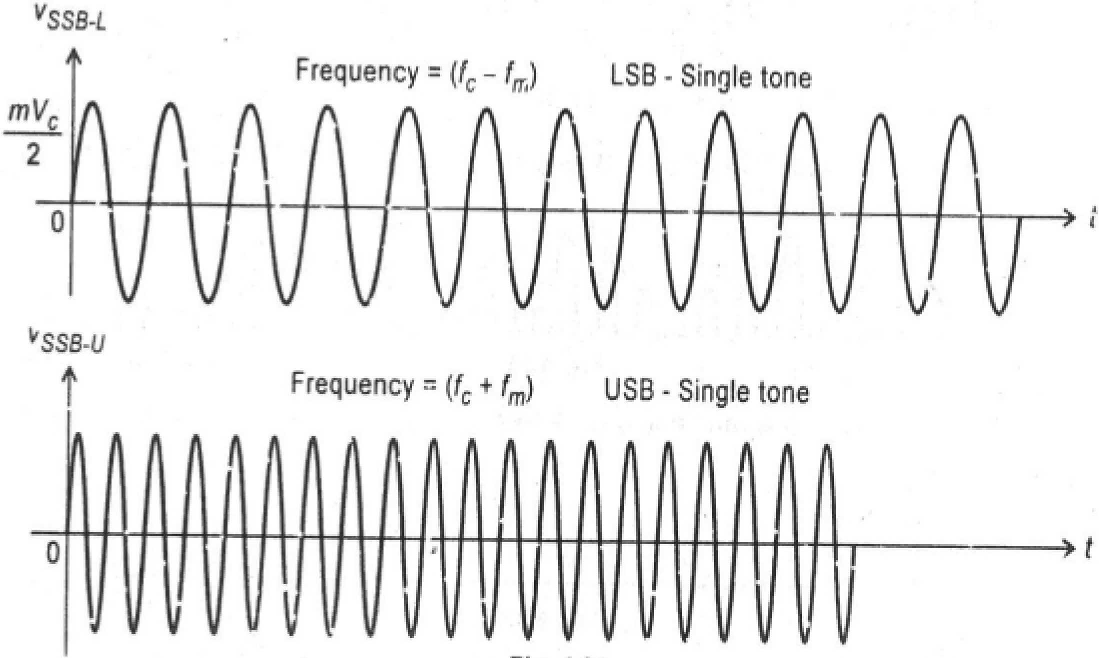


Fig. 4.21

Q.4. ***Explain why a linear envelope detector can not be used for demodulation of DSB-SC wave.***

Ans. Linear envelope detector is used for demodulation of DSB-FC signal. In this detector, the diode clips the negative half cycle and at the output we get original signal back.

Refer *figure 4.22.*

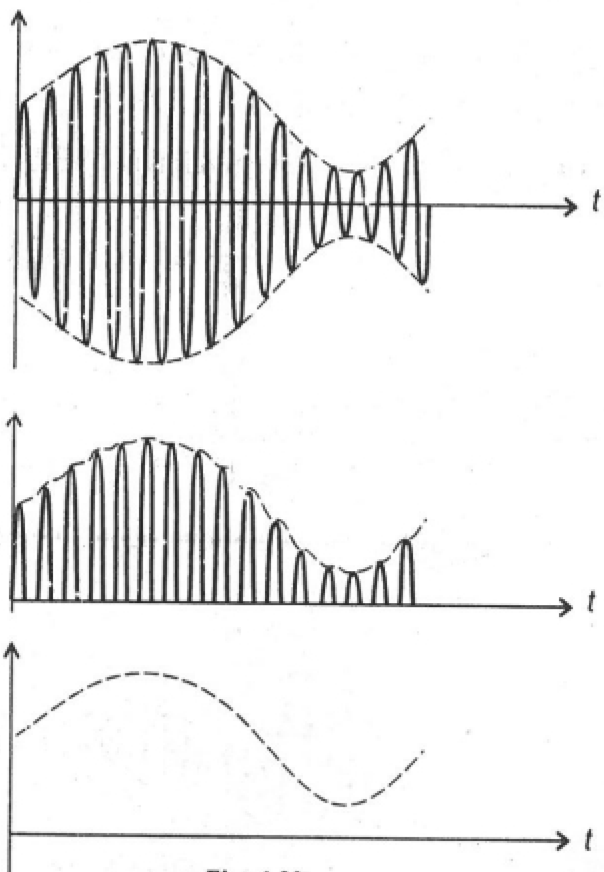


Fig. 4.22

Let us see the DSB-SC signal.

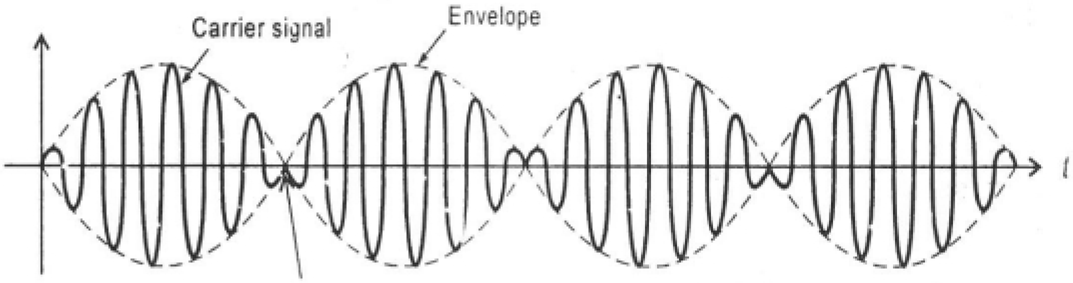


Fig. 4.23

If this signal is passed to linear envelope detector then we get the signal as

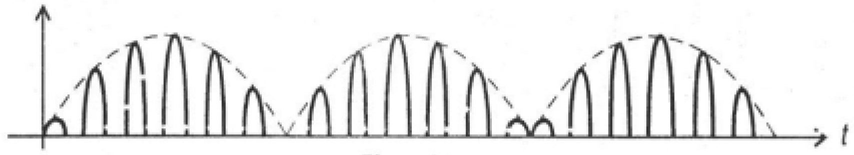


Fig. 4.24

Then the recovered signal would look like

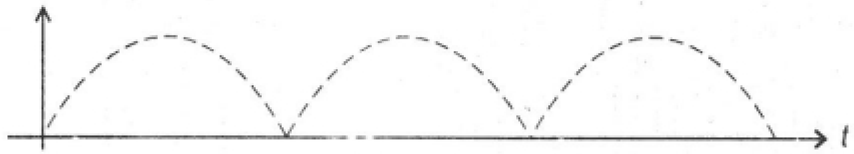


Fig. 4.25

The signal we get is not the original signal.

Q.5. ***Explain why SSB-SC wave can not be demodulated without presence of carrier wave.***

Ans. Let us consider the equation of SSB-SC wave

$$v_{USB} = \frac{mV_c}{2} \sin(\omega_c + \omega_m)t$$

$$v_{LSB} = \frac{mV_c}{2} \sin(\omega_c - \omega_m)t$$

Lets take $v_{USB} = \dfrac{mV_c}{2} \sin(\omega_c + \omega_m)t$

Now to demodulate above signal, we need to multiply it with carrier as shown :

$$V_c \sin \omega_c t \left[\frac{mV_c}{2} \sin(\omega_c + \omega_m)t\right]$$

$$= \frac{mV_c^2}{4}\left[2 \sin \omega_c t \sin(\omega_c + \omega_m)t\right]$$

$$= \frac{mV_c^2}{4}\left[\cos(\omega_c t - \omega_c t - \omega_m t) - \cos(\omega_c t + \omega_c t + \omega_m t)\right]$$

$$= \frac{mV_c^2}{4}\left[\cos \omega_m t - \cos (2\omega_c + \omega_m)t\right]$$

If we pass above generated signal to LPF then high frequency component will get rejected, i.e. at the output of LPF we will get sin $\omega_m t$ which is nothing but original signal. Hence, without carrier we can not demodulate SSB-SC.

Q.6. Compare following amplitude modulated systems : (1) DSB-FC (2) DSB-SC (3) SSB-SC.

Ans.

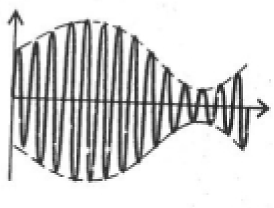
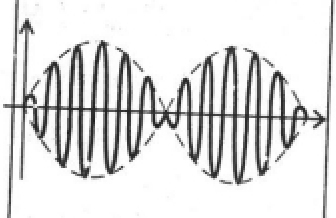
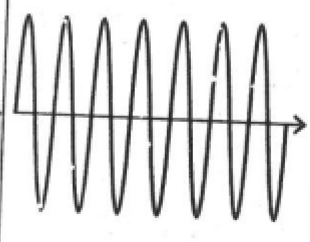
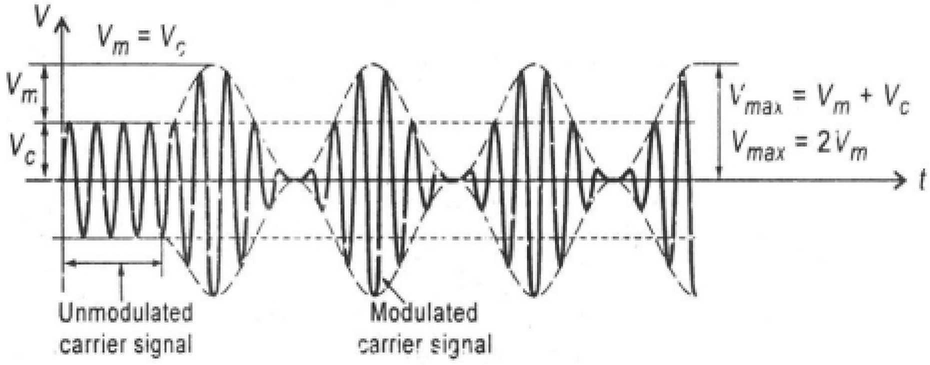
Parameter	**DSB-FC**	**DSB-SC**	**SSB-SC**
Carrier	Full carrier present	Suppressed carrier	Suppressed carrier
Number of sidebands	Two	Two	One
Equation	$v = V_c \sin \omega_c t$ $- \frac{mV_c}{2} \cos (\omega_c + \omega_m)t$ $+ \frac{mV_c}{2} \cos (\omega_c - \omega_m)t$	$v = \frac{mV_c}{2} \cos (\omega_c - \omega_m)t$ $- \frac{mV_c}{2} \cos (\omega_c + \omega_m)t$	$v = \frac{mV_c}{2} \cos (\omega_c + \omega_m)t$ OR $v = \frac{mV_c}{2} \cos (\omega_c - \omega_m)t$
Physical appearance			
Power saving	No	67 %	83 %
Non coherent reception	Possible	Not possible	Not possible
Bandwidth	$2f_m$	$2f_m$	f_m

Table 4.2

Q.7. Give difference between 100 % modulated DSB-FC and DSB-SC signal.

Ans. The *figure 4.26(a)* and *(b)* shows physical appearance of both

DSB-FC Signal



AM wave with modulation index = 1

Fig. 4.26(a)

DSB-SC Signal

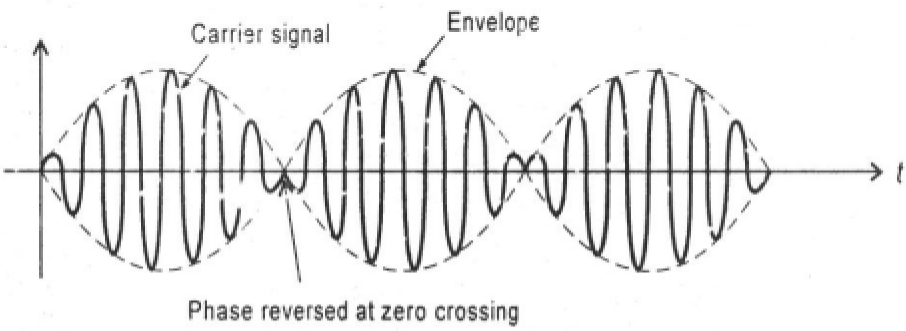


Fig. 4.26(b)

Q.8. Why the carrier wave cannot be recovered from SSB-SC wave by squaring method ?

Ans. The equation of SSB-SC wave is

$$v_{\text{SSB-L}} = \frac{m_a V_c}{2} \cos(\omega_c - \omega_m)t$$

$$v_{\text{SSB-U}} = \frac{m_a V_c}{2} \cos(\omega_c + \omega_m)t$$

We only transmit one of the two sidebands i.e. either USB or LSB.

In SSB-SC systems, carrier is suppressed. It is not transmitted along with one of the sideband.

At the receiver side, SSB-SC wave is received. Now if we square the received wave at the receiver end then output is,

$$\text{Output} = \frac{m^2V_c^2}{4}\cos^2(\omega_c - \omega_m)t \quad \ldots \{\text{Assuming LSB}\}$$

$$= \frac{m^2V_c^2}{4}\left(\frac{1+\cos 2(\omega_c - \omega_m)t}{2}\right)$$

$$\therefore \boxed{\text{Output} = \frac{m^2V_c^2}{8} + \frac{m^2V_c^2}{8}\cos 2(\omega_c - \omega_m)t}$$

Thus, we can see that output does not contain any term with frequency ω_c.

Thus, the carrier wave cannot be recovered from SSB-SC wave.

Q.9. ***Sketch the block diagram of Coherent SSB receiver and briefly explain the working.***

Ans. Write about pilot carrier receiver system from section 4.4.

Q.10. ***State advantages of SSB over DSB. Explain phase shift method to generate SSB AM.***

Ans. Refer section 4.2.2 for the phase shift method and refer Q.14 from this section.

Q.11. ***Compare the following amplitude modulation systems :***
(i) D.S.B.F.C. (ii) D.S.B.S.C. (iii) S.S.B. (iv) V.S.B.

Ans. Refer Q.6 from same section and write the following points for VSB.

Parameter	**VSB**
Carrier	Full carrier
Number of sidebands	One full sideband and Vestige of other sideband
Equation	$v = V_c \sin \omega_c t - \sum_{i=0}^{m} A \cos(\omega_c + \omega_i)t + \sum_{i=0}^{k} B \cos(\omega_c - \omega_m)t$ where, k = Vestige of LSB and $(k < m)$ A, B = Constants
Power saving	Less than SSB-SC but more than DSB-SC
Non coherent reception	Not possible
Bandwidth	$BW = (k + f_m)$

Table 4.3

Q.12. Draw the block diagram of an I.S.B. transmitter operating at 22.275 MHz with pilot carrier and with two 3 kHz side bands.

Ans. Refer question 1 from same section.

Q.13. Mention special requirement of SSB and ISB receiver.

Ans. Special requirement of SSB and ISB receiver are :

(1) Carrier Generator

In SSB and ISB, both the systems require carrier generator at the receiver as suppressed carrier is transmitted.

(2) Product Demodulator

Write about Product Detector from section 4.3.1.

Q.14. Mention advantages and disadvantages of SSB over DSB and AM.

Ans. **Advantages of SSB over DSB and AM :**

(1) In SSB only single sideband is transmitted, hence the bandwidth required is f_m whereas for DSB-FC and DSB-SC it is $2f_m$.

(2) In SSB only one sideband is transmitted and no carrier, hence power saved is more approximately 83 % whereas in DSB-SC the carrier is suppressed, hence power saved is less than SSB approximately 67 %. In DSB-FC there is no power saving.

Disadvantages of SSB over DSB and AM :

(1) Removal of one sideband and carrier can sometimes attenuate the information content in other sideband near to the carrier i.e. loss of information is possible.

(2) SSB receivers need to generate the carrier for demodulation.

❋❋❋

5 Angle Modulation and Generation

Topic	Theory imp	Oral imp
Introduction		
Equation of FM	☆☆	
Phase Modulation	☆	
Spectrum of FM	☆	☆
Noise Triangle	☆☆☆☆	☆☆☆
Pre-emphasis, De-emphasis	☆☆☆☆	☆☆
Comparisons	☆☆☆☆☆	☆☆☆
Reactance Modulator	☆☆	☆
Varactor Diode Modulator	☆☆	☆
Armstrong Method	☆☆☆☆	☆☆☆
FAQ's	☆	☆
Problems	☆☆☆	

5.0 Introduction

In the process of modulation, the information is carried by the signal in the form variations. The variations can be in any one of its parameters like amplitude, frequency or phase.

We have studied Amplitude Modulation earlier. In this chapter we will study the modulation of carrier signal either by changing frequency or phase of the carrier signal, together we call it angle modulation.

5.1 Angle Modulation

Q. *Define FM and derive equation for FM wave.*

Let us consider an unmodulated carrier wave given by

$$X = A \sin(\omega_c t + \phi) = A \sin \psi$$

where ϕ = Phase angle

$$\psi = \omega_c t + \phi$$

$$\omega_c = 2\pi f_c \text{ and } f_c = \text{Frequency}$$

If the parameter A of carrier is varied according to the amplitude of message signal then the result is amplitude modulation.

If the parameter ψ is varied according to the modulating signal, the result is angle modulation.

We can achieve angle modulation either by varying frequency or phase of carrier wave.

Accordingly we have two types of angle modulation :

(i) Frequency Modulation

(ii) Phase Modulation.

5.1.1 Frequency Modulation

Q. Derive an expression for mathematical representation of FM.

If the frequency of the carrier is modulated according to the instantaneous value of modulating signal, then it is called *frequency modulation*.

Equation of FM

$$\text{Let } f_c = \text{Carrier frequency}$$

$$v_m = V_m \cos \omega_m t = \text{Modulating signal}$$

Refer *figure 5.1*.

In FM, the carrier frequency is modulated. The new f is given as :

$$f = \text{New frequency} = f_c + \text{Deviation}$$

$$\text{where deviation} = k\, f_c\, v_m \quad \begin{pmatrix} \because \text{ frequency depends on} \\ \text{instantaneous value of } v_m \end{pmatrix}$$

$$\text{i.e.} \quad = f_c + k\, f_c\, V_m \cos \omega_m t \qquad \text{..... (1)}$$

$$\text{i.e. } 2\pi f = 2\pi f_c + 2\pi f_c\, k\, V_m \cos \omega_m t$$

$$\therefore\ \omega = \omega_c + \omega_c\, k\, V_m \cos \omega_m t$$

$$\text{where } \omega = 2\pi f$$

$$\therefore\ \omega = \omega_c [1 + k\, V_m \cos \omega_m t] \qquad \text{..... (2)}$$

In FM, the amplitude of the signal remains constant but the frequency varies according to modulating signal.

$\therefore$ we can write equation of FM as

$$v_{FM} = A \sin \theta$$

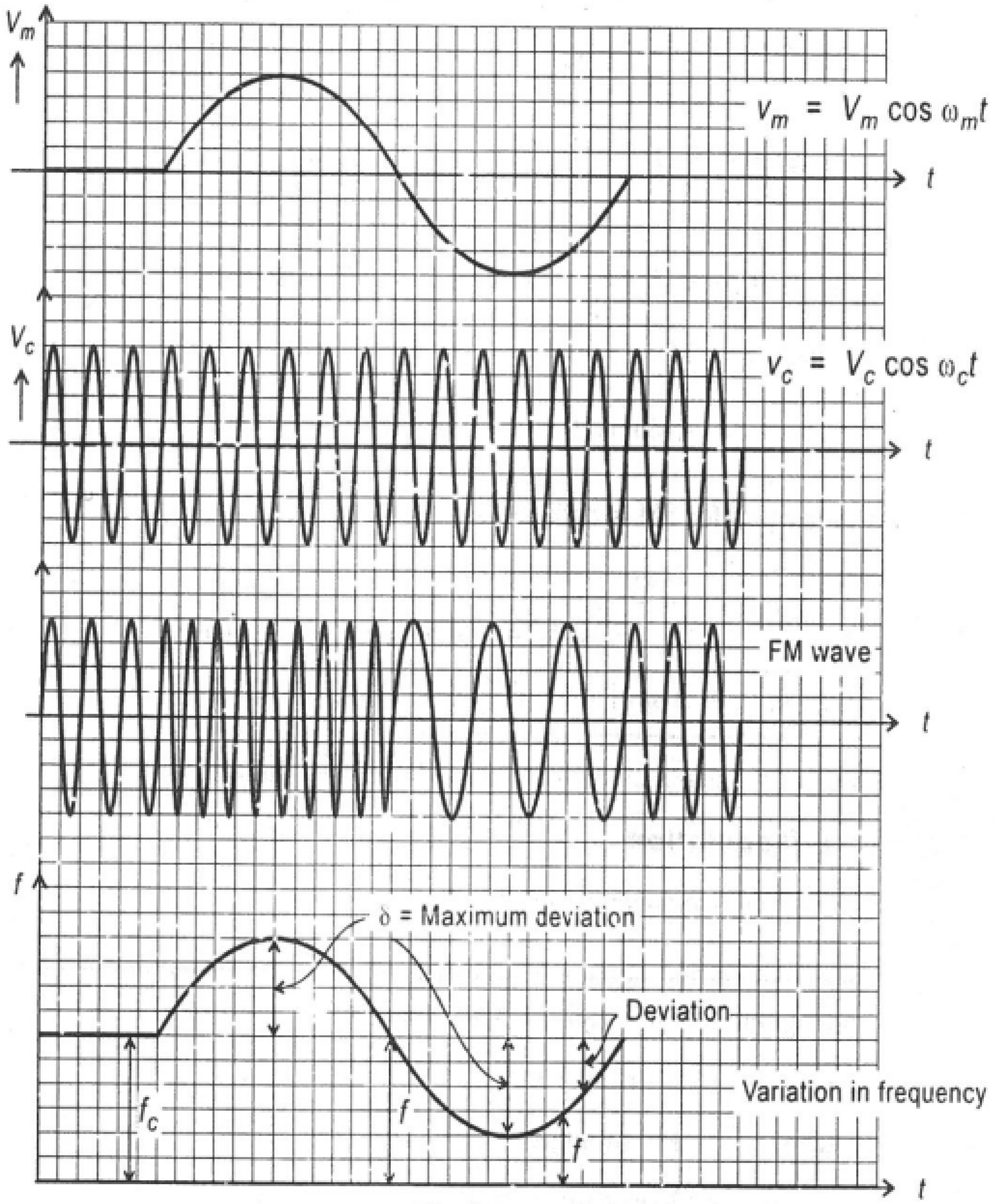


Fig. 5.1

We can find value of θ as follows :

$$\omega = \frac{d\theta}{dt}$$

$$\therefore \ \theta = \int \omega \, dt$$

Substitute equation (2),

$$\therefore \ \theta = \int \omega_c [1 + k\, V_m \cos \omega_m t] \, dt$$

$$= \omega_c t + \omega_c k\, V_m \left[\frac{\sin \omega_m t}{\omega_m}\right] + \phi \qquad \text{where, } \phi = \text{Phase}$$

$$\therefore \ \theta = \omega_c t + \omega_c k\, V_m \left[\frac{\sin \omega_m t}{\omega_m}\right] \qquad \text{..... } \{\phi = 0 \text{ at initial position}\}$$

$$\theta = \omega_c t + k\frac{2\pi f_c}{2\pi f_m} V_m \sin \omega_m t$$

$$\therefore \theta = \omega_c t + \frac{k V_m f_c}{f_m} \sin \omega_m t$$

$$\therefore \theta = \omega_c t + \frac{\delta}{f_m} \sin \omega_m t \qquad ..... (3)$$

where δ = Maximum frequency deviation

$$\therefore v_{FM} = A \sin\left[\omega_c t + \frac{\delta}{f_m} \sin \omega_m t\right] \qquad ..... (4)$$

Modulation index (m_f) for FM is defined as :

$$m_f = \frac{\text{Maximum frequency deviation}}{\text{Modulating frequency}} = \frac{\delta}{f_m}$$

$$\therefore \boxed{v_{FM} = A \sin [\omega_c t + m_f \sin \omega_m t]} \qquad ..... (5)$$

The above expression gives frequency modulated carrier signal.

5.1.2 Phase Modulation

Q. Explain the relationship between FM and PM.

If the phase of carrier signal is varied according to the instantaneous amplitude of modulating signal, then it is called *phase modulation.*

Let $v_{PM} = A \sin [\omega_c t + \phi]$

where ϕ = Phase angle

and $\phi \propto$ Modulating signal

$$\therefore \phi = k\, v_m = k V_m \sin \omega_m t$$

$$\therefore v_{PM} = A \sin [\omega_c t + k V_m \sin \omega_m t]$$

$$\therefore v_{PM} = A \sin [\omega_c t + m_p \sin \omega_m t] \qquad ..... (6)$$

where $m_p = k V_m$ = Phase modulation index

and it is proportional to amplitude of modulating signal and independent of frequency.

5.2 Frequency Spectrum of FM

FM signal is given as

$$v_{FM} = A \sin [\omega_c t + m_f \sin \omega_m t]$$

Above signal can be expanded by using Bessel's function of the first kind $J_n(m_f)$ to get different component in FM.

$$v_{FM} = A\Big\{J_0(m_f)\sin\omega_c t + J_1(m_f)\,[\sin(\omega_c+\omega_m)\,t - \sin(\omega_c-\omega_m)\,t] + J_2(m_f)\,[\sin(\omega_c+2\omega_m)\,t + \sin(\omega_c-2\omega_m)\,t] + \ldots\ldots\Big\}$$

From the above equation, it is seen that output consists of carrier frequency along with infinite number of sidebands.

If we substitute different values of m_f and n then the amplitude of each sideband can be calculated as :

$$J_n(m_f) = \sum_{r=0}^{\infty}\frac{(-1)^r\left(\frac{m_f}{2}\right)^{2r+n}}{r!\,\Gamma(n+r+1)}$$

$$= \left(\frac{m_f}{2}\right)^n\left\{\frac{1}{n!} - \frac{\left(\frac{m_f}{2}\right)^2}{1!\,(n+1)!} + \frac{\left(\frac{m_f}{2}\right)^4}{2!\,(n+2)!} - \ldots\ldots\ldots\right\}$$

> **Note :** *To expand a sine of sine function i.e. a function like $f(x) = \sin(a + b\sin x)$ we need Bessel's function.*

Table 5.1 gives amplitude of number of sidebands for different values of m_f .

m_f	J_0	J_1	J_2	J_3	J_4	J_5	.....
0.0	1.0						
0.25	0.98	0.12	–	–	–		
0.5	0.94	0.03	–	–	–		
1.0	0.77	0.44	0.11	0.02	–		

Table 5.1

If $m_f = 1$ and $A = 1$ V, then

Amplitude of carrier = 0.77

Amplitude of 1st sideband $(\omega_c \pm \omega_m)$ = 0.44 V

Amplitude of 2nd sideband $(\omega_c \pm 2\omega_m)$ = 0.11 V

Amplitude of 3rd sideband $(\omega_c \pm 3\omega_m)$ = 0.02 V

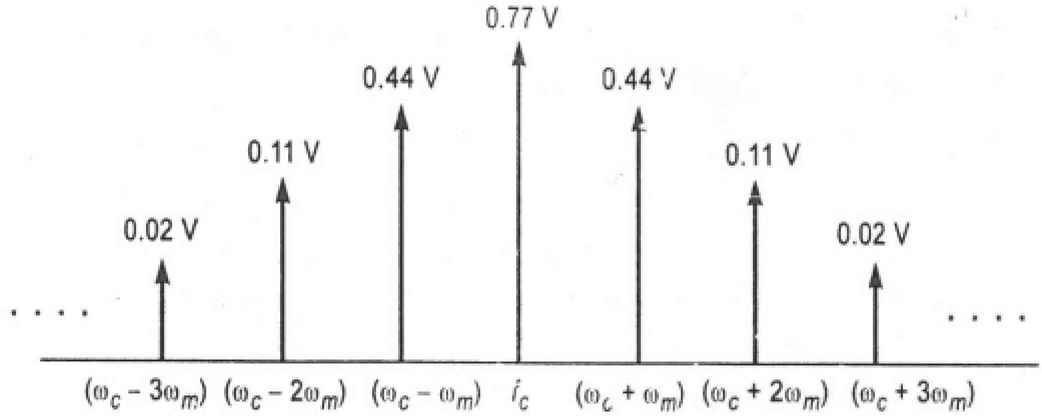


Fig. 5.2

5.3 Carson's Rule

This rule is used to estimate BW (bandwidth) of all angle modulated systems regardless of modulation index.

Carson's rule states that we can neglect any frequency component (i.e. sideband) beyond n such that $n = m_f + 1$ and m_f is modulation index.

$$\therefore \text{ BW of FM } = 2n f_m = 2(m_f + 1) f_m \quad \text{where } m_f = \frac{\delta}{f_m}$$

$$= 2\left(\frac{\delta}{f_m} + 1\right) f_m$$

$$\text{BW of FM } = 2(\delta + f_m)$$

5.4 Noise Triangle

Q. Explain FM noise triangle.

If the signal noise frequency falls within the passband of the receiver then it affects the output of the receiver.

The noise frequency gets added to the FM signal and produces unwanted changes as shown in *figure 5.3*.

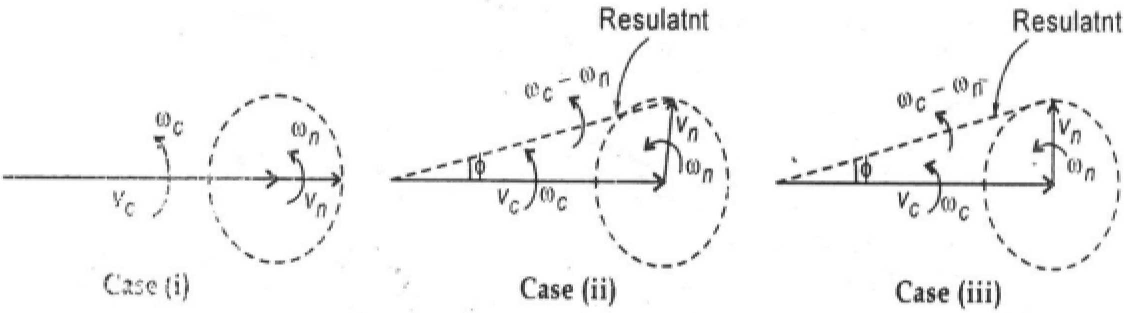


Fig. 5.3

As we can see from *figure 5.3*,

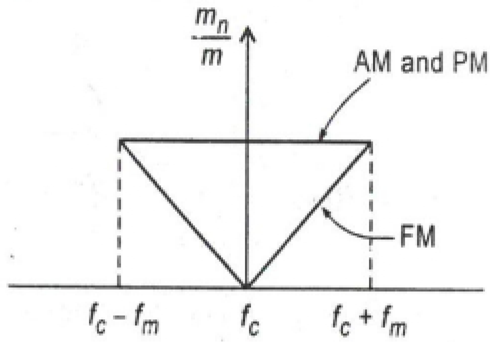
In case (i) : Amplitude changes but phase change is zero.

In case (ii) : Amplitude as well as phase changes.

In case (iii) : Amplitude changes and phase changes to its maximum.

The amplitude change can be eliminated by Amplitude limiter. The phase change can not be eliminated. But it can be minimized using pre-emphasis and de-emphasis (explained later). Noise triangle thus shows how noise is related to frequency of the incoming signal.

Noise triangle is shown in *figure 5.4*.



Here,

$m = m_a$ for AM

$m = m_p$ for PM

$m = m_f$ for FM

Fig. 5.4

- Noise triangle is a graph of frequency versus $\frac{\text{Modulation index of noise}}{\text{Modulation index of signal}}$.
- The noise modulation index remains almost constant for all frequencies.
- In AM, modulation index is $\frac{V_m}{V_c}$ i.e. it is independent of frequency. Hence noise in AM is almost constant.
- Similarly we can say that, noise in PM is almost constant. Since, $m_p = kV_m$.
- For FM, modulation index m_f is $\frac{\delta}{f_m}$.

  Hence as frequency f_m increases, the modulation index m_f decreases.
- Hence $\frac{m_n}{m_f}$ increases. This gives shape of Triangle. Hence known as *Noise Triangle.*

5.5 Pre-emphasis and De-emphasis

Q.1. Why is pre-emphasis required in FM generation. Explain the working of pre-emphasis and de-emphasis.

Q.2. Explain pre-emphasis and de-emphasis in F.M.

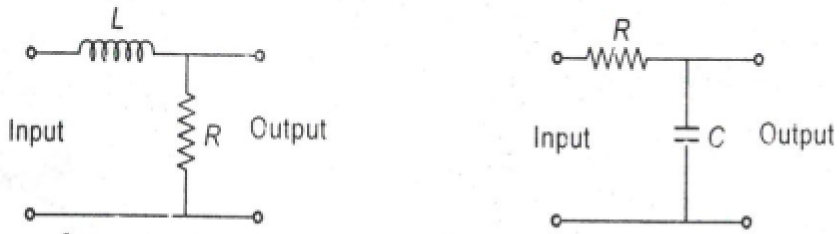
- From noise triangle of FM, we can see that the effect of noise increases if higher audio frequencies are used for frequency modulation.

- To overcome above problem, the higher audio frequencies are boosted at the transmitter and accordingly attenuated at the receiver.
- The artificial boosting of higher order frequencies at the transmitter is done with the help of pre-emphasis circuit.
- Similarly attenuation of the boosted frequencies at receiver is done with the help of de-emphasis circuit.
- **Pre-emphasis Circuit** : This circuit consists of high pass circuit with time constant of 60 μs.

$\frac{L}{R}$ = Time constant = 60 μs $\qquad$ RC = Time constant = 60 μs

Fig. 5.5

- **De-emphasis Circuit** : This circuit is nothing but a low pass circuit with a time constant of 60 μs.



$\frac{L}{R}$ = 60 μs $\qquad$ RC = 60 μs

Fig. 5.6

- The pre-emphasis and de-emphasis in done according to the pre-defined curve shown in *figure 5.7*.

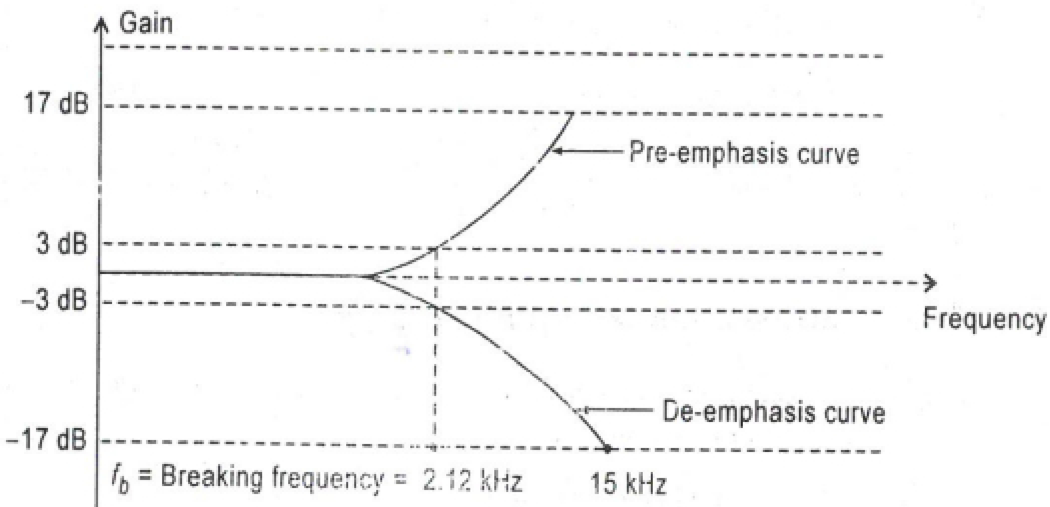


Fig. 5.7

- When the signal at the transmitter is boosted, care should be taken not to over modulate the higher order frequencies. Otherwise distortion will take place.

5.6 Comparisons

(1) Comparison Between Wideband FM and Narrowband FM

Q. *Compare Wideband and Narrow band FM.*

Sr.	Wideband FM	Narrowband FM
(1)	It is type of FM where m_f normally exceeds unity $m_f > 1$.	it is type of FM where m_f slightly greater than 0 but less than 1.
(2)	Maximum deviation $\delta = 75$ kHz.	Maximum deviation $\delta = 5$ kHz
(3)	Range of modulating frequency f_m is 30 Hz to 15 kHz.	Range of modulating frequency f_m is upto 3 kHz.
(4)	Bandwidth is almost 15 times greater than that of narrowband FM.	Bandwidth is comparatively less.
(5)	Due to large bandwidth effect of noise is more.	Effect of noise less.
(6)	There are more than two sidebands having significant amplitude.	There are only two sidebands.

Table 5.2

(2) Comparison Between AM and FM

Q. *Compare AM with FM.*

Sr.	AM	FM
(1)	In amplitude modulation, the amplitude of carrier is varied w.r.t instantaneous amplitude of modulating signal.	In frequency modulation frequency of carrier is varied w.r.t instantaneous amplitude of modulating signal.
(2)	Equation : $v_{AM} = [V_c + mV_c \sin \omega_m t] \sin \omega_c t$	Equation : $v_{FM} = V_c \sin [\omega_c t + m_f \sin \omega_m t]$
(3)	Modulation index $m = \dfrac{V_m}{V_c}$	Modulation index $m_f = \dfrac{\delta}{f_m}$
(4)	Bandwidth $2 f_m$.	Bandwidth $= 2[\delta + f_m]$.
(5)	Bandwidth is much less than FM.	Bandwidth is large.

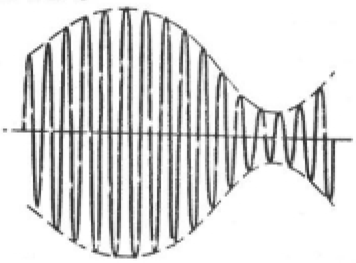
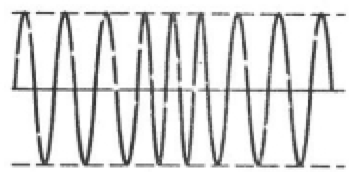
(6)	Frequency range of modulated signal f_s = 540 kHz to 1650 kHz.	Frequency range of modulated signal f_s = 88 MHz to 108 MHz.
(7)	Less complex.	More complex.
(8)	Information is contained in amplitude variation of carrier.	Information is contained in frequency variation of carrier.
(9)	AM wave 	FM wave 
(10)	**Applications** : Radio and TV transmission.	**Applications** : Radio and TV transmission, point to point communication.

Table 5.3

(3) Comparison Between FM and PM

Q. Differentiate FM and PM.

Sr.	FM	PM
(1)	The frequency of the carrier is varied according to instantaneous value of modulating signal.	The phase of carrier is varied according to instantaneous value of modulating signal.
(2)	Equation : $v_{FM} = V_c \sin [\omega_c t + m_f \sin \omega_m t]$	Equation : $v_{PM} = V_c \sin [\omega_c t + m_p \sin \omega_m t]$
(3)	Modulation index depends upon the modulating frequency. $m_f = \frac{\delta}{f_m}$	Modulation index does not depend upon modulating frequency. $m_p = kV_m$
(4)	Noise immunity is better than both AM and PM	Noise immunity is better than AM but worse than FM.
(5)	**Applications** : Radio, TV transmission, wireless devices, point to point communication.	**Applications** : Used in Mobile systems.

Table 5.4

5.7 FM Generation

Q. Explain any one method for the generation of FM signal with neat sketches.

Mainly, there are two ways in which an FM wave can be generated.

FM Generation
- Direct Method
  - Reactance Modulator
  - Varactor Diode Modulator
- Indirect Method
  - Armstrong Method

Tree 5.1

[1] Direct Method

Principle : We know that the output frequency of any tank circuit depends on the value of L and C. Thus, if the value of L and C is varied then the output frequency will also vary and an FM wave will be generated.

There are two methods :

- Reactance modulator (Basic FET reactance modulator).
- Varactor diode modulator.

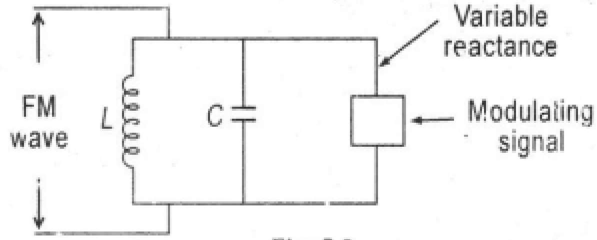
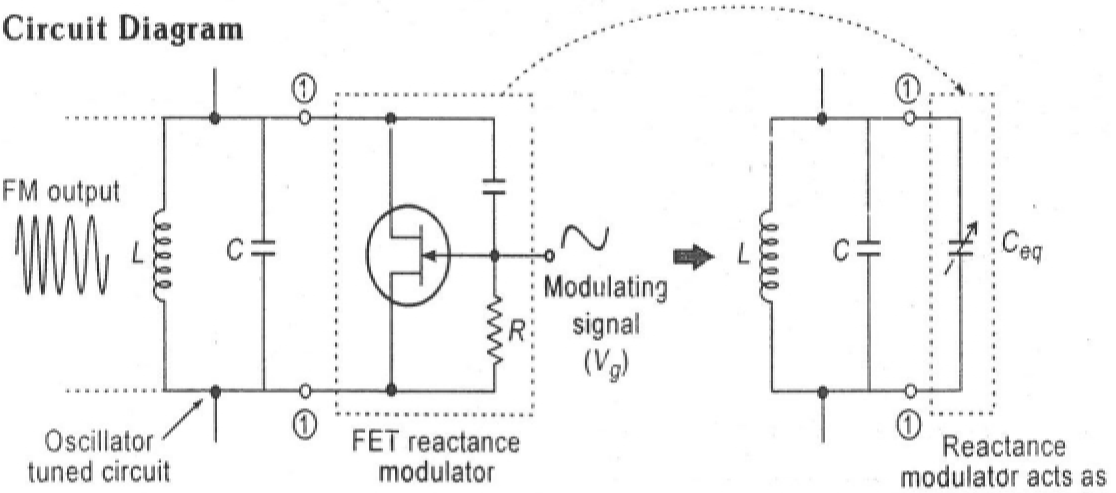


Fig. 5.8

> **Note :** *Each of the above circuits is explained in two steps.*
> *Step (i) : How the reactance of circuit varies with the modulating signal.*
> *Step (ii) : Working*

5.7.1 Reactance Modulator

Circuit Diagram



(a) Frequency modulation using FET reactance modulator.

(b) Equivalent circuit.

Fig. 5.8

Step (i) : How the reactance of circuit varies with the modulating signal.

Consider the part of the circuit which acts as variable reactance modulator.

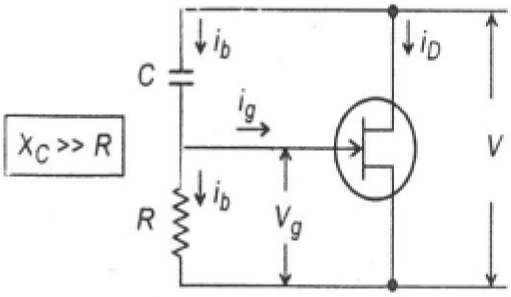


Fig. 5.10

Assumptions

(i) $i_b << i_D$

(ii) $X_C >> R$

Mathematical Explanation

- From the circuit :

$$V_g = i_b R \quad \{\because \text{ Current to the gate of FET is very low } (i_g \approx 0)\}$$

$$V_g = \frac{VR}{(R - jX_C)} \quad \{\text{Using voltage divider law}\}$$

- Also, the drain circuit is given by

$$i_D = g_m \times V_g \qquad \text{where } g_m \text{ is the transconductance}$$

$$i_D = \frac{g_m \times RV}{(R - jX_C)}$$

- Now the overall impedance of circuit is

$$Z = \frac{V}{i_D} = \frac{V}{\frac{g_m RV}{(R - jX_C)}}$$

$$Z = \frac{R - jX_C}{g_m R}$$

$$Z = \frac{1}{g_m}\left(1 - \frac{jX_C}{R}\right)$$

as $X_C >> R$ then $1 - \frac{jX_C}{R} \approx \frac{-jX_C}{R}$

$$\therefore \boxed{Z = \frac{-jX_C}{g_m R}}$$

Now from above equation, Z depends on g_m and also g_m depends on the gate voltage applied

$$\left(\because\ g_m = \frac{i_D}{V_g}\right)$$

Thus the reactance of the circuit can be changed by varying the gate voltage. Hence, modulating signal is applied at the gate.

> **Note :** *In the above circuit the overall impedance of the circuit behaves almost like a capacitive reactance where equivalent capacitive reactance is*
>
> $$X_{eq} = \frac{X_C}{g_m R}$$

Step (ii) : Working

Case (i) : When V_g Increases

When V_g increases

$\downarrow$

g_m decreases $\left(\because\ g_m = \frac{\overset{i_D}{i_n}}{V_g}\right)$

$\downarrow$

X_{eq} increases $\left(X_{eq} = \frac{X_C}{g_m R}\right)$

$\downarrow$

Frequency increases

Case (ii) : When V_g Decreases

As V_g decreases

$\downarrow$

g_m increases

$\downarrow$

X_{eq} decreases

$\downarrow$

Frequency decreases

5.7.2 Varactor Diode Modulator

Q. With a neat circuit diagram, explain varactor diode FM modulator.

Circuit Diagram

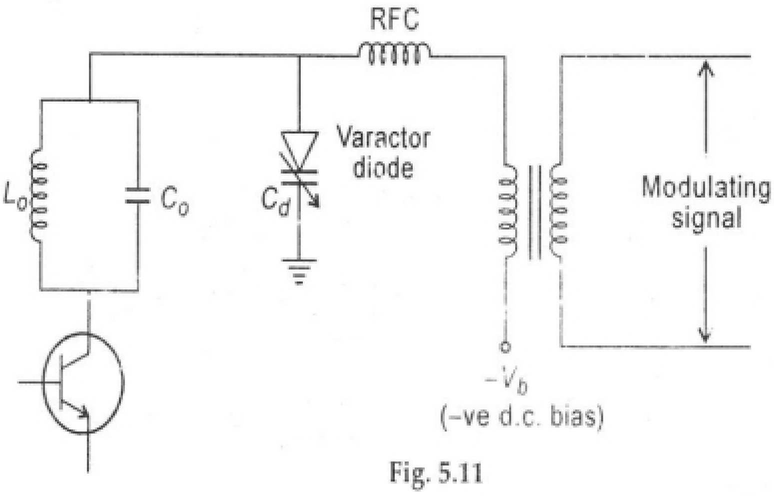


Fig. 5.11

Step (i) : How does the reactance of circuit change w.r.t. modulating signal.

> **Note :** *Please first read what is a varactor diode and how does it work in section 5.9.*

- We know that varactor diode can act as a capacitor if operated in reversed bias mode.
- Now, since the overall capacitance is the parallel combination of C_o and C_d, then $C_{eq} = C_o + C_d$.
- Also, C_d depends on the negative voltage applied and this voltage applied is proportional to the modulating voltage.

$$\therefore \boxed{C_d \propto V_m}$$

- Thus, the reactance of circuit can be changed by varying the voltage applied to the varactor diode.

Step (ii) : Working :

Case (i) : When Modulating Signal Increases

As V_m increases

$\downarrow$

C_d decreases

$\downarrow$

C_{eq} decreases

$\downarrow$

Output frequency increases $\left(\because f \propto \dfrac{1}{C_{eq}}\right)$

> **Note :** *RFC in the figure 5.11 is a radia frequency coil. It is used for noise reduction.*

Case (ii) : When Modulating Signal Decreases

As V_m decreases

$\downarrow$

C_d increases

$\downarrow$

C_{eq} increases

$\downarrow$

Output frequency decreases

Advantages of Direct Method

- Simple to design.

Disadvantages of Direct Method

- Since frequency of LC oscillator keeps on drifting, some additional circuits like AFC are required to maintain the consistency.

[2] Indirect Method

Q *With a neat block diagram. explain the principle and generation of indirect method of FM generation.*

Principle : The indirect method of FM generation generates FM through PM.

We know that

$$\text{if} \quad v_c = V_c \sin \omega_c t$$

$$\text{and} \quad v_m = V_m \cos \omega_m t$$

> ***Note :*** *$v_m = V_m \sin w_m t$ and $v_m = V_m \cos w_m t$ are the same signal. just have a 90^0 phase difference. But in real world that doesn't make any difference.*

Now, equation of FM wave with $v_c = V_c \sin \omega_c t$ and $v_m = V_m \sin \omega_m t$ is

$$v_{FM} = V_c \sin (\omega_c t + m_f \sin \omega_m t)$$

$$m_f = \frac{k\, V_m f_c}{\omega_m} \qquad \text{Let } k_f = k\, f_c$$

$$\boxed{\therefore \; v_{FM} = V_c \sin\left[\omega_c t + \frac{k_f V_m}{\omega_m} \sin \omega_m t\right]} \qquad \text{..... (1)}$$

Now, equation of a PM wave with $v_c = V_c \sin \omega_c t$ as carrier and $v_m = V_m \cos \omega_m t$ as modulating signal is

$$v_{PM} = V_c \sin (\omega_c t + m_p \cos \omega_m t)$$

$$m_p = kV_m$$

$$\therefore \boxed{v_{PM} = V_c \sin\left[\omega_c t + kV_m \cos \omega_m t\right]} \quad \text{..... (2)}$$

Comparing the two equations we can conclude that : *if the modulating signal is integrated and then applied to a phase modulator then we can generate FM from PM.* (Assuming $k = k_f$). The block diagram is as shown in *figure 5.12*.

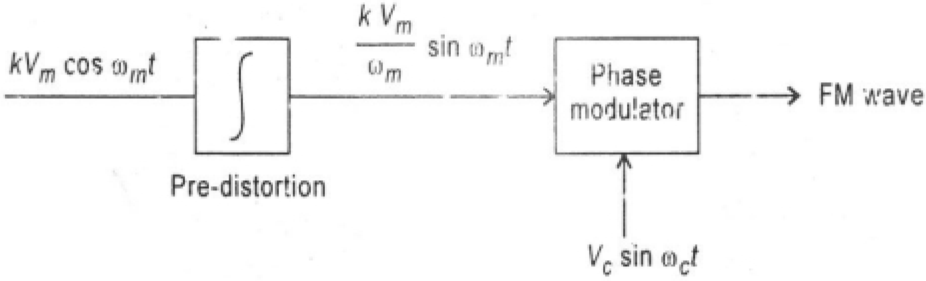


Fig. 5.12

5.8 Armstrong Method of FM Generation

Q. Draw block diagram of Armstrong frequency modulation and explain the functions of mixer and multiplier.

- This method normally develops narrow band FM.
- To get wide band FM, some multipliers are used.

Prerequisites (for Understanding the Circuit)

We know that equation of a Narrowband FM signal with carrier suppressed is

$$v_{NBFM} = \frac{1}{2}\sin(\omega_c + \omega_m)t - \frac{1}{2}\sin(\omega_c - \omega_m)t \quad \text{..... (1)}$$

Also equation of a DSB-SC wave is

$$v_{DSB\text{-}SC} = \frac{-m_a V_c}{2}\cos(\omega_c + \omega_m)t + \frac{m_a V_c}{2}\cos(\omega_c - \omega_m)t$$

$$= \frac{m_a V_c}{2}\sin(\omega_c + \omega_m + 90^o)t - \frac{m_a V_c}{2}\sin(\omega_c - \omega_m + 90^o)t \quad \text{... (2)}$$

Thus by comparing equations (1) and (2) we can conclude that the narrowband and DSB-SC wave have a phase difference of 90^o. Also in both the cases *modulating signal should be a sine wave.* (Assuming $m_a = 1$ and $V_c = 1$ V)

Circuit Diagram

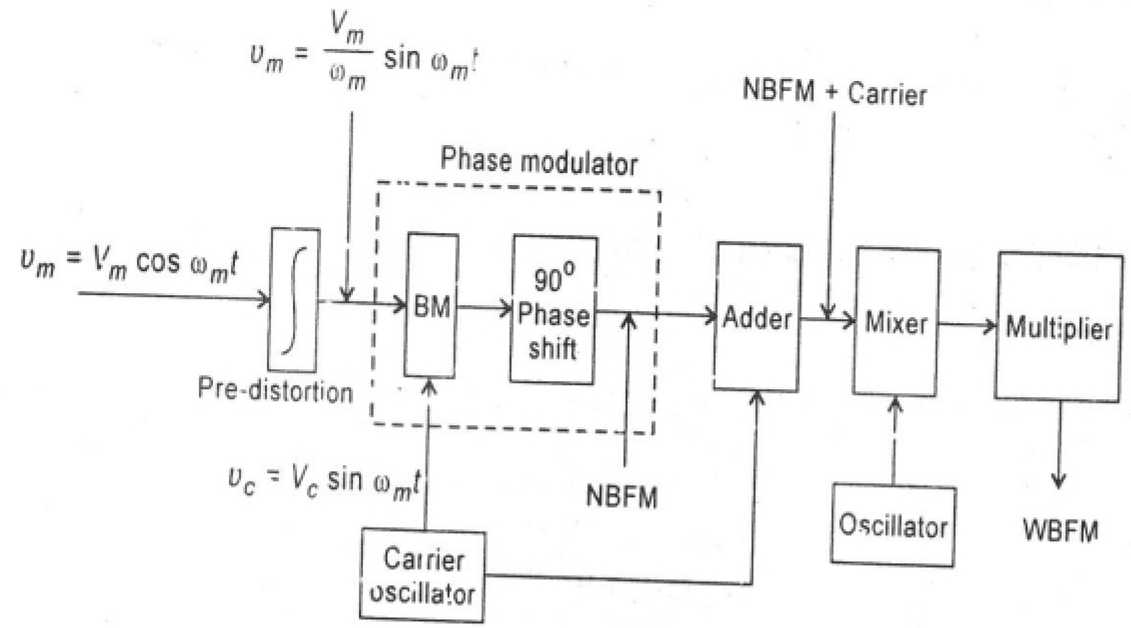


Fig. 5.13 : Armstrong Method of FM Generation

Working

- The modulating signal is first integrated so that $\cos \omega_m t$ is replaced by $\frac{\sin \omega_m t}{\omega_m}$.
- This is then applied to a balanced modulator with a carrier to get the two side bands i.e. DSB-SC signal.
- This signal is then shifted in phase by 90° to get a narrowband FM signal.
- The balanced modulator and phase shifter together act as phase modulator.
- The Adder adds the carrier to get a Narrowband FM with carrier.
- Mixer is just used for frequency up conversion.
- Frequency Multiplier converts Narrowband FM to WBFM.

5.9 Formulae

(1) FM wave $= V_c \sin(\omega_c t + m_f \sin \omega_m t)$

(2) Spectrum of FM

$$v_{FM} = V_c\Big\{J_0(m_f)\sin\omega_c t + J_1(m_f)\,[\sin(\omega_c+\omega_m)t - \sin(\omega_c-\omega_m)t] + J_2(m_f)\,[\sin(\omega_c+2\omega_m)t + \sin(\omega_c-2\omega_m)t] + \ldots\Big\}$$

(3) Bandwidth of FM $= 2(\delta + f_m)$

(4) Deviation of FM wave $\delta = k f_c\, v_m$

(5) Modulation index of FM $m_f = \frac{\delta}{f_m}$

(6) PM wave $= V_c \sin [\omega_c t + m_p \sin \omega_m t]$

(7) Modulation index of PM $m_p = kV_m$

5.10 Frequently Asked Questions

Q.1. How varactor diode acts as variable capacitance ?

Ans. A diode physically looks like

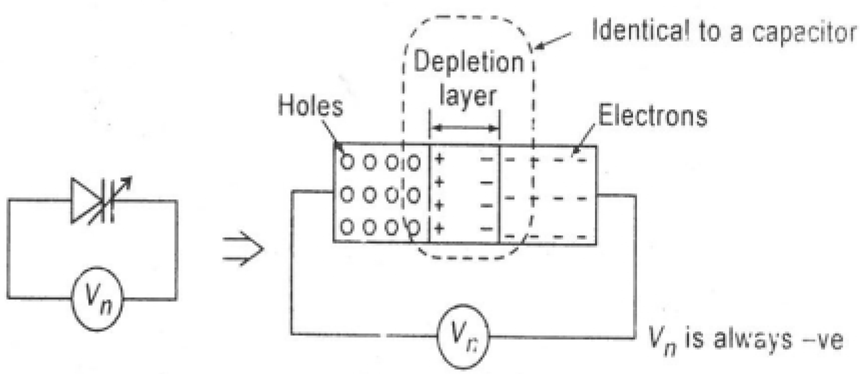


Fig. 5.14

Thus it is almost identical to two different charges (+,–) separated by any material. Thus it is almost similar to a capacitor.

Now, it can act as a variable capacitance by varying V_n.

Q.2. What is δ and m_f ?

Ans.

(1) **Maximum Frequency Deviation (δ) :** The maximum frequency deviation is directly proportional to the amplitude of audio input.

This parameter decides by how much the frequency f changes above or below frequency f_c with respect to instantaneous amplitude of modulating signal.

$$\delta = kV_m$$

(2) **Modulation Index (m_f)**

$$m_f = \frac{\text{Maximum frequency deviation } (\delta)}{\text{Modulating frequency } (f_m)}$$

Depending upon the modulating frequency, the modulation index changes as maximum frequency deviation is constant for a given modulating signal.

Q.3. Difference between Modulation Index of AM, FM and PM.

Ans.

Parameters	AM	FM	PM
(1) Formulae	$m_a = \frac{V_m}{V_c}$	$m_f = \frac{\delta}{f_m}$ where $\delta = kf_c v_m$, k = Constant	$m_p = kv_m$ where k = Constant
(2) Practically feasible value	Between 0 to 1	Greater than 0	Greater than 0
(3) Depends on modulating frequency	No	Yes	No
(4) Depends on carrier frequency	No	Yes	No
(5) Depends on carrier voltage	Yes	No	No

Table 5.5

Q.4. In relation to FM, explain :

(i) Maximum frequency deviation.

(ii) Modulation Index.

(iii) Frequency spectrum and bandwidth.

(iv) Pre-emphasis.

(v) De-emphasis.

Ans. For maximum frequency deviation and modulation index refer Q.2.

Frequency spectrum - refer section 5.2.

Bandwidth - refer section 5.3.

Pre-emphasis and De-emphasis - refer section 5.5.

Q.5. List the different methods for FM generation. Sketch the circuit and explain the principle of reactance modulator. Why is direct modulation not preferred for FM generation ?

Ans. Refer section 5.7 and 5.7.1.

Q.6. Explain the Dynamic range of receiver.

Ans. Dynamic Range of Receiver : The receiver should operate over a considerable range of signal strengths.

If a strong signal is passed, it may overload one or more stage in the receiver.

On the other hand a weak signal is usually limited by noise generated within the receiver.

The difference between decibel values of these two signal levels is known as *dynamic range of receiver*.

Definition : Dynamic range is the input power range over which receiver is useful.

The dynamic range of receivers is decided based on following parameters:

(a) Front end noise

(b) Noise figure

(c) Desired signal quality

(d) Gain of receiver.

Q.7. Describe the term FM thresholding.

Ans. FM Thresholding : If the noise gets added to the FM signal, use of amplitude limiters improves S/N ratio during detection. It is known as *FM thresholding*.

Amplitude limiters are circuits which are used to make the amplitude of received FM wave constant.

Q.8. Compare the bandwidth and signal to noise ratio of FM and AM, giving necessary expressions.

Ans.

Parameters	AM	FM
(1) Bandwidth	BW = $2f_m$	BW = $2(\delta + f_m)$
(2) S/N ratio	Less	Large because of 1. Amplitude limiter is used. 2. Pre-emphasis and De-emphasis is done. 3. Increase in δ, increases S/N.

Table 5.6

Q.9. Explain advantages and disadvantages of FM.

Ans. **Advantages :**

- FM is more immune to noise.
- Amplitude limiters used in FM receivers improve the SNR of the received wave.
- FM has more number of side bands compared to AM.
- Bandwidth is more compared to AM.

Disadvantages :

- It is very complex.
- Due to line of sight communication, area covered is less.
- Noise induced depends on frequency of incoming signal, whereas in AM it is independent of frequency.
- Pre-emphasis and De-emphasis circuits are required.

Q.10. How can a frequency modulator be converted to a phase modulator; a phase modulator to a frequency modulator ?

Ans. **For FM from PM :** Refer Indirect Method of FM Generation.

For PM from FM :

Just replace the integrator block by the differentiator block in the block diagram *figure 5.12* to get

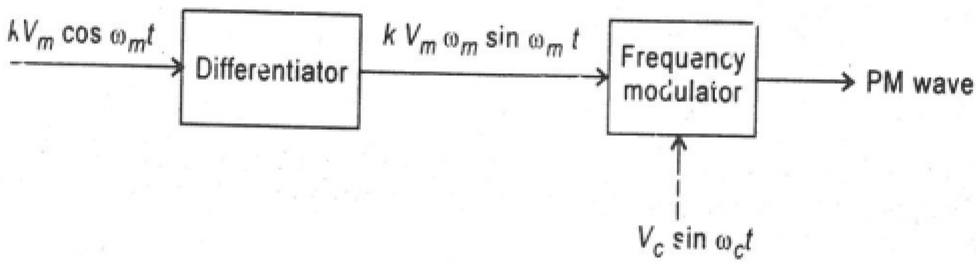


Fig. 5.15

> **Note :** $\frac{d}{dt} \cos t = -\sin t$, *but here we have taken* $\sin t$, *as* $\sin t$ *and* $-\sin t$ *are just* 180^o *out of phase and this does not affect the information it carries in its frequency variations.*

5.11 Solved Problems

Problem 1 : *If modulating frequency in FM system is 400 Hz and modulating voltage is 2.4 V, the modulation index is 60. Calculate maximum deviation. What is modulation index when modulating frequency is reduced to 250 Hz and modulating voltage is simultaneously raised to 3.2 V. Also calculate band width in both cases.*

Given :

$$f_{m_1} = 400 \text{ Hz}$$
$$V_{m_1} = 2.4 \text{ V}$$
$$m_f = 60$$

To find :

$$\delta_1 = ?$$

Solution : **(i)** We know that

$$m_{f_1} = \frac{\delta_1}{f_{m_1}}$$

$$60 = \frac{\delta_1}{400 \text{ Hz}}$$

$$\boxed{\delta_1 = 24 \text{ kHz}}$$

BW of FM is

$$BW_1 = 2(\delta + f_m) = 2(24000 + 400)$$

$$\boxed{BW_1 = 48.8 \text{ kHz}}$$

(ii) In this case $V_{m_2} = 3.2$ V and $f_{m_2} = 250$ Hz.

$$m_{f_2} = \frac{\delta_2}{f_{m_2}}$$

Now,

$$\delta = k\, V_m f_c$$

The term $k\, f_c$ is common in both cases.

Now, from case (i),

$$k\, f_c = \frac{\delta_1}{V_{m_1}} = \frac{24 \text{ kHz}}{2.4 \text{ V}}$$

$$\boxed{k\, f_c = 10 \text{ kHz/V}}$$

Now,

$$\delta_2 = k f_c \times V_{m_2}$$

$$\boxed{\delta_2 = 32 \text{ kHz}}$$

$$\therefore \; m_{f_2} = \frac{\delta_2}{f_{m_2}} = \frac{32 \text{ kHz}}{250 \text{ Hz}}$$

$$\boxed{m_{f_2} = 128}$$

Band width is

$$BW_2 = 2(\delta_2 + f_{m_2}) = 2(32000 + 250)$$

$$\boxed{BW_2 = 64.5 \text{ kHz}}$$

Problem 2 : *In FM system, audio frequency is 1 kHz and audio voltage is 2 V. The deviation is 4 kHz, if the AF voltage is now increased to 8 V and its frequency is dropped to 500 Hz, find the modulation index and B.W. in each case.*

Solution : Same as previous problem just you need to identify what is given.

Given :

$$f_{m_1} = 1 \text{ kHz} \qquad f_{m_2} = 500 \text{ Hz}$$

$$V_{m_1} = 2 \text{ V} \qquad V_{m_2} = 8 \text{ V}$$

$$\delta_1 = 4 \text{ kHz}$$

$$m_{f_1} = \frac{\delta_1}{f_{m_1}} = 4$$

Solve this problem on your own.

Problem 3 : *An FM signal is given by*

$$V = 10 \sin (5 \times 10^8 \, t + 4 \sin 1250t)$$

Find (i) Carrier and modulating frequencies.

(ii) Modulation index and maximum deviation.

(iii) The power dissipated by this wave in a 5 Ω register.

Solution : Comparing the given equation with FM equation.

$$v_{FM} = V_c \sin (\omega_c t + m_f \sin \omega_m t)$$

We get

$$V_c = 10 \text{ volts}$$

$$m_f = 4$$

$$\omega_c = 5 \times 10^8 \text{ rad/s}$$

$$\omega_m = 1250 \text{ rad/s}$$

(i) f_c and f_m

$$f_c = \frac{\omega_c}{2\pi} = \frac{5 \times 10^8}{2\pi}$$

$$\therefore \boxed{f_c = 79.577 \text{ MHz}}$$

$$f_m = \frac{\omega_m}{2\pi} = \frac{1250}{2\pi}$$

$$\therefore \boxed{f_m = 199.045 \text{ Hz}}$$

(ii) δ and m_f

$$\boxed{m_f = 4}$$

Now,

$$m_f = \frac{\delta}{f_m}$$

$$\therefore \delta = m_f \times f_m$$

$$\therefore \boxed{\delta = 796.18 \text{ Hz}}$$

(iii) Power dissipation in 5 Ω register.

Power in FM wave is

$$P_T = \frac{V_c^2}{2R} = \frac{100}{10}$$

$$\boxed{P_T = 10 \text{ watts}}$$

Note : *Some times Bandwidth is also called* ***carrier*** *swing.*

Problem 4 : *If modulating frequency in FM system is 400 kHz and modulating voltage is 2.4 V, and m_f = 60.*

Calculate :

(i) Maximum deviation.

(ii) Modulation index when modulating frequency is reduced to 250 Hz and modulating voltage is simultaneously raised to 3.2 V.

Solution : Same as *Problem 1*. Only difference is f_m = 400 kHz, which is wrongly printed it is 400 Hz only because modulating frequency cannot be so high.

Problem 5 : *If a FM wave is represented by the equation :*

$$V = 10 \sin (8 \times 10^8 + 4 \sin 1000t) \text{ calculate :}$$

(i) Carrier frequency

(ii) Modulating frequency

(iii) Modulation index

(iv) *Maximum deviation.*

(v) *Bandwidth.*

Solution : Refer problem 3.

Problem 6 : *A modulator has a sensitivity of k_p = 4 rad/V. How much frequency deviation does it produce with a sine wave input of 2 V peak at a frequency of 1 kHz ?*

Given :

$$k_p = 4 \text{ rad/V}$$

$$V_m = 2 \text{ V}$$

$$f_c = 1 \text{ kHz}$$

Solution :

Frequency deviation

$$\delta = k f_c V_m$$

$$= 4 \times 1 \times 10^3 \times 2$$

$$= 8 \text{ kHz}$$

Problem 7 : *The carrier frequency of FM broadcast transmitter is 100 MHz and maximum frequency deviation is 75 kHz. It the highest audio frequency modulating carrier is 15 kHz. What is the bandwidth of the signal ?*

Given :

$$f_c = 100 \text{ MHz}$$

$$\delta = 75 \text{ kHz}$$

$$f_m = 15 \text{ kHz}$$

Solution :

$$\text{BW} = 2(\delta + f_m)$$

$$= 2(75 + 15) = 180 \text{ kHz}$$

❋❋❋

6 AM AND FM DETECTORS

Topic	Theory imp	Oral imp
Introduction		
Envelope Detector	☆	☆
Practical Diode Detector	☆☆	☆
Distortion in Diode Detector	☆☆☆	☆☆
Slope Detector	☆	☆
Balanced Slope Detector	☆☆☆☆	☆☆
Foster - Seeley	☆☆☆☆	☆☆
Ratio Detector	☆☆	☆☆☆
FAQ's	☆	☆☆
Problems	----	----

6.0 Introduction

Detectors are circuit which detect the original signal from the incoming signal. They are an integral part of receivers.

Input $\longrightarrow$ $\boxed{\text{Detector}}$ $\longrightarrow$ Original signal

In this chapter we are going to discuss two types of detectors :

(i) AM detectors

(ii) FM detectors.

Note : *PM detectors are not in syllabus.*

6.1 AM Detectors/ Demodulators (DSB-FC Detectors)

6.1.1 Simple Detector/ Peak Detector/ Envelope Detector

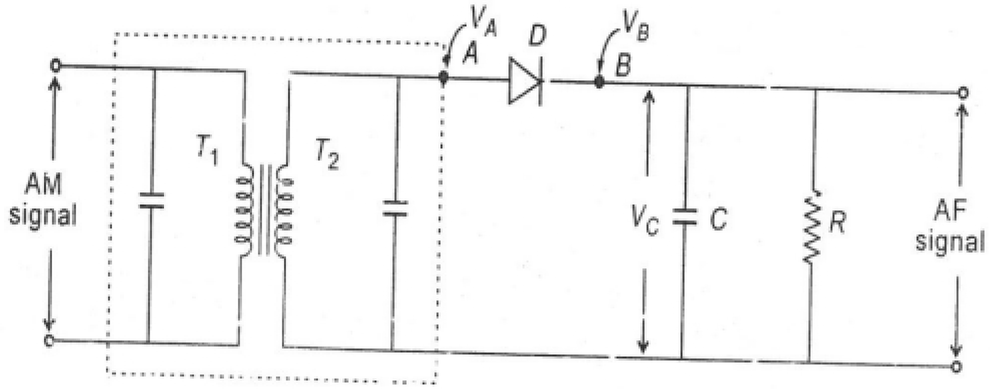


Fig. 6.1

(a) AM wave at the input.

(b) Half wave rectified signal.

(c) Output of the diode.

(d) Output of envelope detector.

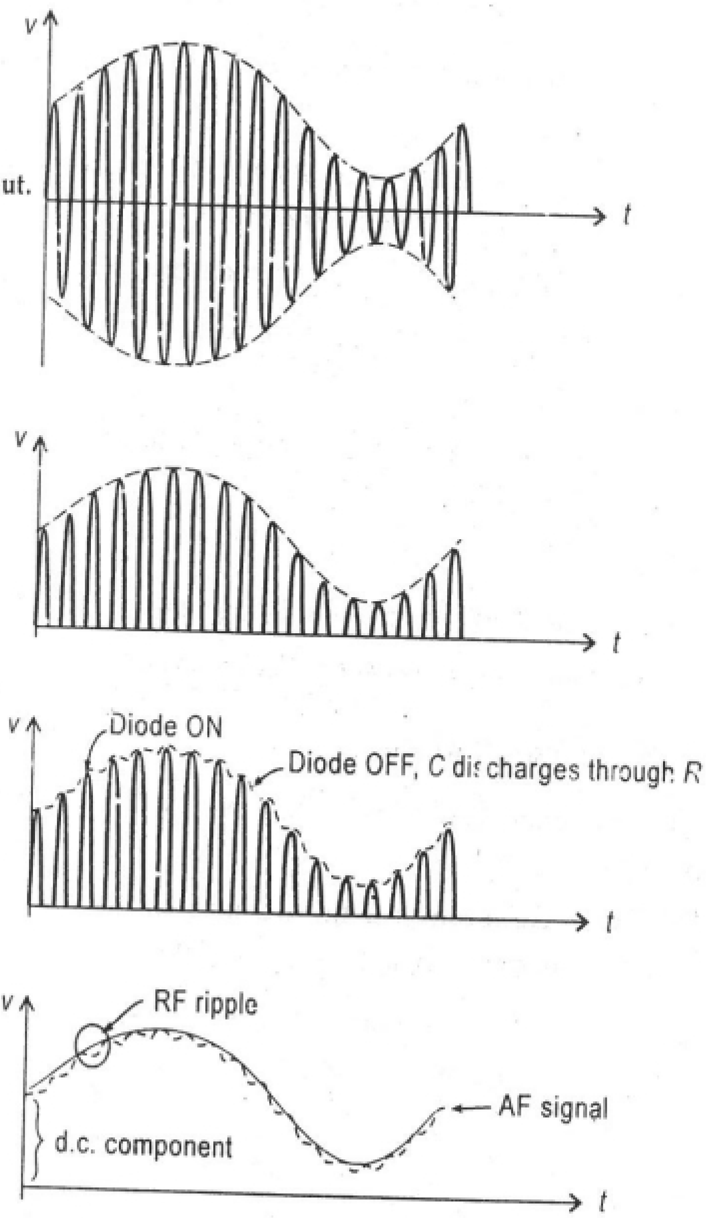


Fig. 6.2

The detector detects the AF signal i.e. separates out modulating signal from AM wave.

Operation

- During positive half cycle of AM input signal, the diode is forward biased and the capacitor charges rapidly to the peak of input signal.
- When the input signal falls, at some point of time $V_A < V_B$, i.e. diode becomes reverse biased and capacitor starts discharging via resistor R.
- The time constant of RC circuit is kept high such that by the time C discharges slightly, next positive half cycle of AM signal makes diode forward biased i.e. V_A again becomes high.
- The diode becomes forward biased and capacitor charges to new peak of input carrier signal.
- This process continues and we get modulating voltage across R as shown in *figure 6.2*.
- The *figure 6.2* shows that output of diode detector follows the envelope of AM wave, therefore it is called *Envelope Detector.*
- The output contains RF ripple, d.c. component and AF signal.
- The RF ripple can be reduced by keeping RC time constant large.

Advantages

Since detector requires very few components it is cheap and normally used in low cost radio receiver.

Disadvantages

(1) At the output of detector, we get modulating signal along with large RF ripple.

(2) The d.c. component is also present in the detector output.

Limitation on the Value of RC

(1) Low RC Time Constant : If the RC time constant is low the output contains large RF ripple.

(2) High RC Time Constant : If the RC time constant is high then it creates a problem called *diagonal clipping* (explained later).

Hence for proper operation of detector, select R and C such that the following condition is satisfied.

$$\frac{1}{RC} \geq \frac{\omega_m \cdot m}{\sqrt{1-m^2}}$$

where m = Modulation index

ω_m = Modulating frequency

6.1.2 Practical Diode Detector

Q.1. Sketch the circuit diagram of practical diode detector and explain its working.

Q.2. Sketch the circuit of a practical diode detector and explain its working. What is negative peak clipping ? Calculate maximum modulation index that the above detector can tolerate without causing negative peak clipping.

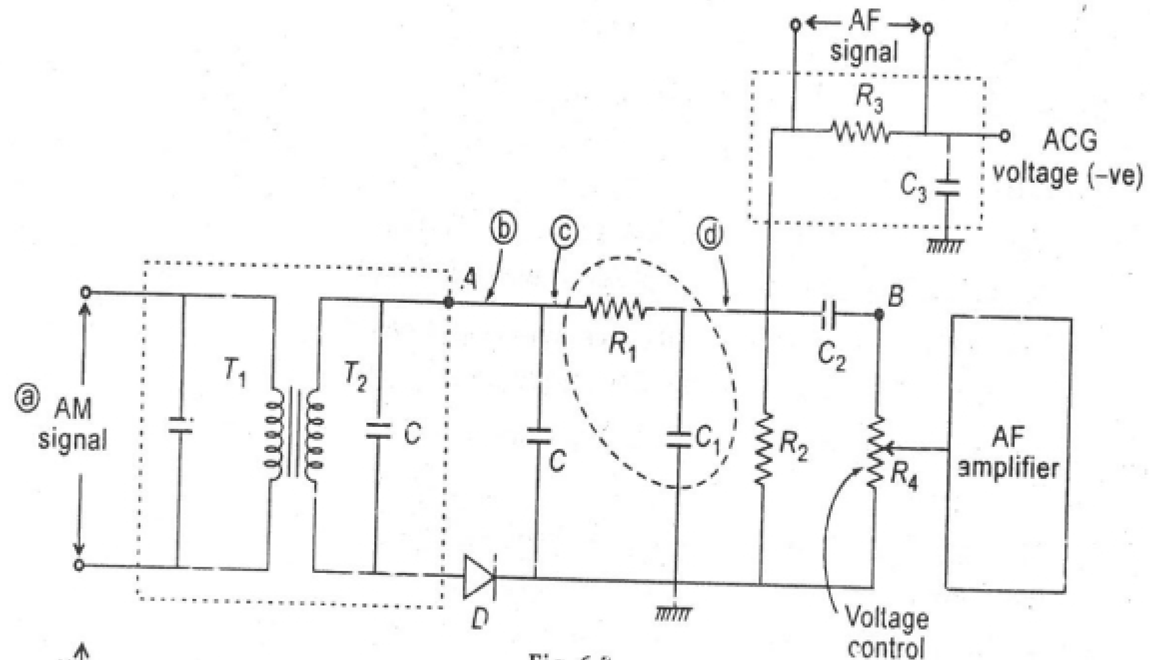
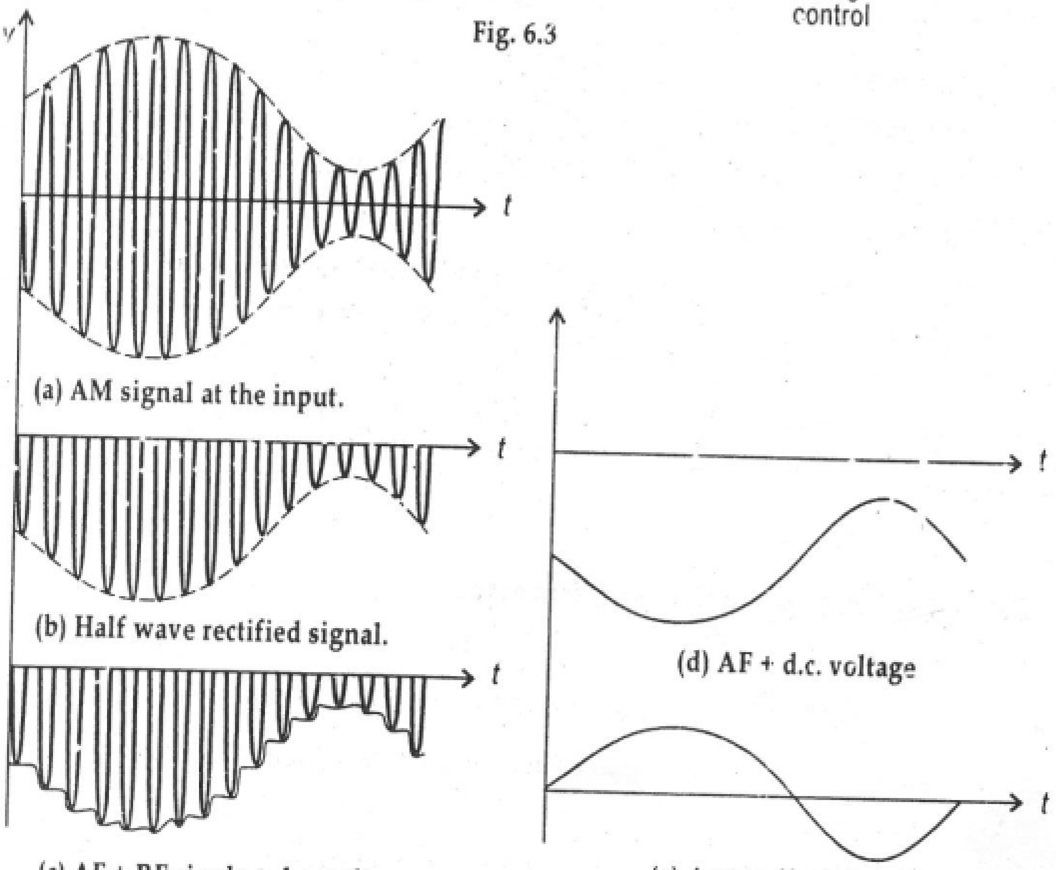


Fig. 6.3



(a) AM signal at the input.

(b) Half wave rectified signal.

(c) AF + RF ripple + d.c. voltage

(d) AF + d.c. voltage

(e) Approximated output

Fig. 6.4

Circuit Description

Circuit diagram of practical diode detector is shown in *figure 6.3*.

Here detector diode D is connected such that –ve d.c. voltage is available for Gain Control of amplifier.

The detector consists of

(1) Low pass filter formed by R_1C_1 to bypass RF ripples.

(2) Low pass filter formed by R_3C_3 to bypass AF signal.

(3) Blocking capacitor C_2 to block d.c. voltage.

Operation

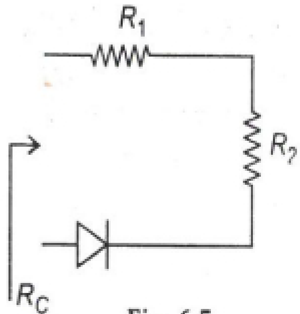
- During negative half cycle of AM wave, the diode is forward biased. Therefore it detects the negative envelope of AM waveform. [*Figure 6.4(b)*]
- The R_1C_1 acts as low pass filter. Therefore it bypasses high frequency RF signal to ground and passes only the low frequency AF signal.
- So, at the output of this filter we get only the AF signal and –ve d.c. voltage. [*Figure 6.4(d)*]
- The capacitor C_2 blocks the dc voltage and passes the AF signal towards R_4, the output of which is given to input of AF amplifier..
- R_4 acts as voltage control of receiver.
- The voltage across R_2 contain AF signal and –ve d.c. voltage which is applied o low pass filter formed by R_3 and C_3.
- The filter will bypass AF signal to ground and at the output of low pass filter we get –ve d.c. voltage.
- The magnitude of this voltage is proportional to strength of carrier. This –ve voltage is called *AGC voltage* which is used to control the gain of RF and IF amplifier.

> Note : *AGC circuit, RF and IF amplifiers are explained in the next chapter.*

Distortion in Practical Diode Detector

(1) Negative Peak Clipping :

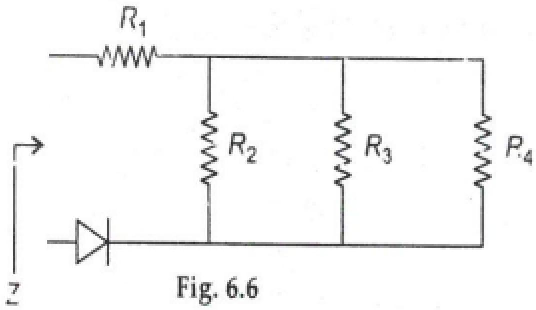
- This problem arises because d.c. load of diode D is not equal to a.c. load.
- To find d.c. and a.c. loading on detector, lets first draw a.c. and d.c. equivalent circuits.
- **D.C. Equivalent Circuit :** To draw d.c. equivalent circuits, all capacitors are considered to be open circuited



$\therefore R_C = R_1 + R_2$

Fig. 6.5

- **A.C. Equivalent Circuit** : To draw a.c. equivalent circuit, capacitor C and C_1 acts as open circuit for AF and capacitor C_2 and C_3 acts as short circuit.



$\therefore Z = R_1 + (R_2 \,||\, R_3 \,||\, R_4)$

Fig. 6.6

- From above equations, it is seen that

$$\therefore R_C > Z \qquad \text{..... (1)}$$

- As we already know, modulation index is

$$m = \frac{V_m}{V_C}$$

- The demodulation index at the receiver is defined as

$$m_d = \frac{I_m}{I_C} \quad \text{where } I_m = \frac{V_m}{Z} = \text{Modulating current}$$

$$I_C = \frac{V_C}{R_C} = \text{Carrier current}$$

From equation (1), we can say that if $V_m \approx V_c$ then,

$$I_m > I_C$$

Therefore we get –ve peak clipping as shown in *figure* 6.7.

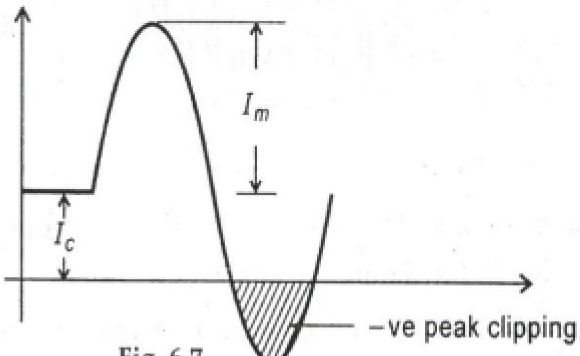


Fig. 6.7

To avoid –ve peak clipping

$$m_d = \frac{I_m}{I_C} = \frac{V_m}{Z} \times \frac{R_C}{V_C} = m\frac{R_C}{Z}$$

$$\because R_C > Z$$

$$\therefore m_d > m$$

- It means after detection, the demodulation index increases, i.e. if m is 100%, then $m_d > 100\%$.
- Therefore during –ve half cycle, AF signal gets clipped.
- To avoid –ve peak clipping demodulation index should be less than or equal to 1 i.e. $m_d \leq 1$

$$\therefore m \cdot \frac{R_C}{Z} < 1$$

$$\therefore m \leq \frac{Z}{R_C}$$

- If the signal is transmitted with $m \leq Z/R_C$ then –ve peak clipping is avoided.

(2) Diagonal Clipping :

- In diode detectors, the RC time constant should be small such that the capacitor does not discharge by large amount.
- The capacitor's charging and discharging should follow the envelope of AM wave.
- If the RC time constant is too high, then the output of detector does not follow the fast changes in envelope of AM wave.
- This results in diagonal clipping as shown in *figure 6.8*.

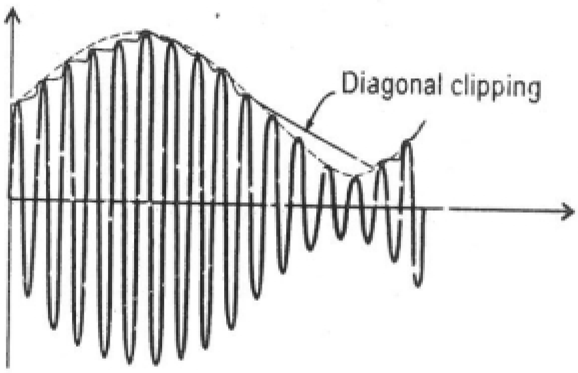


Fig. 6.8

- Because of high RC time constant, the slope of output waveform cannot follow the trailing slope of envelope.
- Due to this problem, we get distorted output.

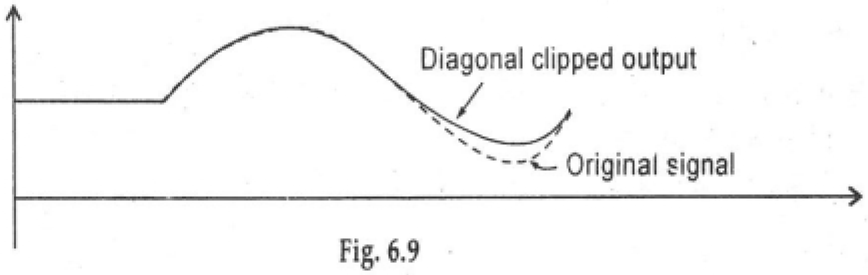


Fig. 6.9

- To avoid diagonal clipping, the discharging time constant should not be too high.
- Also, the modulating index used at transmitter should be less than 60%.

6.2 FM Detectors

FM detectors or demodulators are circuits which take as input an FM wave and gives the original signal as the output.

Basic Principle of FM Demodulators/Detectors

The basic principle of FM detector is to first convert the frequency variations to amplitude variations and then apply it to an AM detector circuit.

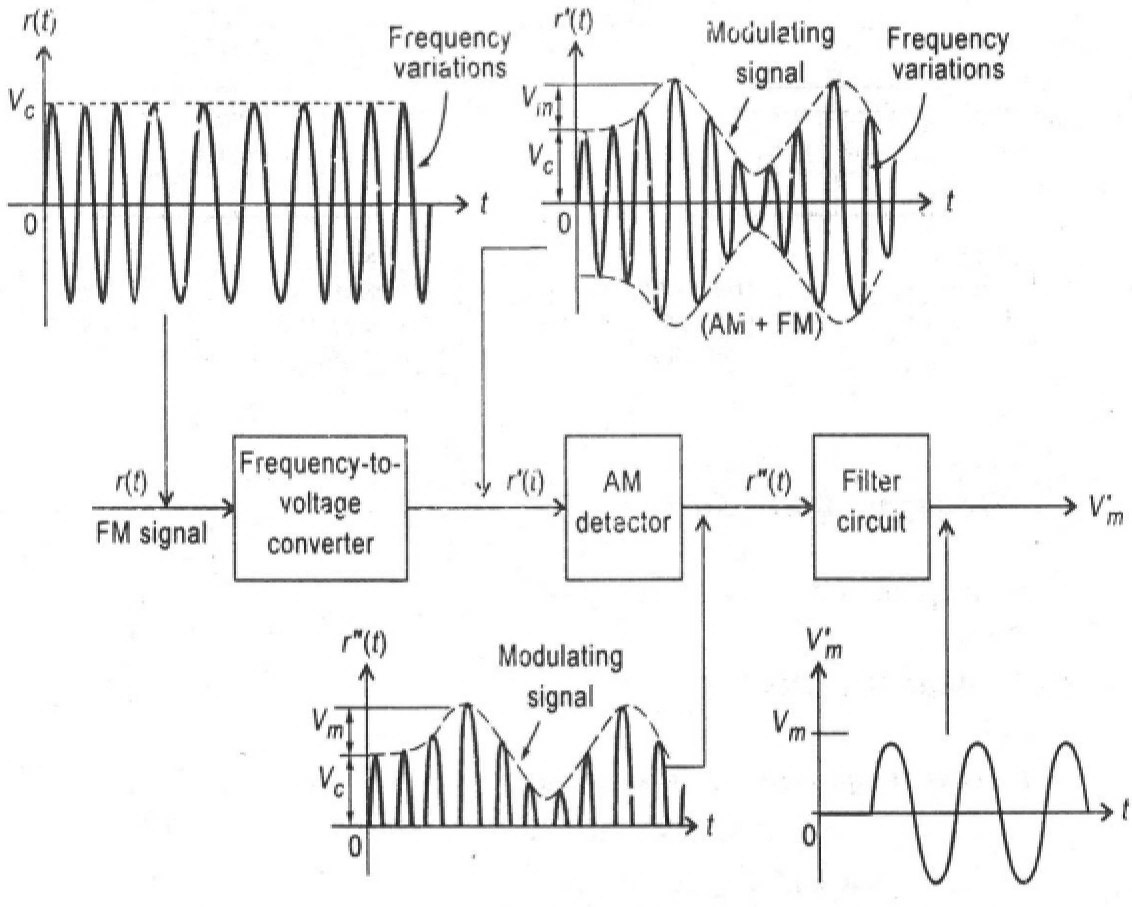


Fig. 6.10 : Block diagram of an FM demodulator based on its basic principle.

In the *figure 6.10*.

- Frequency to voltage converter converts frequency variations to voltage (amplitude variations).
- AM detector detects the original signal.
- Filter circuit is used to make the output smooth. (optional)

Note : *The frequency to voltage converter circuit is normally a tuned circuit or a tank circuit which has an inductor and capacitor connected in parallel.*

To understand how it works please refer FAQ's.

For the time being just keep in mind tank circuit takes as input FM wave and outputs an (AM + FM) wave.

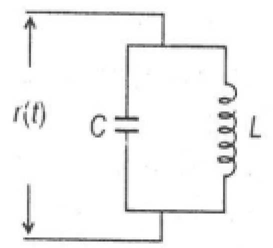


Fig. 6.11

The classification of FM detectors is

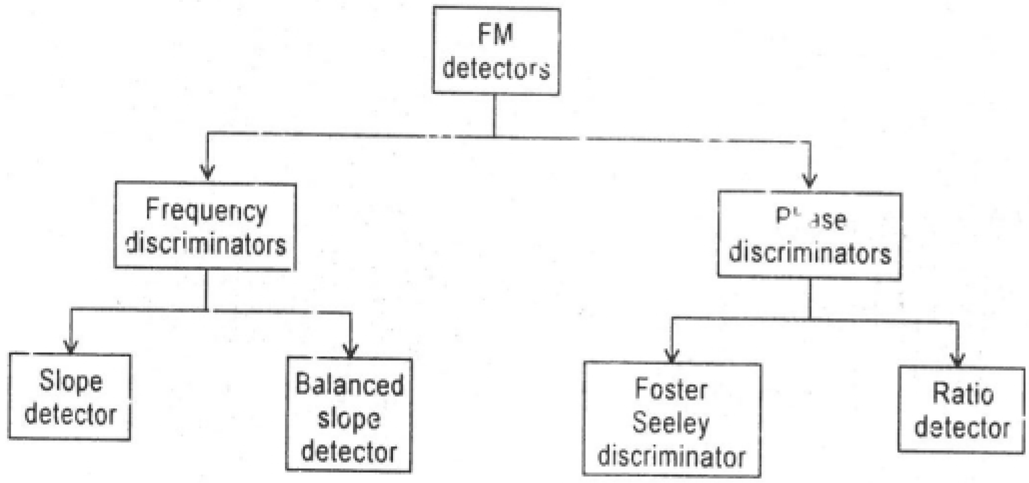


Fig. 6.12

6.2.1 Frequency Discriminator

They use frequency deviation to obtain the output.

6.2.1.1 Slope Detector

- Also called *single tuned slope detector*.

Circuit Diagram

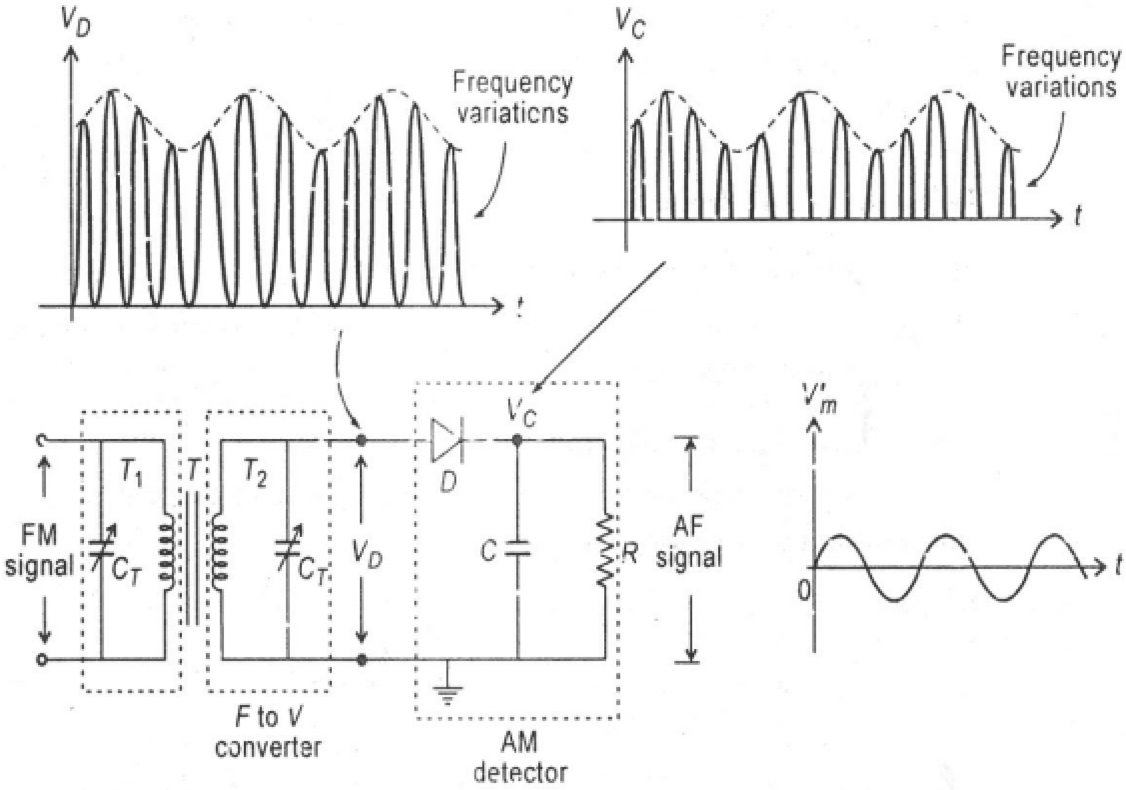


Fig. 6.13 : Circuit diagram of a single-tuned slope detector.

Working

- The primary winding of the transformer T i.e. tank circuit 1 (T_1)is used to select the desired frequency (It is tuned to f_c i.e. signal frequency).
- Tank circuit T_2 acts as frequency to voltage converter and converts frequency variations to amplitude variations. (It is tuned to $f_0 > f_c$).
- Diode D clips the –ve half cycle of the signal.
- Capacitor C detects the envelope and the original signal is obtained.

Advantages

- Simple in design.
- Low in cost.

Disadvantages

- Since the response of the tank circuit is not always linear (which is practically very difficult) the output is often distorted.
- There is no inherent capability to reject noise.

Applications

- Normally not used in practice.

> **Note :** *Slope detector and simple diode detector are almost similar if their circuit diagrams are viewed properly. The main difference is that in simple diode detector both the tank circuits are tuned to the signal frequency (f_c) and in slope detector, only T_1 is tuned to f_c, T_2 is tuned to $f_0 > f_c$.*

6.2.1.2 Balanced Slope Detector

Q.1. ***Explain the working of balanced slope detector.***

Q.2. ***Explain the operation of balanced slope detector using a circuit diagram and a responsive characteristics.***

- Also called *stragger tuned/double tuned/ dual slope detector.*
- Improved version of single slope detector as the response of the tank circuit is made more linear by using two tank circuits.

Circuit Diagram

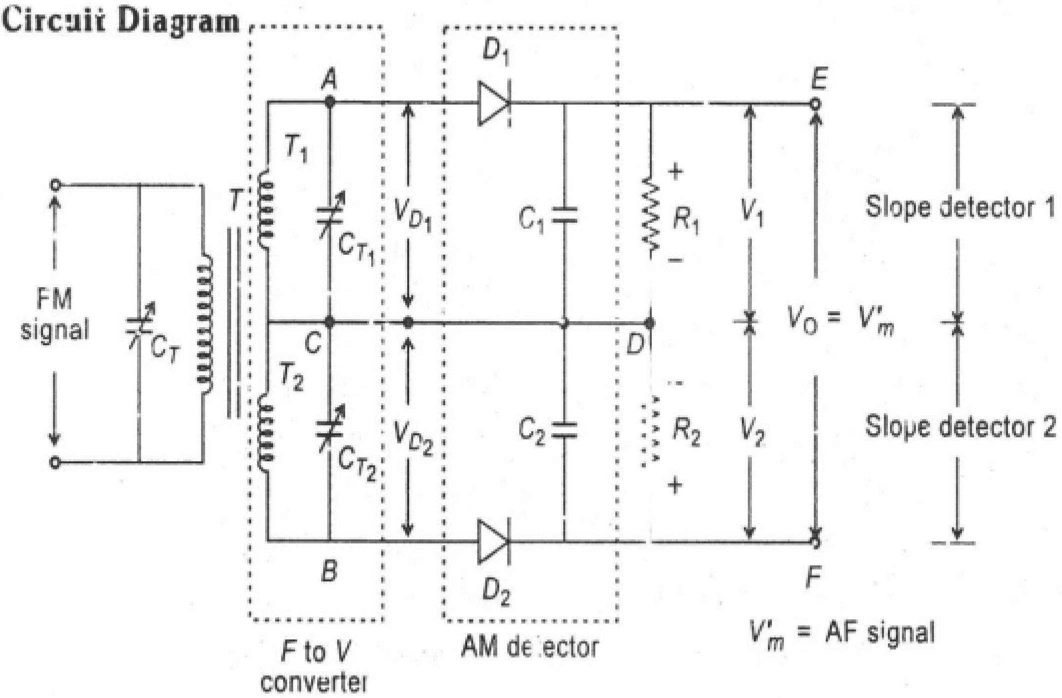


Fig. 6.14 : Circuit diagram of a balanced slope detector.

Description

- Two tank circuits (slope detectors) are used, each one is tuned at a different frequency.
- Slope detector 1 is tuned at a frequency f_1 such that

$$\boxed{f_1 = f_C + \Delta f}$$

$$f_C = \text{Carrier frequency}$$

$$\Delta f = \text{Small amount}$$

It is used to detect the positive cycle of the *modulating* signal

- Slope detector 2 is tuned at a frequency f_2 such that

$$\boxed{f_2 = f_C - \Delta f}$$

It is used to detect the negative half cycle of the *modulating* signal.

Working

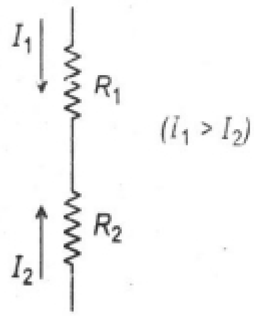
- From the circuit we come to know that, $V_O = V_1 - V_2$

We will understand the working of the circuit by analysing two cases.

(a) Case (i) : During positive half cycle of modulating signal

- Since it is the positive cycle, the frequency of FM wave is more i.e. between f_C and $f_C + \Delta f$.
- Thus, slope detector 1 is working.
- Thus, voltage V_1 is obtained across R_1.
- Also, little amount of current flows through R_2 as well.
- Thus, small voltage V_2 is obtained at R_2 but it is negative.
- Output voltage is

$$V_O = V_1 - V_2$$

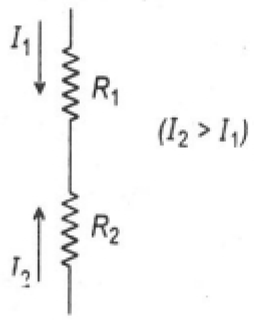


- Thus output is positive.

(b) Case (ii) : During negative cycle of the modulating signal

- Works exactly in the reverse order of case (i).
- Since, it is negative cycle, the frequency of FM wave is less i.e. between $f_C - \Delta f$ and f_C.
- Thus, slope detector 2 is working.
- Thus, voltage across R_2 is generated but in opposite direction i.e. $(-V_2)$.
- Again, some amount of voltage is developed at R_1 also i.e. (V_1).
- Output voltage is

$$\therefore\ V_O = V_1 - V_2$$



- Thus output is negative.

Overall Response Graphically

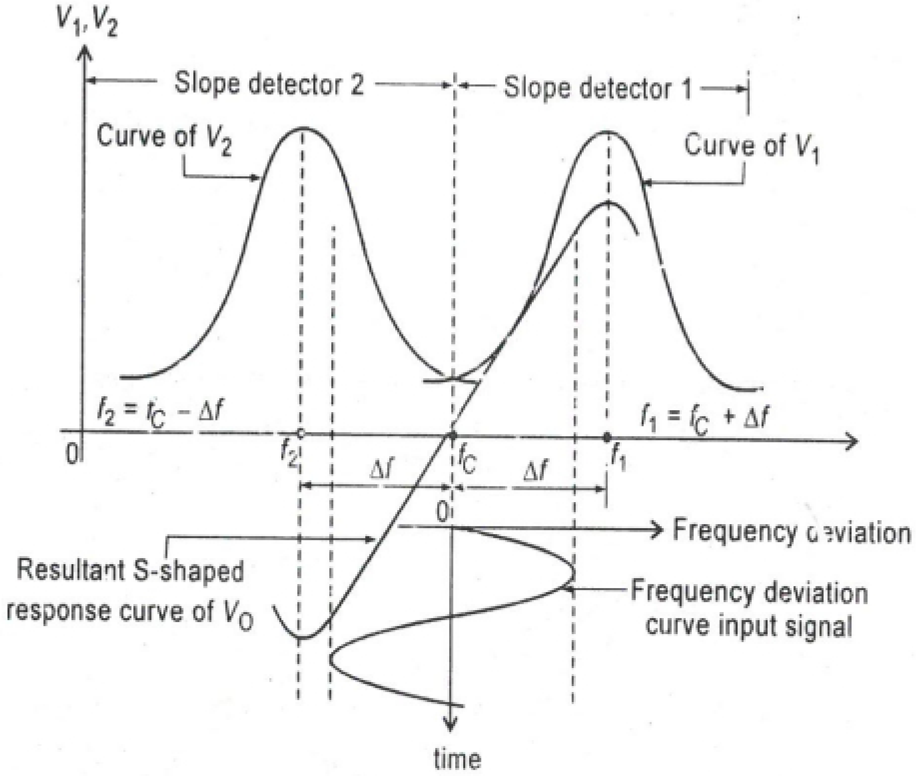


Fig. 6.15

Advantages

- Linearity is better than single slope detector.

Disadvantages

- Tuning all the tank circuits simultaneously is difficult.
- No inherent capability to reject noise.

Applications

- Used in some receivers, but they are outdated.

6.2.2 Phase Discriminator

- They use phase variations to get the output.
- All the drawbacks of frequency discriminators are overcome.

6.2.2.1 Foster-Seeley Discriminator

Q.1. Explain the working of Foster-Seeley discriminator with a neat circuit diagram and phasor diagram.

Q.2. Sketch the circuit and explain working of Foster-Seeley discriminator. Give phasor diagrams for

(1) $f_{in} = f_c$ (2) $f_{in} > f_c$ (3) $f_{in} < f_c$

Q.3. Explain Foster-Seeley discriminator in detail.

> **Note :** *This circuit is really very difficult and to remember it one will definitely have to understand it. The best possible way to remember this circuit is just to remember the steps.*
>
> *If it is asked for 10 marks, then, give the circuit diagram and do not prove any of the steps given in circuit analysis, just write down the final result of every step and then explain working using phasor diagram.*

- The primary and two secondary windings are tuned to the same frequency f_C.

Circuit Diagram

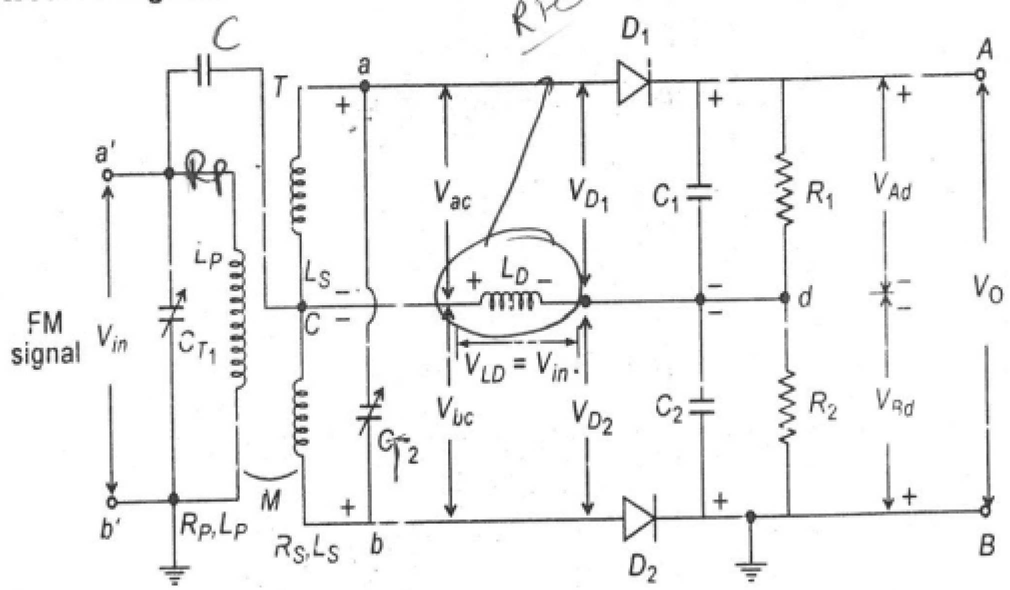


Fig. 6.16 : Circuit diagram of a Foster-Seeley phase discriminator.

Circuit Analysis

Here we will have to prove and find a lot of things so we will start one by one stepwise.

Step (i) : To prove $V_O = V_{D_1} - V_{D_2}$

Proof : From the *figure 6.16*

$$V_O = V_{Ad} - V_{Bd}$$

Since the output voltages (V_{Ad} and V_{Bd}) depend on input diode voltage (V_{D_1} and V_{D_2})

$$\therefore\ V_{Ad} \propto V_{D_1} \text{ and } V_{Bd} \propto V_{D_2}$$

$$\therefore\ V_O \propto V_{D_1} - V_{D_2}$$

$$\therefore \; V_O = k(V_{D_1} - V_{D_2})$$

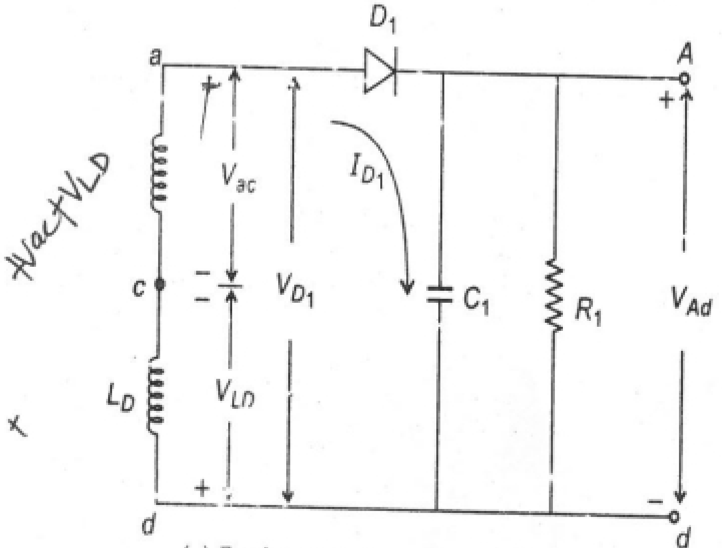
where k = Constant

Let $k = 1$

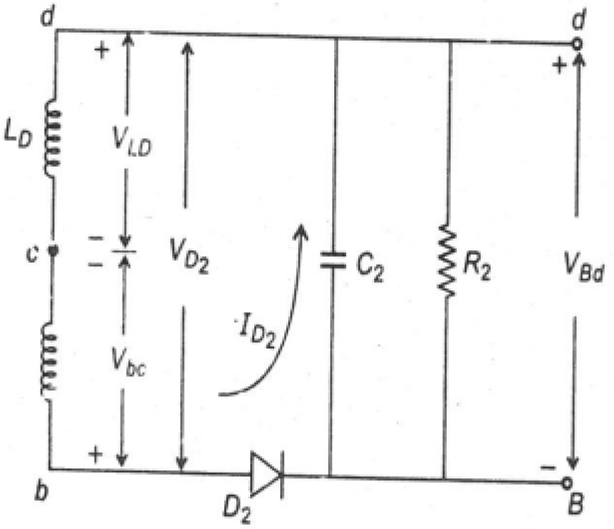
$$\therefore \; \boxed{V_O = V_{D_1} - V_{D_2}}$$

Step (ii) : To find input diode voltages (V_{D_1} and V_{D_2})

The original diagram can be split as



(a) For input voltage V_{D_1} to diode D_1.



(b) For input voltage V_{D_2} to diode D_2.

Fig. 6.17 : Part of Foster-Seeley phase discriminator for input diode voltages.

From *figure 6.17* we can write

$$\left.\begin{aligned} &\boxed{V_{D_1} = V_{ac} + V_{LD}} \\ \text{and } &\boxed{V_{D_2} = V_{bc} + V_{LD}} \end{aligned}\right\} \text{ by applying KVL}$$

Step (iii) : To find voltage across RFC

We will consider a part of the circuit to find V_{LD} (voltage across RFC)

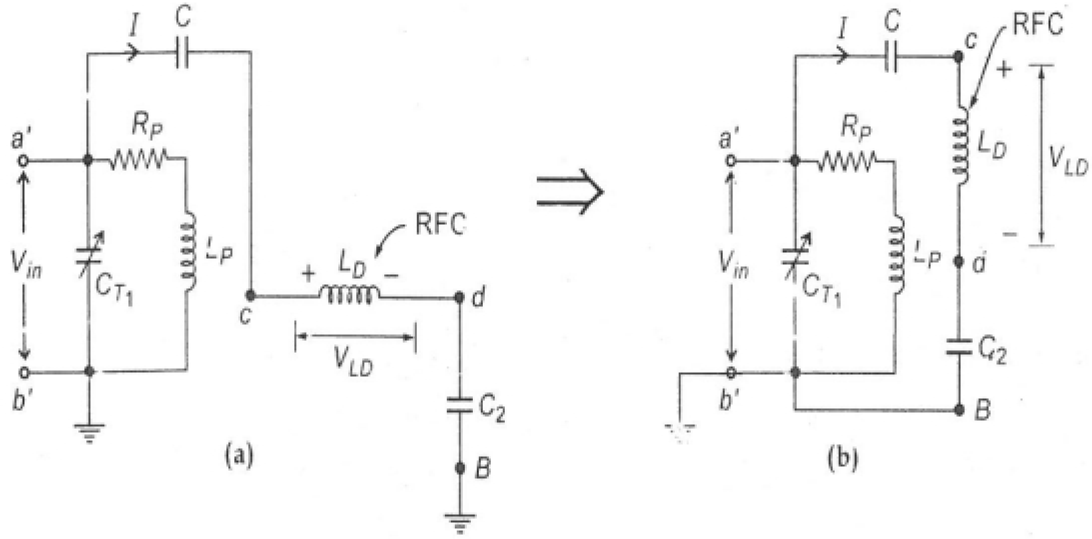


Fig. 6.18 : (a) Part of Foster-Seeley phase discriminator to obtain voltage across RFC, V_{LD} and (b) Redrawn circuit diagram (a) to show how the loop is closed.

Vac ≠ VLD

Vac = VLD =

− VLD

Let X_C = Reactance of C

X_{C_2} = Reactance of C_2

X_{LD} = Reactance of RFC

The current I is

$$i = \frac{V_{in}}{X_C + X_{LD} + X_{C_2}} \qquad \text{(Ohm's law)}$$

$$I = \frac{V_{in}}{\left(\frac{1}{j\omega C}\right) + j\omega L_D + \left(\frac{1}{j\omega C_2}\right)}$$

$$I = \frac{V_{in}}{j\omega L_D - j\left(\frac{1}{\omega C} + \frac{1}{\omega C_2}\right)} \qquad \text{..... (1)}$$

∴ The voltage drop across L_D (RFC) is

$$V_{LD} = I \times X_{LD}$$

$$\therefore\ V_{LD} = \frac{V_{in} \times j\omega L_D}{j\omega L_D - j\left(\frac{1}{\omega C} + \frac{1}{\omega C_2}\right)} \qquad \text{..... (2)}$$

Now, always the RFC has a very high reactance as compared to the two capacitors.

Thus we can say

$$j\omega L_D >>> j\left(\frac{1}{\omega C} + \frac{1}{\omega C_2}\right)$$

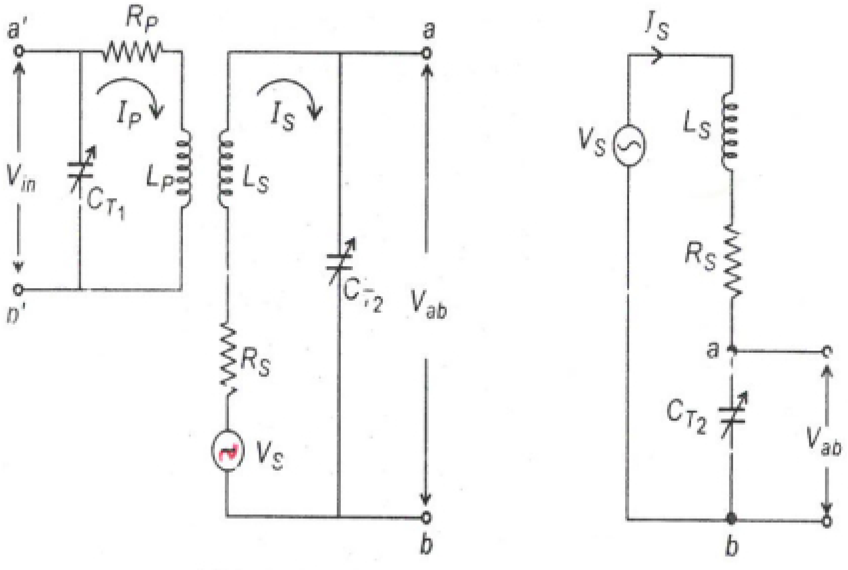
$$\therefore\ V_{LD} \cong \frac{V_{in} \times \cancel{j\omega L_D}}{\cancel{j\omega L_D}}$$

$$\therefore\ \boxed{V_{LD} \cong V_{in}}$$

Thus V_{LD} is nearly same as input voltage.

Step (iv) : To find voltage V_{ab}

To find V_{ab}, we will have to draw the equivalent circuit of the transformer *T*. *Figure 6.19* shows the equivalent circuit.



(a) (b)

Fig. 6.19 : (a) Equivalent circuit of transformer for secondary voltage V_{ab} and (b) Redrawn circuit diagram (a).

In the *figure 6.19*

V_S = induced voltage in the secondary winding due to current I_P in primary winding.

Let M = Mutual inductance.

Now by theory on transformers,

$$V_S = -j\omega M I_P \quad \left\{\because \text{ the two windings are wound in opposite direction}\right\}$$

$$\text{and} \quad I_P = \frac{V_{in}}{j\omega L_P} \quad \left\{\text{Assuming } R_S \text{ and } R_P \text{ are negligible}\right\}$$

$$\therefore\ V_S = -\cancel{j}\omega M \frac{V_{in}}{\cancel{j}\omega L_P}$$

$$\therefore\ V_S = -\frac{MV_{in}}{L_P}$$

Now from *figure 6.19(b)*

$$\therefore\ V_{ab} = -\frac{V_S}{jX_{L_S} + R_S - jX_{CT_2}} \times (-jX_{CT_2})$$

$$\therefore\ V_{ab} = \cancel{-}\frac{V_{in}\, M\,(\cancel{-}jX_{CT_2})}{L_P(jX_{L_S} + R_S - jX_{CT_2})}$$

r 7 Figure

$$\therefore\ V_{ab} = \frac{jMV_{in}}{L_P} \cdot \frac{X_{CT_2}}{R_S + j(X_{L_S} - X_{CT_2})}$$

Let $\boxed{X_S = X_{L_S} - X_{CT_2}}$

$$\therefore\ V_{ab} = \frac{V_{in} X_{CT_2} M \angle 90^\circ}{(R_S + jX_S)L_P}$$

$\left\{ \because\ j \text{ represents a vector at an angle } 90^\circ \text{ in phasor diagram} \right\}$

Step (v) : To find V_{ac} and V_{bc}

Since, the secondary winding is divided into two equal halves, the voltages are also equally distributed.

$$\therefore \boxed{V_{ac} = \frac{1}{2} V_{ab}}$$

and $\boxed{V_{bc} = -\frac{1}{2} V_{ab}}$

Step (vi) : To find V_{D_1} and V_{D_2}

:urrent I_P

From step (ii) and (v) and step (iv)

$$V_{D_1} = V_{ac} + V_{LD}$$

and $V_{D_2} = V_{bc} + V_{LD}$

$$\therefore\ V_{LD} = V_{in}$$

$$\therefore\ V_{ac} = \frac{1}{2} V_{ab}$$

and $V_{bc} = -\frac{1}{2} V_{ab}$

$$\left.\begin{aligned} \therefore \quad \boxed{V_{D_1} = \frac{1}{2}\frac{V_{in}X_{CT_2}}{R_S + jX_S}\cdot\frac{M\angle 90^\circ}{L_P} + V_{in}} \\ \therefore \quad \boxed{V_{D_2} = -\frac{1}{2}\frac{V_{in}X_{CT_2}}{R_S + jX_S}\cdot\frac{M\angle 90^\circ}{L_P} + V_{in}} \end{aligned}\right\} \quad \ldots\ldots (A)$$

From the above equations we can conclude that the voltages V_{D_1} and V_{D_2} depend on X_S where

$$X_S = X_{L_S} - X_{CT_2}$$

Since, X_{L_S} and X_{CT_2} depends on the incoming frequency, the output voltage,

$$\boxed{V_O \propto \text{Incoming frequency}}$$

Working

The working of this circuit can be best understood by using phasor diagram.

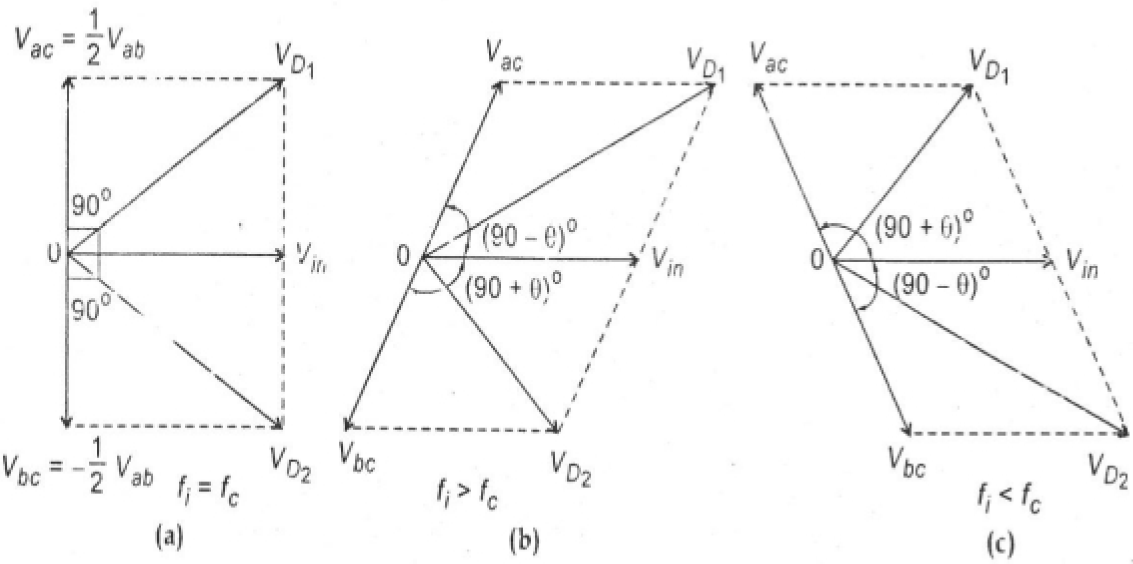


Fig. 6.20 : Phasor diagrams for diode input voltages V_{D_1} and V_{D_2}, in Foster Seeley phase discriminator : (a) When $f_i = f_C$ (b) When $f_i > f_C$ and (c) When $f_i < f_C$.

We will analyze this circuit using three cases.

Case (i) : $f_i = f_C$ (Incoming frequency = Carrier frequency)

- In this case, resonance will occur and

$$X_{L_S} = X_{CT_2} \text{ and } X_S = 0$$

- Now, the equations of V_{D_1} and V_{D_2} can be represented in phasor form as shown in *figure 6.20(a)*.
- Thus, magnitude wise

$$V_{D_1} = V_{D_2} \quad \therefore \; V_O = 0$$

Case (ii) : $f_i > f_C$

- Here, $X_{L_S} > X_{CT_2}$ and $X_S > 0$
- Magnitude wise $V_{D_1} > V_{D_2}$ $\therefore$ $\boxed{V_O = +\text{ve}}$

Case (iii) : $f_i < f_C$

- Here, $X_{L_S} < X_{CT_2}$ and $X_S < 0$
- Magnitude wise $V_{D_1} < V_{D_2}$ $\therefore$ $\boxed{V_O = -\text{ve}}$

Advantages

- Linearity is better.
- It is easy to align compared to balanced slope detector as there are only two tuned circuits tuned at same frequency.

Disadvantages

- It responds to amplitude variations received in FM wave (due to noise).

6.2.2.2 Ratio Detector

Q.1. What is the advantage of ratio detector over Forster Seely discriminator ?

Q.2. Sketch the circuit diagram of a ratio detector and explain how it demodulates an FM signal. How amplitude limiting is achieved in this ?

- Ratio detector does not respond to the amplitude variations in a received FM signal.
- Main motive of ratio detector is to maintain

$$V_{AB} = V_{D_1} + V_{D_2} = \text{Constant}$$

Circuit Diagram

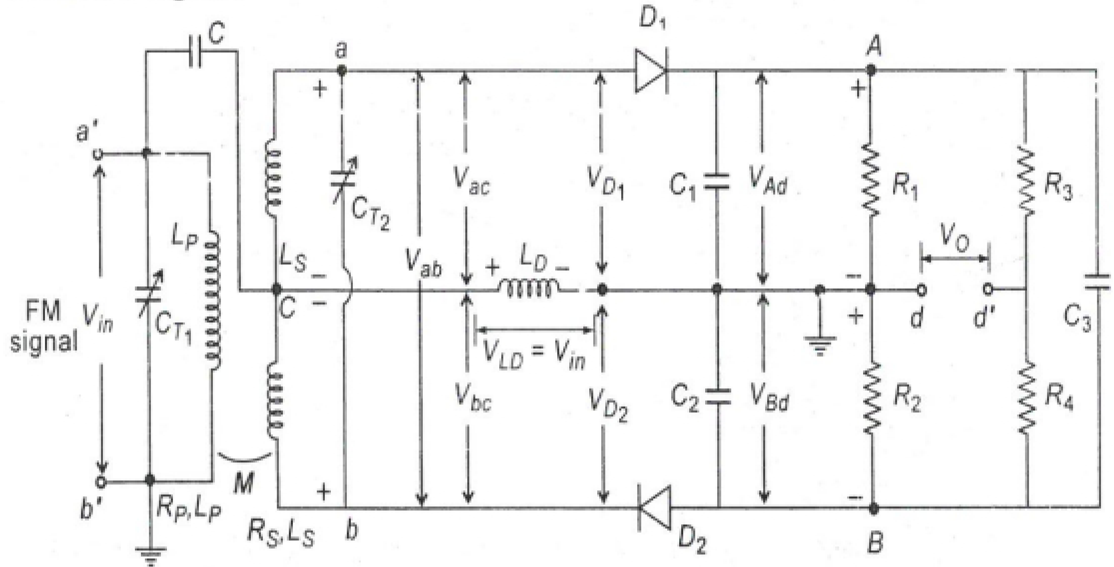


Fig. 6.21 : Circuit diagram for a ratio detector.

Changes made in Foster Seeley to get ratio detector are

- The diode D_2 is reversed so that V_{Ad} and V_{Bd} come in series and

$$V_{AB} = V_{Ad} + V_{Bd}$$

$$\therefore \boxed{V_{AB} = V_{D_1} + V_{D_2}}$$

- The capacitor C_3 is used to sum up V_{ad} and V_{bd} so that the output V_{AB} is a constant. According to a theory, $V_{D_1} + V_{D_2}$ is always constant in Forster Seeley or ratio detector circuit.

- Output is $V_O = V_{Bd} - V_{Bd'}$
  V_{AB} is always constant.

- R_3 and R_4 are added to provide discharge path to C_3 ($R_3 = R_4$).

Working

Again phasor diagrams are used for understanding the working.

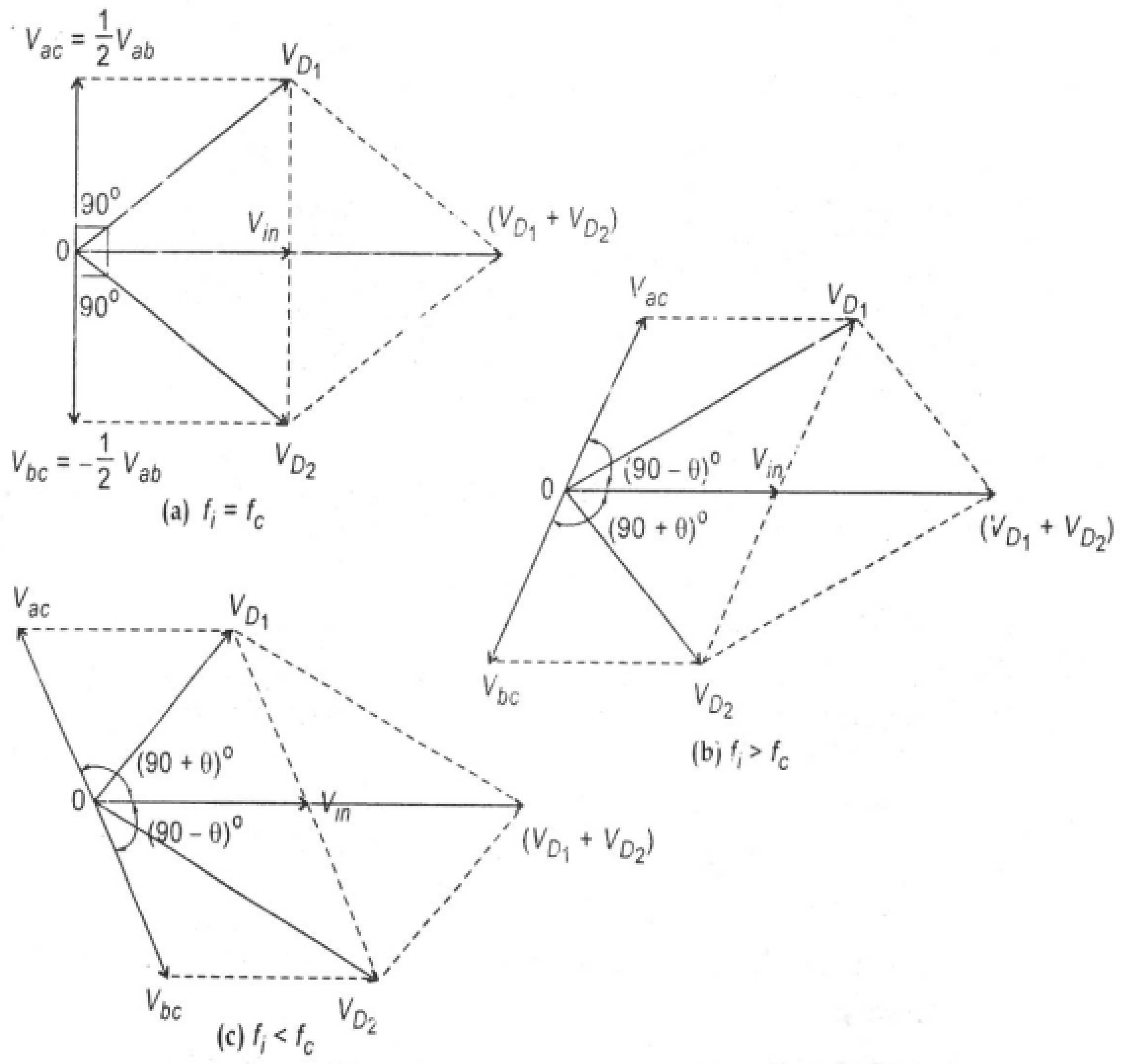


Fig. 6.22 : Phasor diagrams showing sum of diode input voltages V_{D_1} and V_{D_2} at different incoming frequencies : (a) When $f_i = f_c$ (b) When $f_i > f_c$ and (c) When $f_i < f_c$.

- The working of ratio detector is very much similar to Foster Seeley.
- We just now need to prove that,

$$V_O \propto (V_{D_1} - V_{D_2})$$

Proof :

$$V_O = V_{Bd'} - V_{Bd}$$

We know,

$$V_{Ad} = V_{D_1}$$

$$V_{Bd} = V_{D_2}$$

$$\text{and } V_{AB} = V_{Ad} + V_{Bd}$$

$$V_{AB} = V_{D_1} + V_{D_2}$$

$$\because R_3 = R_4$$

$$\therefore V_{R_4} = V_{Bd'} = \frac{V_{AB}}{2}$$

$$\therefore V_O = \frac{V_{D_1} + V_{D_2}}{2} - V_{D_2}$$

$$\therefore V_O = \frac{V_{D_1} - V_{D_2}}{2}$$

$$\therefore \boxed{V_O \propto (V_{D_1} - V_{D_2})}$$

- Thus, output is directly proportional to difference of V_{D_1} and V_{D_2}.

Advantages

- It does not responds to noise i.e. amplitude variations in FM signal.

How Ratio Detector Acts as Amplitude Limiter ?

The capacitor C_3 is connected across the points A and B to limit the amplitude so that amplitude variations do not affect the output of the ratio detector. The capacitor C_3 keeps the sum of the output voltages of the diodes, V_{Ad} and V_{Bd}, to a constant value at all incoming frequencies. If there is a change in amplitude in the FM signal due to noise, C_3 acts accordingly and nullifies its effect.

To understand the amplitude-limiting effect of a ratio detector, it is necessary to understand the following :

- The sum of V_{Ad} and V_{Bd} is constant and is independent of the incoming frequency.
- The action of the capacitor changes when the amplitude of V_{in} changes.

Sum of V_{Ad} and V_{Bd}

To prove that the sum of V_{Ad} and V_{Bd} is always constant and independent of the incoming frequency, consider the three possible cases of incoming frequency, assuming V_{in} is constant.

Case (i) : $f_i = f_c$, V_{in} = Constant

It has already been explained that under this condition, the output voltages of the two diodes are equal, as shown in *figure 6.22(a)*. If the output of each diode is V volts, we have :

$$V_{AB} = V_{Ad} + V_{Bd} = V + V$$

$$V_{AB} = 2V$$

Case (ii) : $f_i > f_c$, V_{in} = Constant

It is observed from *figure 6.22(b)* that under this condition V_{D_1} is greater than V_{D2}. When f_i is equal to f_c, the two voltages are equal to V, as considered in case (i). However, V_{D_1} increases from V volts, and V_{D_2} decreases from its value of V volts in equal amounts. If the increase in V_{D_1} is ΔV volts, the voltage V_{D_1} will decrease from V volts by the same amount, ΔV.

The two voltages for this case are given as:

$$V_{D_1} = V + \Delta V$$

$$V_{D_2} = V - \Delta V$$

The sum of the two voltages is obtained as:

$$V_{AB} = V + \Delta V + V - \Delta V$$

$$V_{AB} = 2V$$

Case (iii) : $f_i < f_c$, V_{in} = Constant

Under this condition, V_{D_1} is less than V_{D_2}, as shown in *figure 6.22(c)*. With the same reasoning as in case (ii), the two voltages can be obtained as:

$$V_{D_1} = V - \Delta V$$

$$V_{D_2} = V + \Delta V$$

So that,

$$V_{AB} = V - \Delta V + V + \Delta V$$

$$V_{AB} = 2V$$

Conclusion

The sum of the output voltages of the diodes, which appears as V_{AB} in a ratio detector, is always constant at all frequencies, as obtained by equations (1), (2), and (3). Therefore, the voltage across the capacitor C_3 is independent of incoming frequency.

Action of C_3 to Limit Amplitude

Consider a situation in which the noise voltage alters the amplitude of the input FM signal, V_{in}. The capacitor comes into action when there is a change in V_{in}. The action of the capacitor can be explained by considering the three possible conditions of V_{in} :

- Case (i) : V_{in} remains constant
- Case (ii) : V_{in} increases
- Case (iii) : V_{in} reduces

Case (i) : V_{in} Remains Constant

This case is explained in the preceding section. It is proved that if V_{in} is constant, the voltage across A and B remains constant at all incoming frequencies.

If voltage V_{AB} remains constant, the capacitor charges to V_{AB}. Since it is a d.c. voltage, being constant at all times, it remains constant across the capacitor C_3. Beyond this point, there is no charging or discharging until V_{in} remains constant. $C_3 =$

This situation indicates that the capacitor C_3 offers infinite impedance. Therefore, the total load of D_1 and D_2 is only due to R_1 and R_2 because C_5 acts as an open circuit, and R_3 and R_4 are very large as compared to R_1 and R_2. Therefore, when V_{in} is constant, capacitor C_3 does not take any action and is therefore constant.

Case (ii) : V_{in} Increases

When there is an increase in the amplitude of V_{in}, the secondary voltages V_{ac} and V_{bc} increase accordingly. This increases biasing of the diodes, and the output voltages of D_1 and D_2 increase in the same proportion. As a result, the total voltage, V_{AB}, across the points A and B increases. Due to the increase in V_{AB}, the capacitor C_3 starts charging through the excess current supplied by the two diodes. This increases the capacitor voltage in the same proportion as the increases in V_{in}. Therefore, under this condition, the voltages across R_2 (V_{BD}) and R_4 (V_{BD}) increases equally. The output voltage V_o is the difference of $V_{Rd'}$ and V_{Rd}. The increase in these two voltages cancel each other at the output terminals. As a result, the rise in V_{in} does not alter the modulating signal.

Case (iii) : V_{in} Decreases

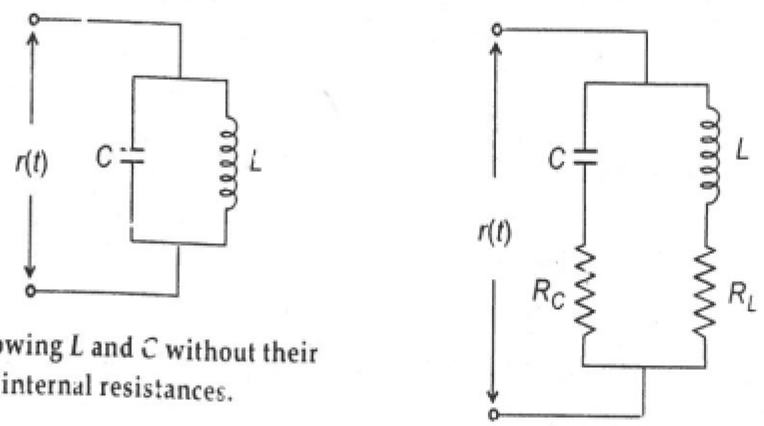
When there is a decrease in the input FM voltage, V_{in}, the input diode voltages V_{D1} and V_{D2} decrease accordingly, and the diode currents are reduced. The voltage V_{AB} is also reduced, but this change does not happen instantaneously because of the presence of C_3, implying that even when there is a decrease in the diode currents, the voltage across A and B remains constant This situation indicates that the load impedance of the diodes has increased.

The output voltage will not be affected because V_{AB} remains constant even when V_{in} decreases. Therefore, the decrease in V_{in} is not reflected at the output of the ratio detector. The ratio detector takes care of any unwanted change in the amplitude of V_{in} due to the timely action of the capacitor C_3. The changes in V_{in} are not reflected at the output, and no additional amplitude limiter is required with a ratio detector.

6.3 Frequently Asked Questions

Q.1. How does a tank circuit work ?

Ans.



(a) Showing L and C without their internal resistances.

(b) Equivalent circuit of (a) showing R_C and R_L as the internal resistances of capacitor and inductor respectively.

Fig. 6 23 : Parallel resonating circuit.

Circuit Description

- The *figure 6.23(a)* shows an ideal tank circuit.
- But as with every inductor or capacitor there is a resistance associated with it. So, *figure 6.23(b)* shows the equivalent circuit from practical point of view.

Working

- The resonating frequency of the above circuit is

$$f_0 = \frac{1}{2\pi\sqrt{LC}}, \quad f_0 = \text{Resonating frequency}$$

> **Note :** *Resonance is a condition where the circuit is purely resistive i.e. Reactance due to L and C cancel out each other i.e.* $X_L = X_C$.

- At any point the output voltage is always inversely proportional to the total reactance of the circuit (by Ohms law).

- The total reactance is the effective reactance due to L and C, thus they are given by

$$\left.\begin{aligned} X_L &= 2\pi f L \\ X_C &= \frac{1}{2\pi f C} \end{aligned}\right\} \quad \begin{aligned} &\text{and total reactance } = |X_L - X_C| \\ &\text{where } f = \text{Incoming frequency} \end{aligned}$$

- Now we will analyze three cases

Case (i) : When $f = f_0$ i.e. $X_L = X_C$

Now, the circuit will resonate and give maximum output,

$$\boxed{\text{Output} = V_{max}}$$

Case (ii) : When $f < f_0$ i.e. $X_L < X_C$

Now, the reactance due to capacitor is more than inductive reactance and thus there is some overall reactance and hence output voltage decreases,

$$\boxed{\text{Output} < V_{max}} \quad \{\text{decreases}\}$$

Case (iii) : When $f > f_0$ i.e. $X_L > X_C$

Again, there is some overall reactance and hence output voltage decreases,

$$\boxed{\text{Output} < V_{max}} \quad \{\text{decreases}\}$$

- The above three cases are graphically expressed as follows :

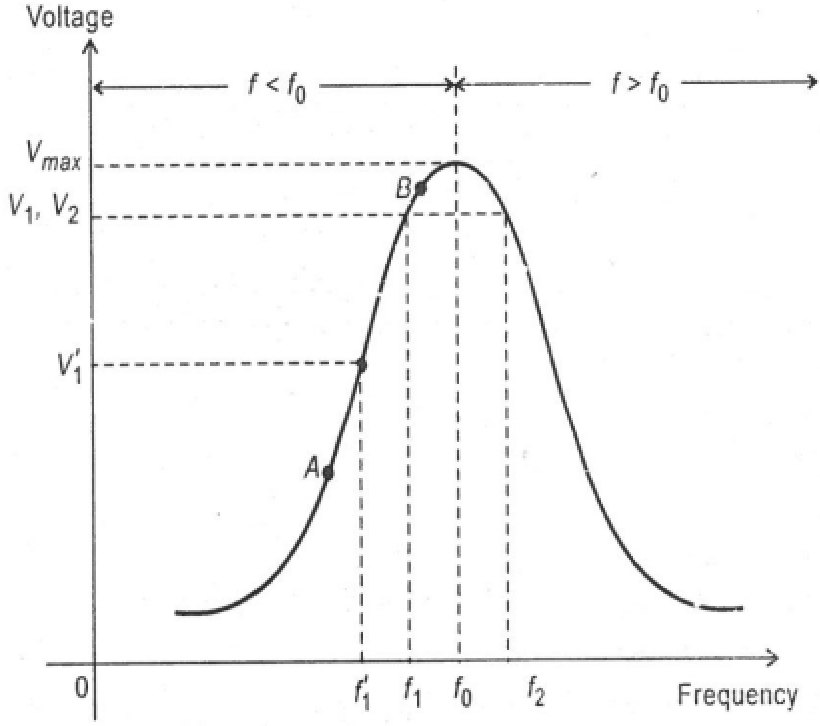


Fig. 6.24 : Voltage versus frequency curve of an LC resonating circuit.

- Thus, we can conclude that the output voltage depends on the frequency of the signal.
- Thus, if the input frequency is changed the output voltage will also change accordingly.
- Thus, using a tank circuit we can convert frequency variations to voltage or Amplitude variations.
- Hence, tank circuit acts as frequency to voltage converter.

> **Note :** *The tank/tuned circuit does not alters the frequency of the incoming signal at its output but varies its output voltage according to the frequency of the input signal.*

Q.2. Why is resonating frequency $f_0 = \frac{1}{2\pi\sqrt{LC}}$

Ans. We know,

$$X_L = 2\pi fL$$

$$X_C = \frac{1}{2\pi fC}$$

At resonance, $f = f_0$ and $X_L = X_C$

Now, $X_L = X_C$

$$\Rightarrow 2\pi f_0 L = \frac{1}{2\pi f_0 C}$$

$$\therefore \boxed{f_0 = \frac{1}{2\pi\sqrt{LC}}}$$

Q.3. Consider the following block diagram.

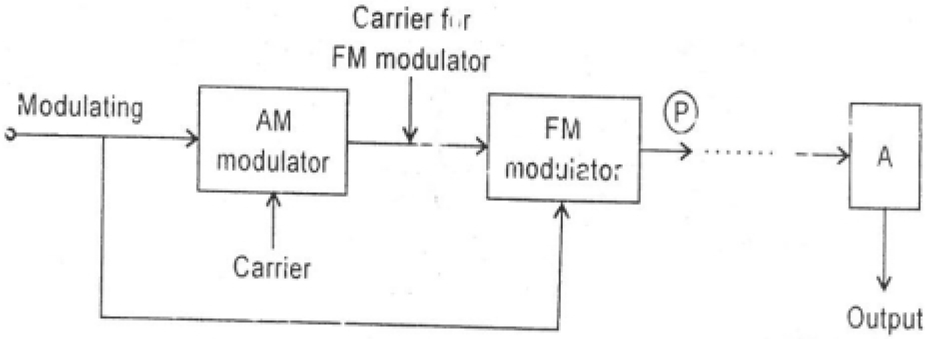


Fig. 6.25

Which circuit should be used in block A to get the original signal.

Ans.

- The first block gives output as AM wave. Thus, information is reflected in the amplitude of the carrier.

- Now, the same signal acts as a carrier for the second block i.e. FM modulator. Thus, original signal or the information is also reflected in the frequency of the same signal.
- Thus, the signal at point P is AM + FM and has the same information in its amplitude variations as well as frequency variations.
- Thus, even an AM detector can be used to detect the original signal or an FM detector can also be used.
- But, Foster-Seeley should not be used because it responds to the amplitude variations also.
- Also, if FM detector is used then, amplitude limiter should also be used.

Q.4. Why f_0 is always selected as $f_0 > f_c$?

Ans. If f_c is selected greater than f_0, then output of the tank circuit will be as shown in *figure 6.26*.

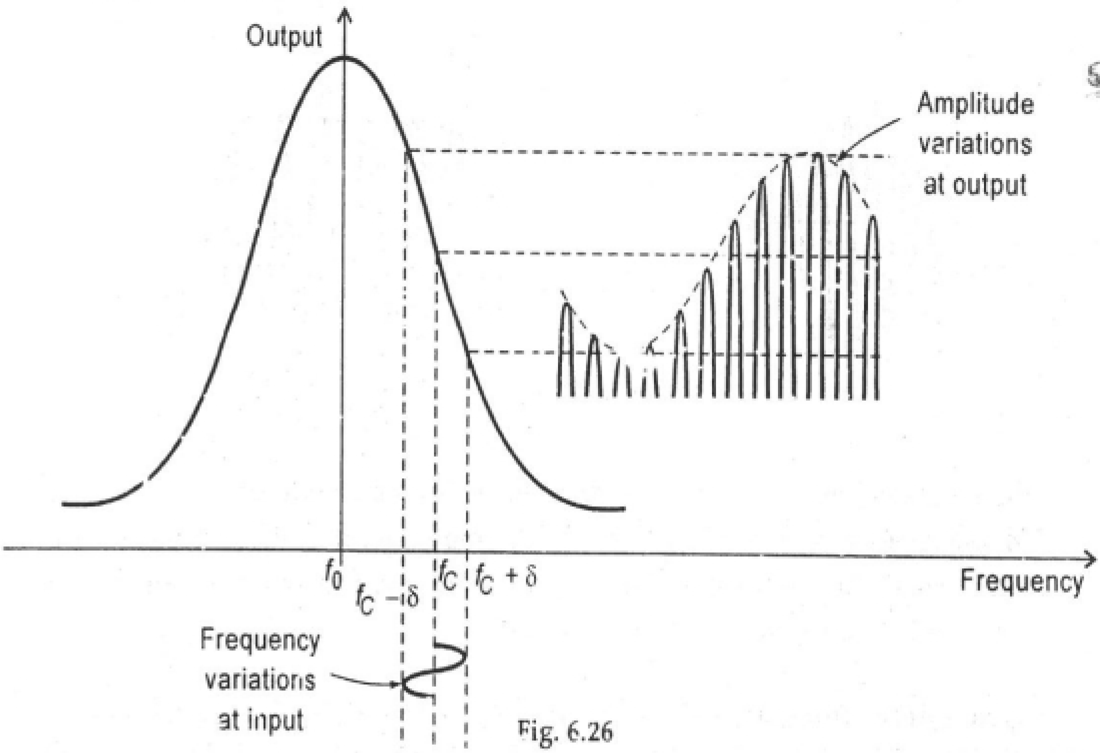


Fig. 6.26

Thus we can see that if f_c is greater than f_0, the output is inverted. Hence f_c is always less than f_0 or $f_0 > f_c$, always.

Q.5. Explain why FM is more immune to noise than AM.

Ans. Consider the modulating signal shown in *figure 6.27*.

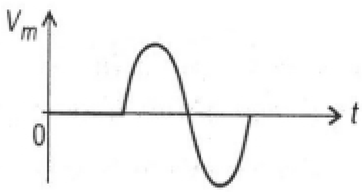


Fig. 6.27

Suppose it modulates the carrier using both amplitude modulation and frequency modulation, we get

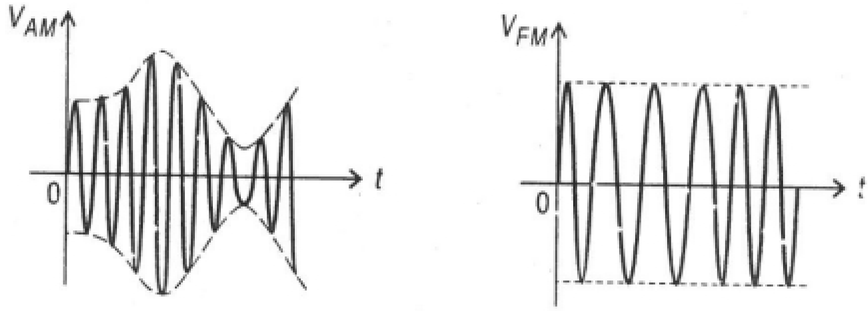


Fig. 6.28

Now when these signals are corrupted by noise the corresponding corrupted signals are as shown in *figure 6.29*.

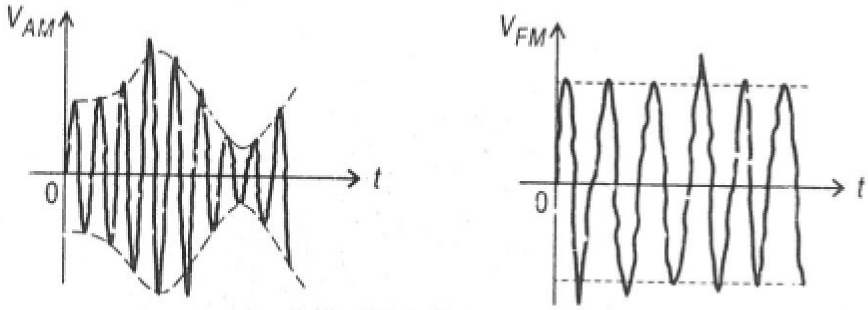


Fig. 6.29

This happens because noise mostly changes the amplitude of the signal. Since AM wave carries entire information in its amplitude, it is affected by noise to a larger extent. And since FM does not carry any information in its amplitude, it is more immune to noise compared to AM.

Q.6. Explain distortions in diode detector.

Ans. Refer section 6.1.2.

Q.7. Explain the diagonal clipping problem in diode detector.

Ans. Refer section 6.1.2.

Q.8. Draw and explain the working of a simple diode detector. Draw input and output waveforms if an over modulated waveform is given as input.

Ans. Refer section 6.1.1.

For an **overmodulated wave, input waveform :**

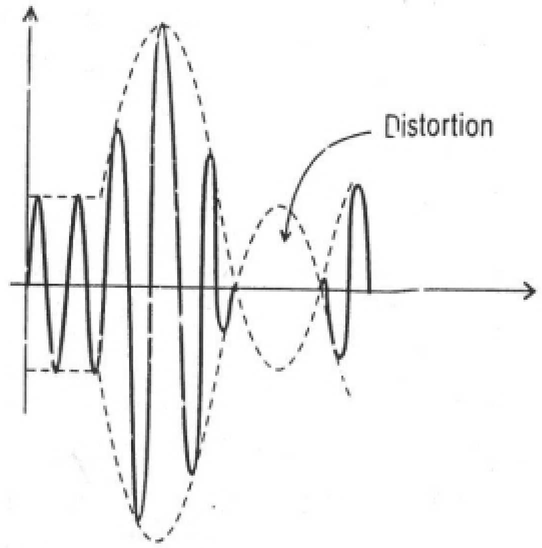


Fig. 6.30

Output waveform :

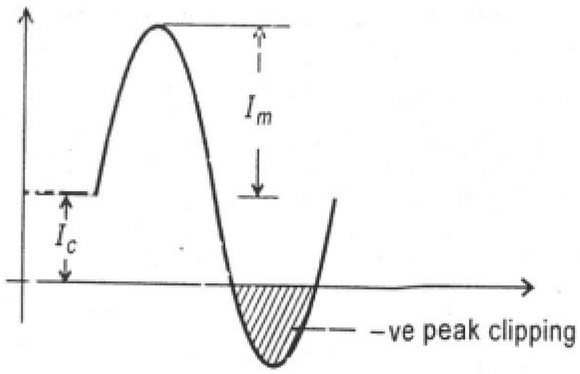


Fig. 6.7

Q.9. Explain - Ratio detectors do not need Amplitude limiters. Why ?

Ans. Write about 'Action of C_3 to Limit Amplitude' from section 6.2.2.2.

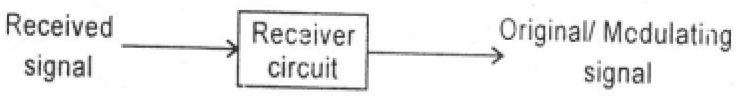
✲✲✲

7 RADIO RECEIVERS

Topic	Theory imp	Oral imp
Introduction		
TRF	☆☆	☆☆
SHR	☆☆☆☆	☆☆☆☆
Receiver Characteristics	☆☆☆☆	☆☆☆☆
Tracking	☆☆	☆☆☆
AGC	☆☆☆	☆☆☆☆
Communication Receiver	☆☆	☆
FAQ's	☆☆	☆☆☆☆
Problems	☆☆☆☆	☆

7.0 Introduction

The main function of the receiver is to regenerate the original signal from the received signal.



According to our syllabus, we have to study two receiver circuits :

(i) TRF Receiver

(ii) SHR Receiver

7.1 Tuned Radio Frequency (TRF) Receiver

Q. *Sketch the block diagram of a TRF radio receiver and briefly describe its working. Explain its predominant disadvantages.*

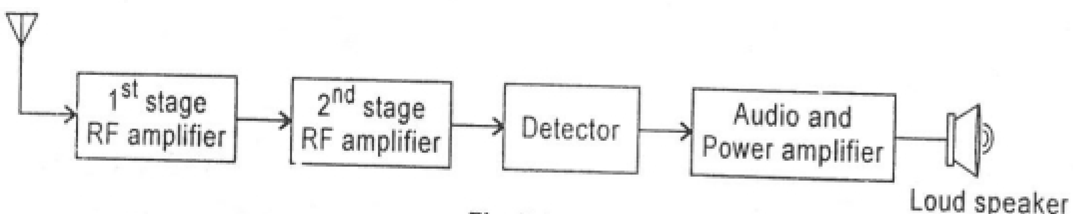


Fig. 7.1

Working

Figure 7.1 shows the block diagram of TRF receiver. The functions of different blocks are

(i) RF Amplifier

- Two or three RF amplifiers are used to select the desired frequency signal and reject all other frequencies.
- Also after selecting the signal, the signal is amplified.

(ii) Detector

- It consists of the detector circuit.
- Original signal is detected here.

(iii) Audio and Power Amplifier

- Used to amplify the detected signal.

Advantages

- Simple in design.

Disadvantages

(i) Instability

- The overall gain of RF amplifiers is extremely high.
- So, a small feedback from the output can make the RF amplifier work as an oscillator.

(ii) Variation in Bandwidth

- For better selectivity, the bandwidth of the receiver should always remain constant. But, in TRF bandwidth changes with the incoming frequency.

> **Note :** *Selectivity is a characteristic of a receiver covered in the later part of this chapter.*

- Normally, bandwidth, $B.W. = \dfrac{f_r}{Q}$

$$f_r = \text{Frequency of received signal}$$

$$B.W = \text{Bandwidth}$$

$$Q = \text{Quality factor of the circuit}$$

Now, Q is constant.

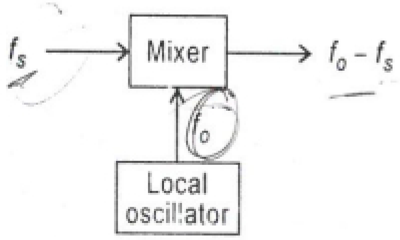
Thus, when the frequency of received signal changes, the bandwidth also changes. This drawback is overcome in SHR with the concept of intermediate frequency.

(iii) Poor Selectivity

- As bandwidth varies with the variation in incoming frequency the selectivity degrades.

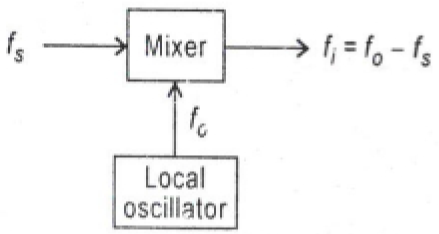
7.2 Concept of Intermediate Frequency in SHR Receiver

- The main drawback of TRF receiver is its variation in BW due to variation in frequency.
- SHR overcomes this drawback by converting every received signal frequency to a constant intermediate frequency called the *IF frequency* or (f_i).
- This is done with the help of a local oscillator and a mixer.
- *Local oscillator* is a circuit which is used to generate signals of a particular frequency.
- *Mixer*



Mixer just gives the difference of the two input frequencies.

7.2.1 How Does Local Oscillator and Mixer Work



f_s = Incoming frequency

f_o = Oscillator frequency

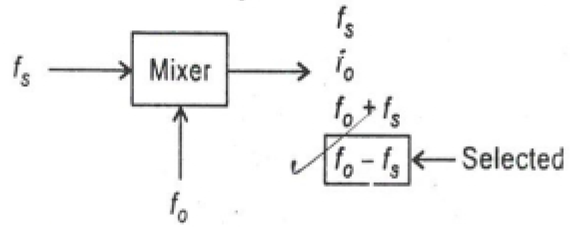
f_i = Intermediate frequency

- Our main aim is to keep the output i.e. f_i at a constant value.

Working

- As f_s changes, if f_o is also changed then, the difference will be constant.
- Hence, f_o is always changed with f_s using ganged tuning.

Note : *The mixer circuit in practice gives many frequencies at its output but only the difference frequency is selected for the next stages.*



7.3 Super Heterodyne Receiver (SHR Receiver)

Q.1. Explain the working of a superheterodyne receiver.

Q.2. Draw the block diagram of super heterodyne radio receiver with waveforms at each stage and explain its working.

Block Diagram

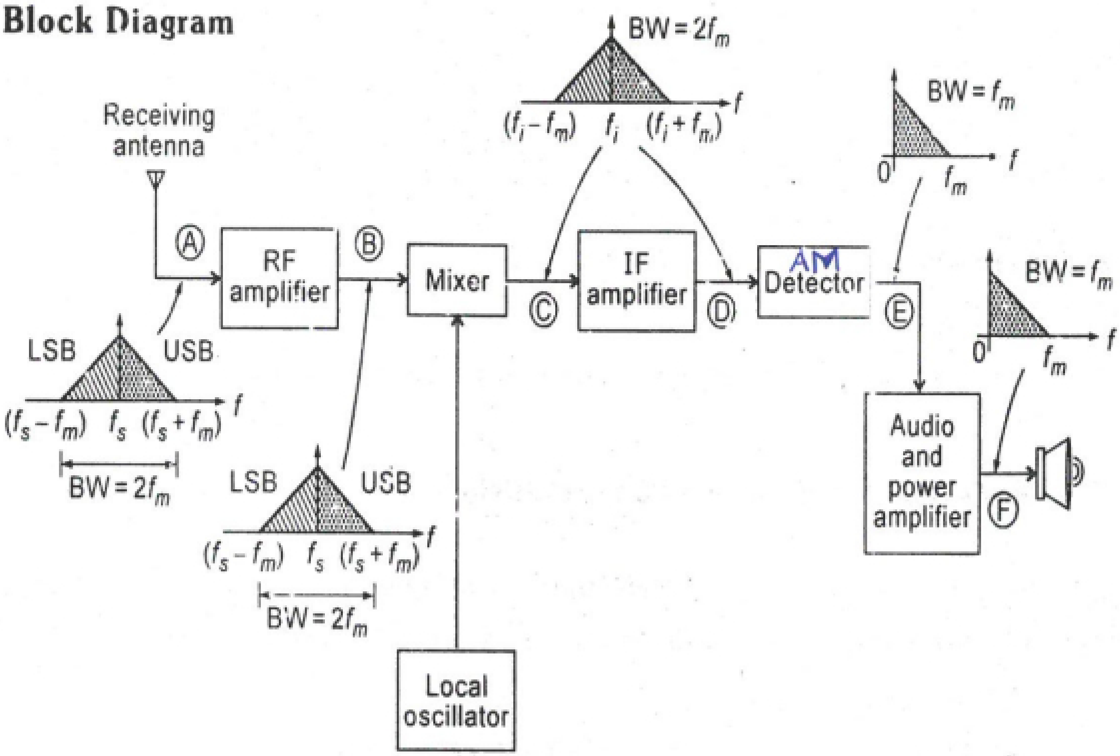


Fig. 7.2 : Super heterodyne receiver with centre frequencies and bandwidth of each block.

Working

(i) RF Amplifier

The functions of RF amplifier are

- Select the desired frequency and reject all other frequencies.
- Amplify the selected signal.

(ii) Mixer

The function of mixer is

- Convert the incoming frequency to a constant intermediate frequency (f_i).

(iii) IF Amplifier

- This amplifier is tuned to the IF frequency (f_i).
- It amplifies the signal given by the mixer.

(iv) Detector

- It consists of the detector circuit.
- The original i.e. modulating signal is detected here.

(v) Audio and Power Amplifiers

- Used to amplify the detected signal.

Advantages

- Bandwidth of the receiver does not change with the variation in incoming frequency.
- Sensitivity and selectivity is more.
- Gain of the amplifier is constant as IF is constant.

Disadvantages

- Cost is more.
- Design is little complicated as compared to TRF receiver.

7.4 Receiver Parameters/Characteristics

Q.1. Define : (i) Sensitivity (ii) Selectivity (iii) Fidelity.

Q.2. Explain the following with reference to Radio receivers :

(i) Selectivity	*(iii) Image frequency*
(ii) Fidelity	*(iv) Tracking error.*

Q.3. Explain the following with reference to Radio receivers :

(i) Selectivity	*(iii) Sensitivity*
(ii) Fidelity	*(iv) Double spotting.*

The performance of any receiver is measured on the basis of the following parameters.

(i) Sensitivity

- Sensitivity of a receiver is defined as its ability to amplify weak signal.
- Sensitivity is measured in micro volts (μv).
- If a receiver has sensitivity of 150 μv, then it implies that minimum 150 μv signal should be applied to the receiver to get a desired output.
- Sensitivity changes with frequency as shown in *figure 7.3*.

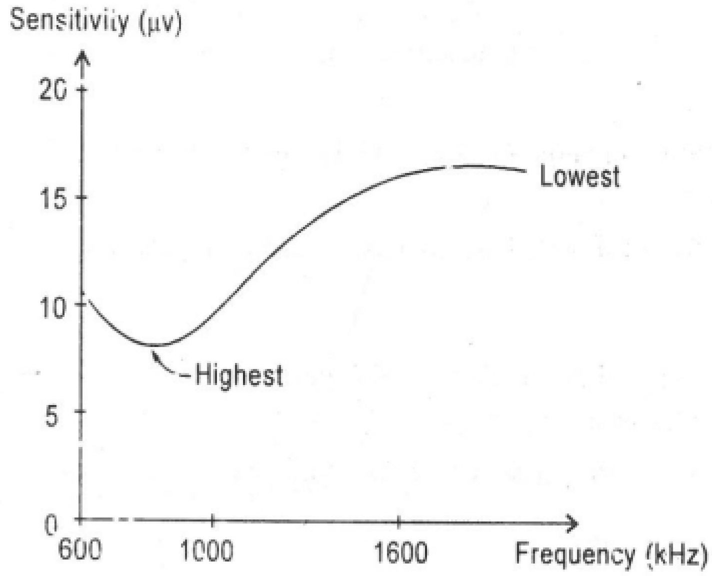


Fig. 7.3 : Sensitivity curve of an AM receiver.

- It depends on the gain of
  - RF Amplifier
  - IF Amplifier.

(ii) Selectivity

- It is defined as the ability of a receiver to reject unwanted signal. → to accept a given band of freq & reject all other
- If a receiver is tuned to 900 kHz then, the selectivity is said to be good if the receiver only selects signal of 900 kHz frequency and rejects other frequencies.
- Selectivity is expressed as a curve called the *selectivity curve*.

eg - Commercial AM broadcast band each station transmitter is allocated a 10 KHz BW

other way - to describe the selectivity of a radio receiver

-3dB pts

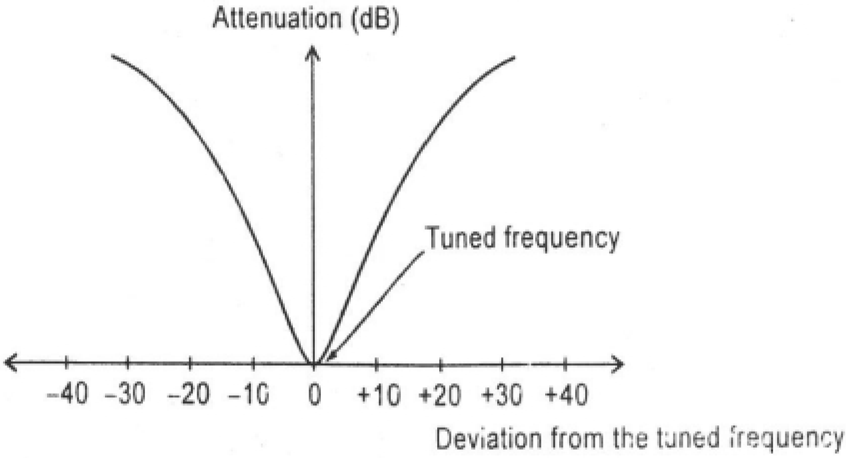


Fig. 7.4 : Selectivity curve of an AM receiver.

- Selectivity depends on the quality factor of the tuned circuit used in IF amplifier.
- Also if BW of IF amplifier increases, selectivity degrades.

(iii) Fidelity

- It represents the variation in the gain of the receiver with variation in the modulating frequency.
- A typical fidelity curve is as shown in *figure 7.5*.

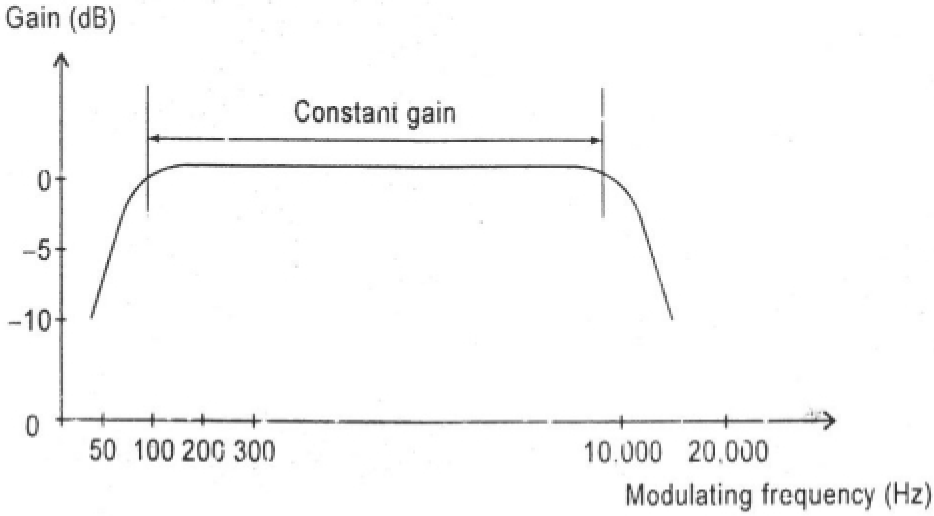


Fig. 7.5 : Typical fidelity curve of a receiver.

- The fidelity of any receiver depends on the audio amplifiers.

(iv) Image Frequency and its Rejection

Image Frequency

To understand image frequency, consider a receiver which is tuned to receive 1000 kHz frequency. Now, consider the following cases :

Case (i)

- The receiver is tuned to a frequency of 1000 kHz.

$$\therefore \; f_s = 1000 \text{ kHz}$$

- As the IF frequency is set to 455 kHz, the local oscillator frequency is

$$f_o = f_s + f_i$$

$$\boxed{f_o = 1455 \text{ kHz}}$$

- Now, the receiver will work for input frequency (f_s = 1000 kHz), since the difference of f_s and f_o is equal to IF frequency. This is the normal operation and nothing is wrong. Now consider the second case.

Case (ii)

- Now consider another case with same value of f_o and f_i in the above case but f_s is now 1910 kHz.

$$\therefore \; f_s = 1910 \text{ kHz}$$

- The difference of f_o and f_s is still equal to f_i i.e. 455 kHz

$$f_i = f_o - f_s = -455 \text{ kHz}$$
$$= 455 \text{ kHz} \qquad \left(\because \text{ We cannot have -ve frequency}\right)$$

- Since, the receiver will always work if the difference of f_s and f_o is f_i, the receiver will detect this signal also.
- The frequency of 1910 kHz in this case is called *image frequency* and it is given by

$$\boxed{f_{si} = f_s + 2f_i}$$

$f_{si} = f_s + 2f_i$

f_{si} = Image frequency

f_i = Intermediate frequency, normally f_i = 455 kHz

f_s = Signal frequency.

Image Frequency Rejection Ratio

- It is denoted by α
- It is the ratio of the gain of the receiver at signal frequency to the gain at image frequency.
- It is given by

$$\alpha = \frac{\text{Gain at signal frequency}}{\text{Gain at image frequency}}$$

- It depends on the quality factor Q in the following manner

$$\boxed{\alpha = \sqrt{1 + Q^2\rho^2}}$$

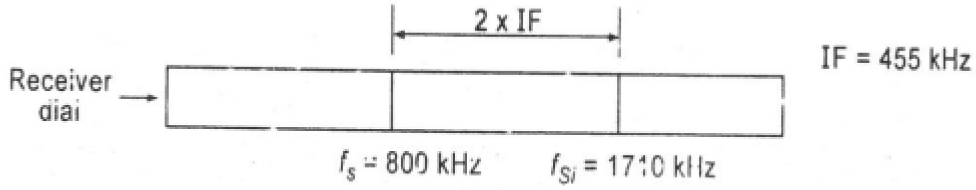
where $$\boxed{\rho = \frac{f_{Si}}{f_s} - \frac{f_s}{f_{Si}}}$$

Double Spotting

Note : *Double spotting is not a receiver characteristic, it is just a problem faced in some receivers due to poor characteristics.*

- Double spotting means the same station gets picked up at two different points on the receiver dial.

- It can occur if a signal is stronger than the signal at its image frequency.



- Consider the figure above

$$f_s = 800 \text{ kHz}$$

$$f_{Si} = 2 \times \text{IF} + 800 \text{ kHz} = 1710 \text{ kHz}$$

Assume that the signal with frequency 800 kHz is of very high strength compared to the signal with frequency 1710 kHz.

Consider the following cases

Case (i) : Receiver Dial is Tuned to 800 kHz

- As the strength is very high, the signal is picked up and nothing is wrong. Problem occurs in the next case.

Case (ii) : Receiver Dial is Tuned to 1710 kHz

- As the signal at 1710 kHz is weak and also, as 1710 kHz is the image frequency of 800 kHz, the receiver can detect both the signals.
- Now, as the strength of original signal is less than the image signal, the image signal is selected.
- Thus, the same signal is selected at two points. This is called *double spotting*.

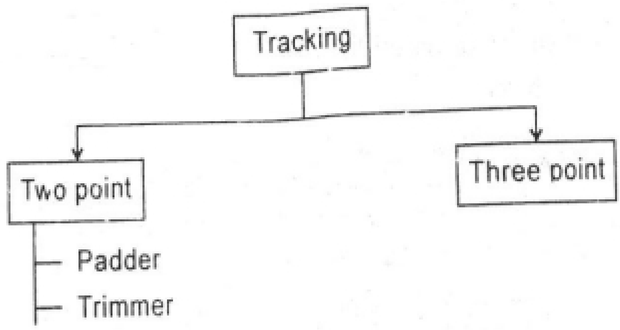
Blocking

- If a radio receiver is tuned to a weak signal, then the corresponding Automatic Gain Control (AGC) will be very low, and the gains of the RF and IF stages will be high. If a strong signal which is close in frequency to the weak signal is present then the AGC voltage may reduce due to its presence. This may suppress the wanted signal completely. Also if the strong signal is fluctuating then the AGC voltage will fluctuate.
- A receiver which has a very little reaction to the nearby unwanted signals is said to have good blocking. To have excellent blocking, high adjacent channel rejection should be there. For this the selectivity of the IF amplifier should be high.

7.5 Tracking

Q. *Explain local oscillator tracking and tracking error in radio receiver.*

- Tracking is a process in which the local oscillator frequency follows or tracks the incoming signal frequency to have a constant frequency difference (i.e. f_i).
- To keep the frequency difference constant, the capacitors of tuned circuit and local oscillator are ganged i.e. mechanically coupled or tuned simultaneously.
- Then too, in practice the frequency difference is never constant resulting in error. This is called *tracking error*.
- Tracking error can not be eliminated, it can just be suppressed by using two point and three point tracking.



Two Point Tracking

- Since tracking error cannot be eliminated, here we try to minimize the tracking error to zero at two different frequencies by connecting capacitors in series or parallel with capacitor of oscillator.
- Hence the name two point tracking.

- It is of two types
  - Padder tracking
  - Trimmer tracking

7.5.1 Padder Tracking

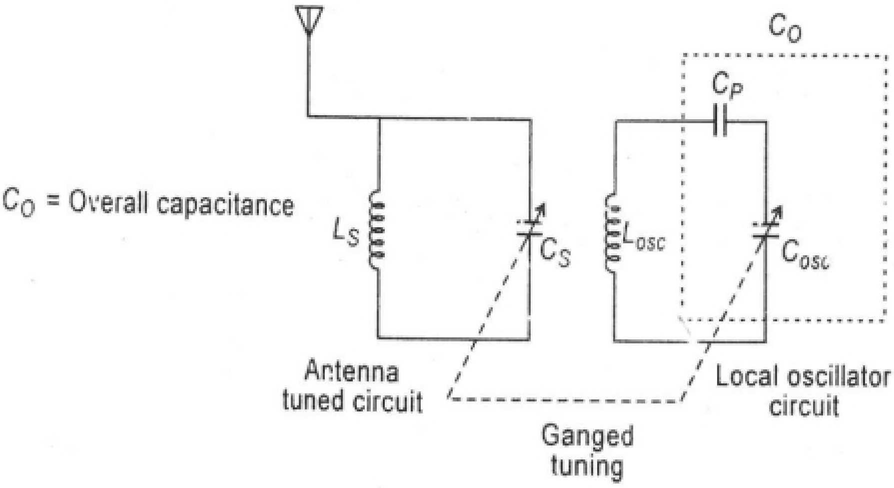


Fig. 7.6 : Padder tracking.

> **Note :** *If overall capacitance C_O is less than C_{osc} then, tracking error is +ve else –ve.*

- A padder capacitor C_P is connected in series with C_{osc}.
- Thus the overall capacitance is

$$C_O = \frac{C_P \; C_{osc}}{C_P + C_{osc}} \text{ and is less than } C_{osc}$$

- Thus, tracking error is +ve.
- The padder capacitor is adjusted to have two points on the frequency dial where the tracking error is zero.

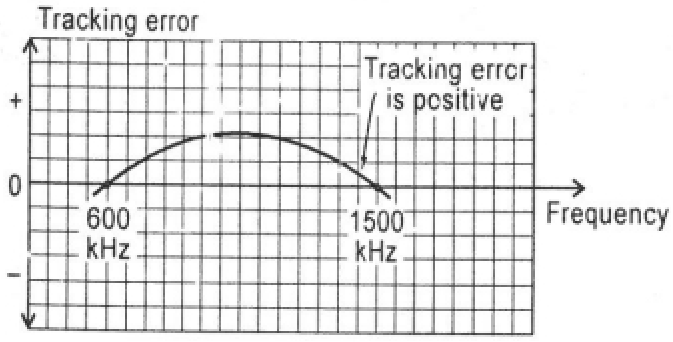


Fig. 7.7 : Error in padder tracking.

> **Note :** *The frequencies at which tracking error is zero depends on the value of capacitors. Here 600 kHz and 1500 kHz are taken as example.*

7.5.2 Trimmer Tracking

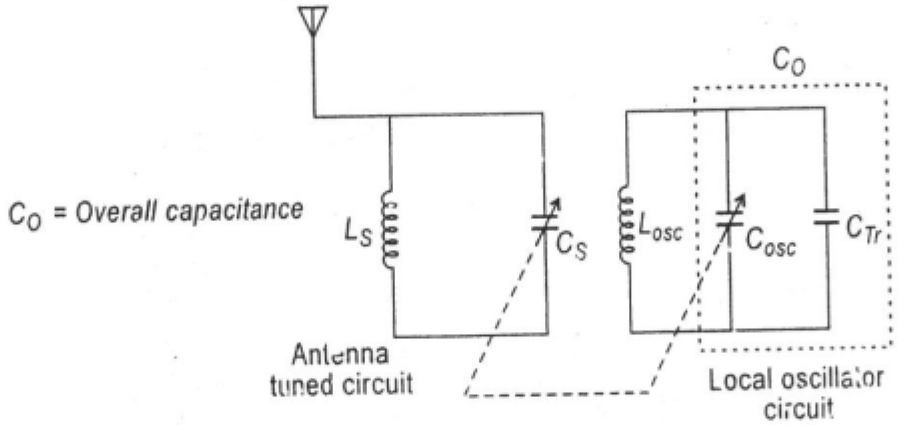


Fig. 7.8 : Trimmer tracking.

- A trimmer capacitor (C_{Tr}) is connected in parallel with C_{osc}.
- Thus, the overall capacitance is

$$C_O = C_{Tr} + C_{osc} \quad \text{and is greater than } C_{osc}$$

- Thus, the tracking error is –ve.
- The trimmer capacitor is adjusted to have two points on the frequency dial where the tracking error is zero.

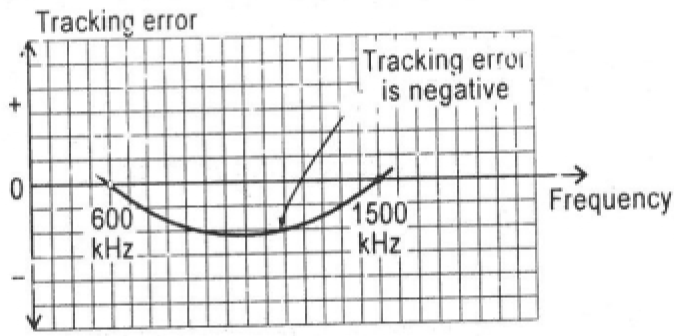


Fig. 7.9 : Error in trimmer tracking.

7.5.3 Three Point Tracking

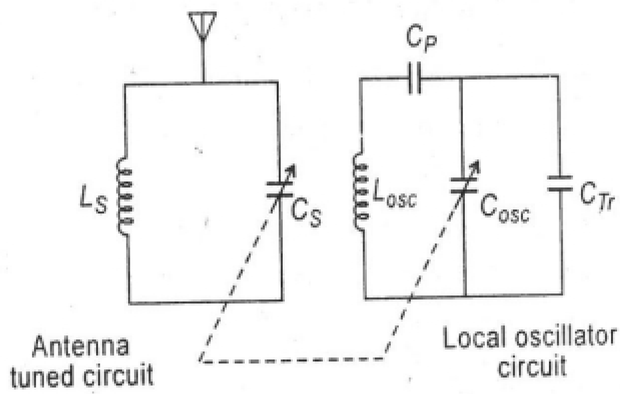


Fig. 7.10 : Three point tracking.

Q. Explain three point tracking.

Refer figure 7.10.

- This combines the padder and trimmer tracking.
- The padder (C_P) and trimmer (C_{Tr}) capacitors are adjusted to get three points on the frequency dial where the tracking error is zero.

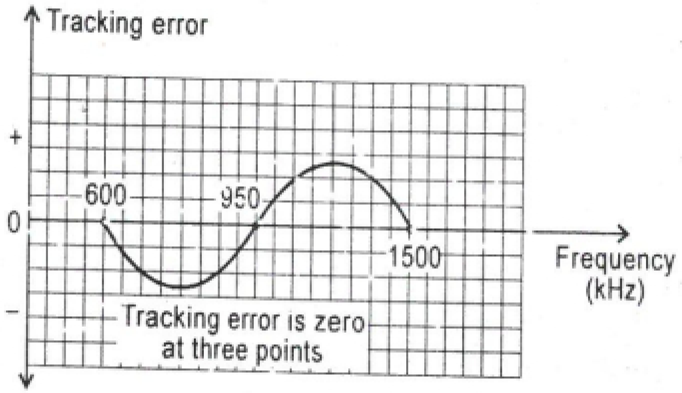


Fig. 7.11 : Three point tracking error .

7.6 Choice of IF Frequency

Q.1. What are the major factors influencing the choice of the intermediate frequency ?

Q.2. Explain selection of IF frequency.

There are three constraints on the selection of IF frequency :

(i) It should not be very high.

(ii) It should not be very low.

(iii) It should not fall within the range of the received signal.

(i) If IF is very high

- If IF is very high then, the BW of IF amplifier increases $\left(\because BW = \frac{f_i}{Q}\right)$ which will result in poor selectivity.
- Also, if IF is high the image frequency rejection ratio i.e. α is better.
- If IF is high then, tracking becomes difficult.

(ii) If IF is very low

- If IF is very low then, the BW of IF amplifier will decrease resulting in better or sharp selectivity.
- If IF is low, then α decreases.
- Tracking becomes easier.

(iii) If IF is selected between tuning range (540 - 1650 kHz)

- Then, there will be unwanted outputs at the loudspeaker due to interference.

Hence, IF is selected between 430 kHz to 460 kHz. Normally 455 kHz.

7.7 Automatic Gain Control (AGC)

Q. Write a short note on AGC.

Need

- Different signals of different frequencies are received by radio stations.
- Each signal has its own strength, some are stronger comparatively and some are weaker.
- The receiver's gain is constant. Thus receiver output varies with the strength of the incoming signal.
- This is undesirable. Hence AGC circuits were introduced to vary the gain of the receiver according to the input signal so that output is not fluctuating.

Definition

- AGC is a circuit by means of which the overall gain of the receiver is varied in accordance with the strength of the received signal.
- Normally, AGC voltage is a –ve voltage that is applied at the base of transistors of IF and RF amplifiers which reduces the overall gain when the signal strength is high.
- When signal is weak, ideally there should be no AGC voltage.

Types of AGC

There are two types :

- Simple AGC
- Delayed AGC

7.7.1 Simple AGC

Q. Discuss the merits of delayed AGC as compared to simple AGC. Sketch the circuit and explain how delayed AGC can be realized.

Note : *Write 7.7.2 Delayed AGC for complete answer.*

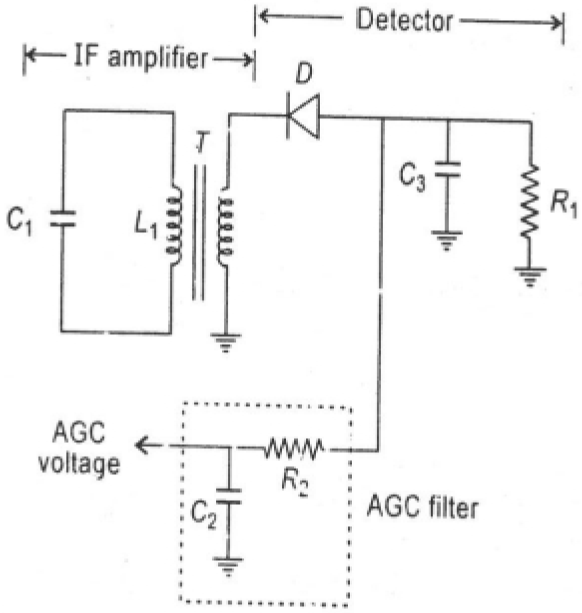


Fig. 7.12 : Circuit Diagram of a Simple AGC

- Diode D is connected such that only –ve peak of modulating signal is detected.
- The diode also produces a d.c. component and which is proportional to the strength of the received signal.
- The AGC filter passes this d.c. component and filters out the a.c. components.
- Thus a –ve AGC voltage is developed which is applied to the base of IF and RF amplifiers.
- Due to this, the gain of IF and RF amplifier reduces in turn reducing the overall gain.

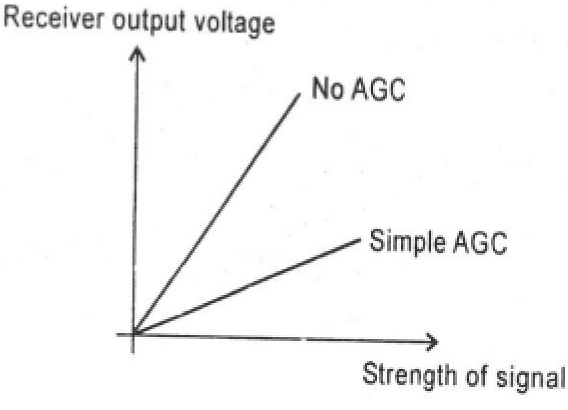


Fig. 7.13

Advantages

- Simple to design

Disadvantages

- Even if the signal is already weak, some amount of AGC voltage is developed reducing the gain of receiver.
- This may result in total loss of weak signals.

7.7.2 Delayed AGC

- Delayed AGC system gets activated only after the signal strength is more than a predetermined value.
- Thus it won't operate for weaker signals.

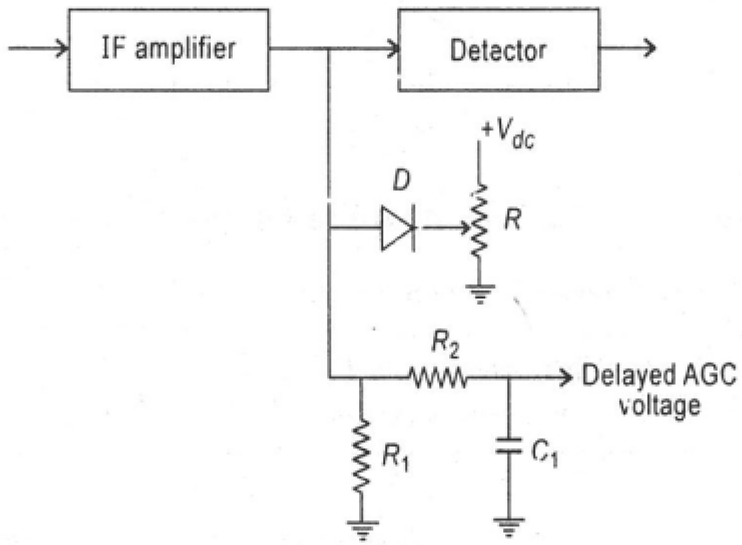


Fig. 7.14 : Circuit diagram of a delayed AGC.

Working

- The cathode of diode D is connected to $+V_{dc}$.
- Now we will consider the following two cases.

Case (i) : Signal is Weak

- If the signal is weak, i.e. less compared to V_{dc} then the diode D is reverse biased.
- Thus all the a.c. signal is by passed by capacitor C_1 to ground resulting in No AGC voltage.
- Thus gain is not affected.

Case (ii) : Signal is Strong

- If the signal is strong, i.e. more compared to V_{dc} then the diode D is forward biased during +ve cycle and reverse biased during –ve cycle.
- Now, diode D produces a d.c. component which varies according to the strength of the signal.
- The capacitor acts as an open circuit for this d.c. voltage.
- Thus a –ve voltage is developed as delayed AGC voltage.
- This is then connected to the IF and RF amplifiers to reduce the gain.

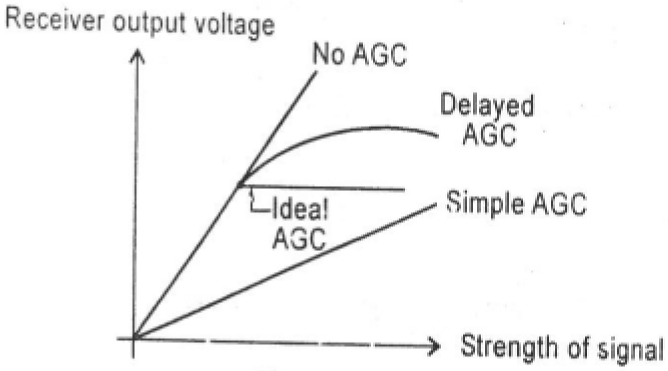


Fig. 7.15

7.8 Communication Receiver (Double Conversion Receiver)

Q. What are Double Communication Receiver.

> **Note :** *just remember the block diagram and the functions of new blocks. Else every thing is simple (For 5 marks don't explain working)*

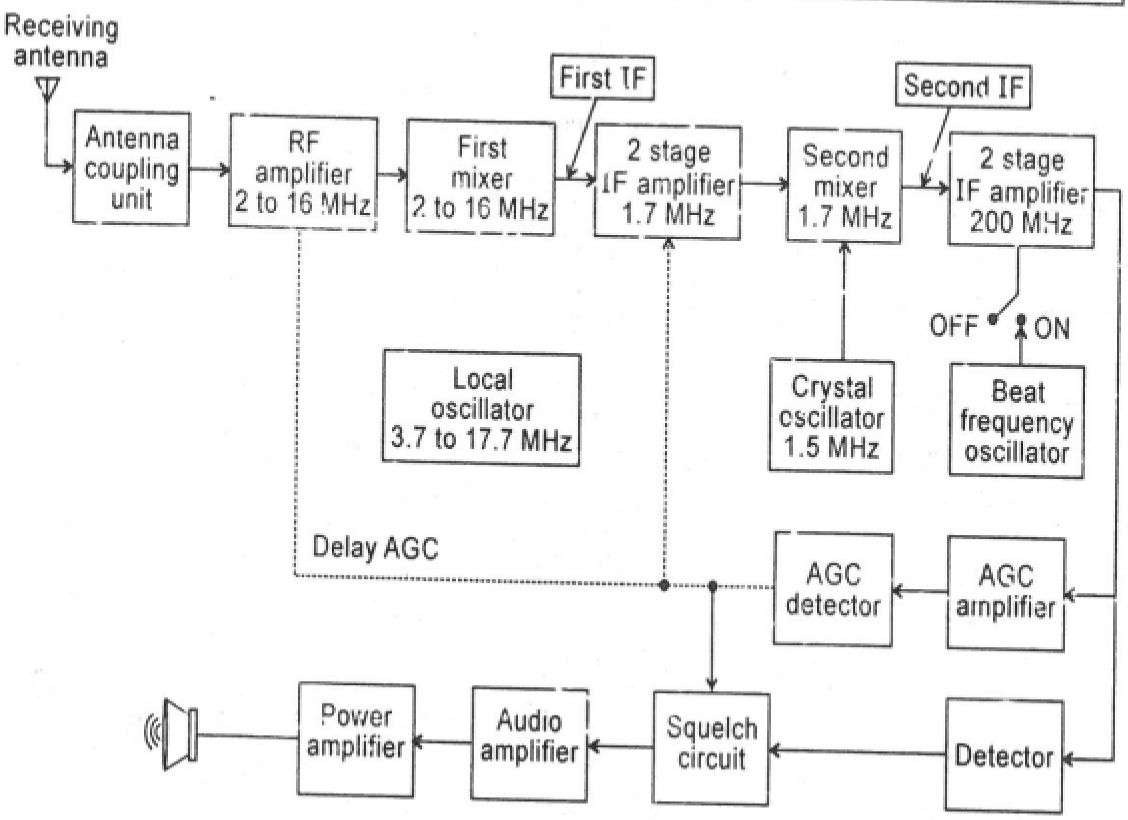


Fig. 7.16 : Block diagram of double conversion receiver.

- The receiver works on the principle of SHR receiver.
- It has two mixers, two local oscillators and two intermediate frequencies. Hence, it is called *double conversion receiver*.
- The first IF is used to reject the image frequency and the second IF is for better selectivity and tracking.

- Delayed AGC is used.
- Antenna coupling unit is used to receive maximum transfer.

Working of Additional Blocks

(i) BFO (Beat Frequency Oscillator)

- It is used to receive Morse Codes.
- Morse Codes were used earlier in telegraphy.
- In morse codes the information was in the form of dots and dashes.
- A switch is provided to enable this block.

(ii) Squelch or Muting Circuit

Need

- When there is no input then AGC output is zero and only noise is received at input terminal.
- As AGC value is zero, gain of circuit is high.
- Thus noise is amplified and there is a loud irritating sound at the loudspeaker.
- Thus a muting circuit is added so that if there is no input signal, there should be no output at loud speaker.

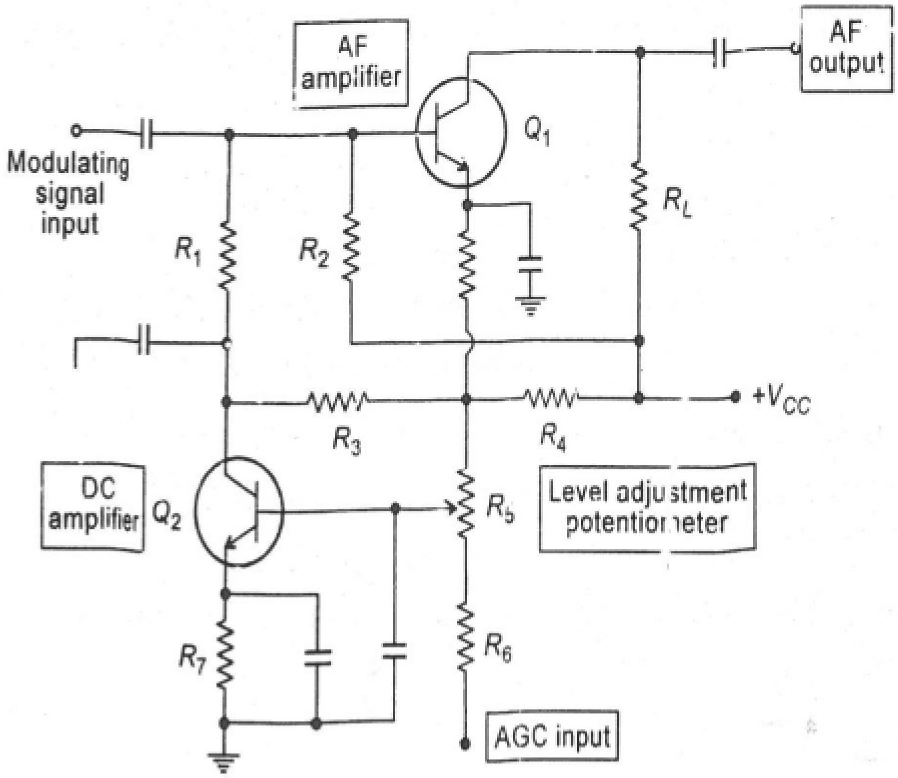


Fig. 7.17 : Squelch circuit.

Working

- When there is no input, the base of Q_1 is zero thus Q_1 acts as open circuit and hence signal is not passed to the next stage.

- Also the entire signal is bypassed through Q_2 as base-emitter junction of Q_2 is forward biased. Thus, resulting in no output at the loudspeaker.

Advantages of Double Conversion Receiver

- It provides two stage IF amplifiers because of which we have better selectivity and sensitivity compared to other receivers.
- It provides higher image frequency rejection.
- It can also receive morse code with the help of BFO.
- Squelch circuit is also used, thus if no input is applied noise does not reaches the output.

Disadvantages

- Cost is more.
- Circuit is very complex.

7.9 Frequently Asked Questions

Q.1. Why f_o should be greater than f_s ?

Ans. In local oscillators or any other circuit, the capacitance ratio of a tunable capacitor is the ratio of maximum capacitance to the minimum capacitance and it is given by

$$\boxed{\frac{C_{Omax}}{C_{Omin}} = \left[\frac{f_{Omax}}{f_{Omin}}\right]^2}$$

where f_{omax} = Maximum oscillator frequency

f_{omin} = Minimum oscillator frequency

Practically the capacitance ratio should always be less than 10.

Now, consider the following analysis.

Let f_i = 455 kHz and f_{smin} = 500 kHz and f_{smin} = 1000 kHz

Now, for the difference of f_o and f_s to be equal to IF we can have two cases.

(i) $f_o > f_s$

(ii) $f_o < f_s$

Case (i) : $f_o > f_s$

Then $f_{omin} = f_{smin} + f_i$ and $f_{omax} = f_{smax} + f_i$

$\therefore$ f_{omin} = 955 kHz and f_{omax} = 1455 kHz

$$\text{then} \quad \boxed{\frac{C_{Omax}}{C_{Omin}} = (1.52)^2 = 2.32 < 10}$$

Case (ii) : $f_o < f_s$

$$\text{Then} \quad f_{omin} = f_{smin} - f_i \quad \text{and} \quad f_{omax} = f_{smax} - f_i$$

$$\therefore \quad f_{omin} = 45 \text{ kHz} \quad \text{and} \quad f_{omax} = 545 \text{ kHz}$$

$$\text{then} \quad \boxed{\frac{C_{Omax}}{C_{Omin}} = (12.11)^2 \approx 146.56 >> 10}$$

This is practically impossible, thus to have capacitance ratio within practical limits $f_o > f_s$.

Q.2. What is Capture Effect ?

Ans.
- It is observed in receivers which travel from one place to another e.g. Mobile phone.
- A mobile phone receiver can receive signal from many transmitters.
- It accepts the signal which is stronger than every other signal.
- Now if a mobile receiver receives two signals at a time from two transmitters and the signal from each transmitters becomes stronger alternately, then the receiver will accept the signals alternately.
- Thus, the receiver is captured by different transmitters alternately.
- This is called *Capture Effect.*
- Normally seen in FM receivers.

Q.3. What is AFC ? Give block diagram.

Ans.
- Normally, in Radio Receivers, the local oscillator frequency is not stable i.e. if the local oscillator should produce a frequency f_o, then due to some external factors like temperature, etc. it produces a frequency between $f_o \pm \Delta f$, where Δf is any small value.
- AFC is a technique by which the local oscillator frequency is kept stable in different receivers.
- Following circuit shown in *figure 7.18* is used to implement this technique.

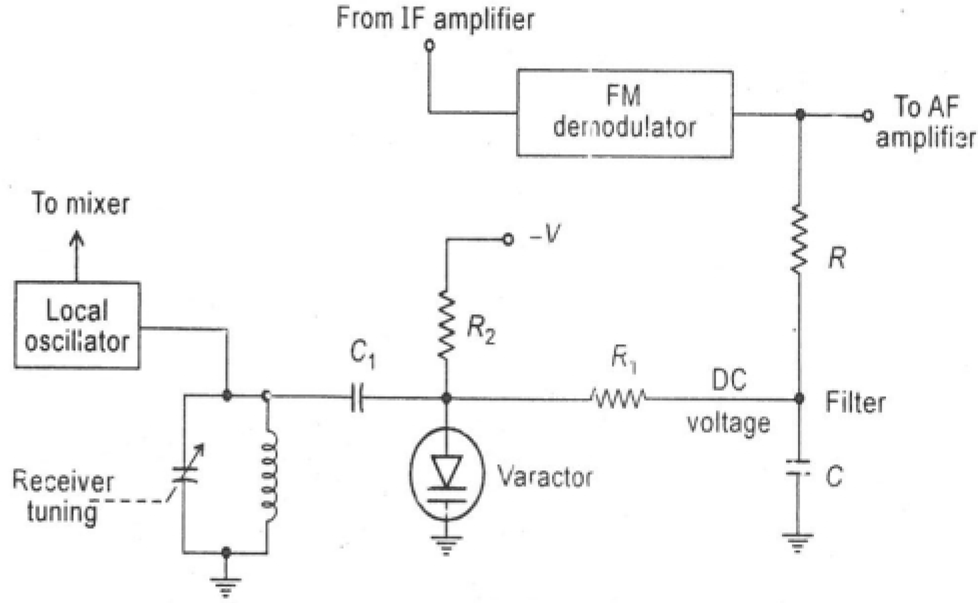


Fig. 7.18 : Automatic Frequency Control (AFC).

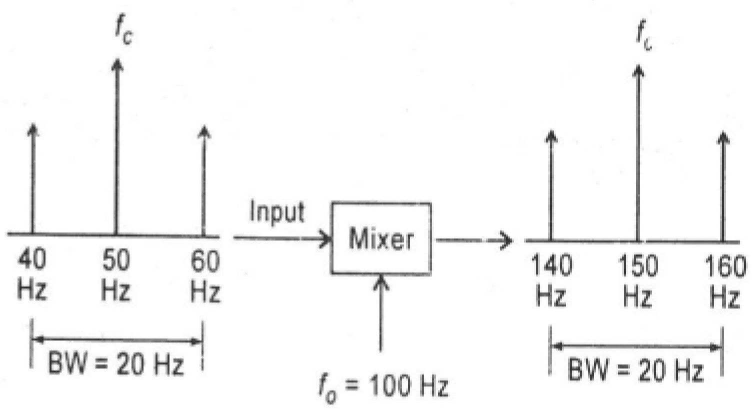
Working

- Output of demodulator is filtered to get a d.c. voltage.
- This d.c. voltage varies with the drift in frequency of local oscillator.
- This d.c. voltage varies the overall capacitance of the local oscillator with the help of varactor diode.
- Thus if the local oscillator frequency drifts away from the required value then it gets back to the original value automatically.

Q.4. ***What is the difference between Mixer and Multiplier ?***

Ans. Consider the following block diagrams

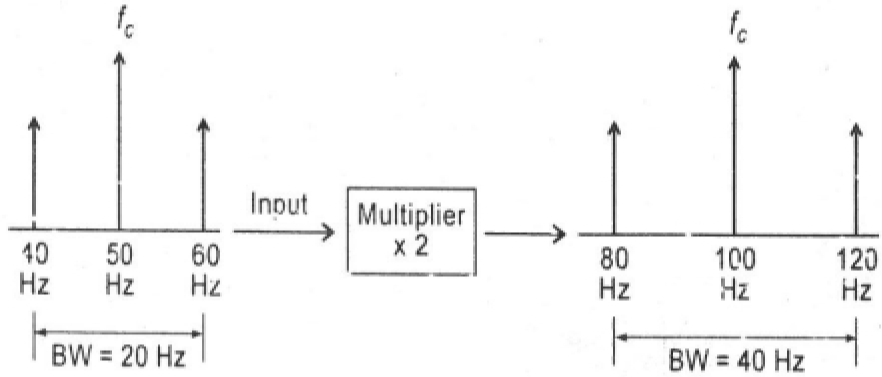
Mixer



Note : *Output of Mixer is f_0 + input in this example, it can also be difference as used in SHR's. Refer section 7.2.1.*

- Thus, Mixer just upscales the input signal.
- Also, bandwidth remains same.

Multiplier



- Thus, BW varies in case of multiplier.

Q.5. *Why is AGC required in radio receivers ? Sketch a diode detector circuit with simple AGC and explain its working.*

Ans. For AGC refer section 7.7 and for diode detector refer chapter 6 section 6.1.2.

Q.6. *Write short note on Squelch circuit.*

Ans. Refer section 7.8.

Q.7. *What is double spotting ?*

Ans. Refer section 7.4.

Q.8. *Explain local oscillator tracking and tracking error in radio receiver.*

Ans. Refer section 7.5.

Q.9. *What is blocking in a receiver ? How is good blocking achieved ? What will be the effect on a communication receiver if it's blocking performance is poor ?*

Ans. For blocking in a receiver refer section 7.4 and for communication receiver refer section 7.8.

Q.10. *Sketch the block diagram of an FM receiver. Why is an Amplitude Limiter required ?*

Ans.

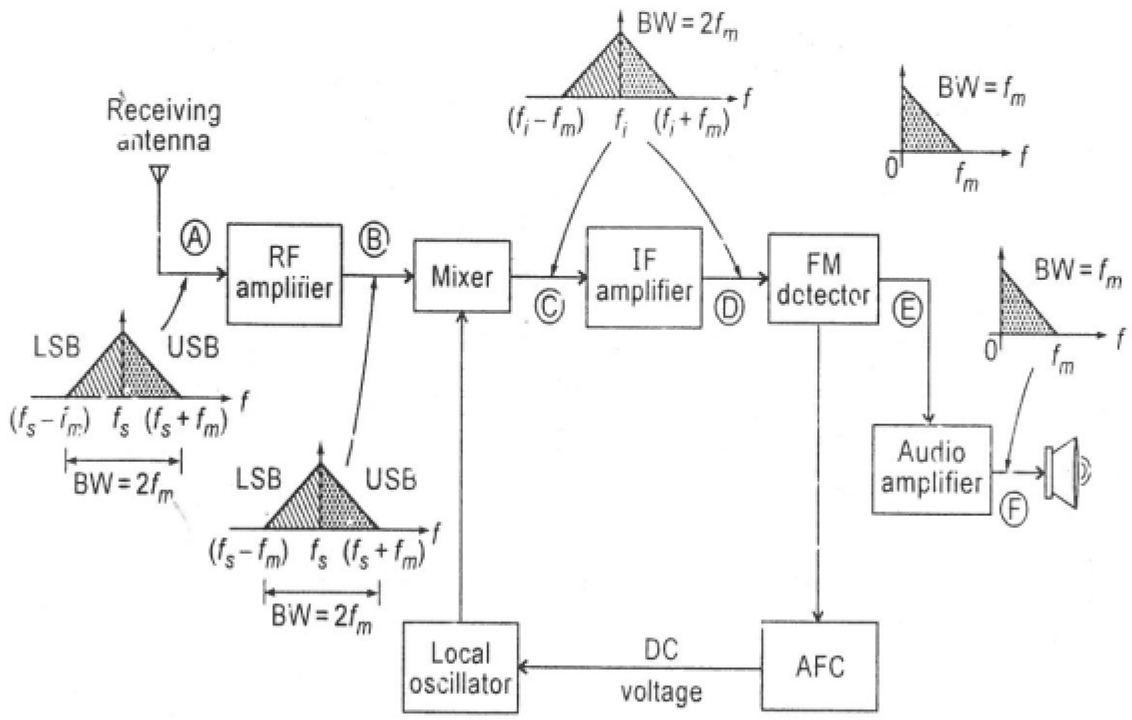


Fig. 7.19

Explain all the blocks from section 7.3 and Q.3. of FAQ's. Also refer previous chapter for use of Amplitude Limiter.

Q.11. Draw the block diagram of Crosby direct FM transmitter and explain its operation.

Ans.

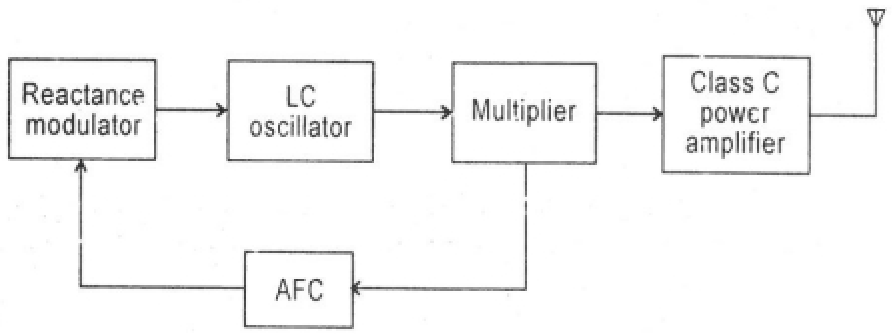


Fig. 7.20

In conventional reactance modulator, LC oscillators are used. But the frequency of LC oscillator drifts due to many factors mainly by temperature, therefore f_c also drifts. To overcome this problem AFC is used.

Write about AFC from Q.3. of FAQ's. Also write about reactance modulator from previous chapter.

Q.12. *Draw and explain AM receiver with forward AGC. What is the purpose of Squelch circuit ?*

Ans. For squelch circuit refer section 7.8.

For AM receiver with forward AGC :

Explain SHR from section 7.3. and

Explain simple AGC from section 7.7.

Q.13. *Explain adjacent channel selectivity.*

Ans. Write about image frequency from section 7.4.

7.10 Formulae

(1) Image frequency $f_{Si} = f_s + 2f_i$

f_i = Intermediate frequency, normally $f_i = 455$ kHz

f_s = Signal frequency.

(2) Image frequency rejection ratio

$$\alpha = \sqrt{1 + Q^2\rho^2}$$

where Q = Quality factor

$$\rho = \frac{f_{Si}}{f_s} - \frac{f_s}{f_{Si}}$$

(3) Local oscillator frequency

$$f_o = f_s + f_i$$

7.11 Solved Problems

Problem 1 : *When a superhetrodyne radio receiver is tuned to 555 kHz, its local oscillator frequency is 1010 kHz. What is image frequency.*

Given : f_s = 555 kHz

f_o = 1010 kHz

To find : f_{si}

Solution :

Image frequency, $f_{si} = f_s + 2f_i$

but $f_i = f_o - f_s$

$$\therefore \boxed{f_i = 455 \text{ kHz}}$$

$$\therefore f_{si} = 555 + 2(455)$$

$$\therefore \boxed{f_{si} = 1465 \text{ kHz}}$$

Problem 2 : A receiver is tuned to 3 - 30 MHz and IF frequency is 40.525 MHz.

(i) Find range of local oscillator frequency and image frequency. Bandwidth = 10 kHz.

(ii) Draw frequency response of IF and AF amplifiers.

Given : $f_{s_{min}} = 3\text{ MHz}$, $f_{s_{max}} = 30\text{ MHz}$

$f_i = 40.525\text{ MHz}$, $BW = 10\text{ kHz}$

To find : $f_{o_{max}}$, $f_{o_{min}}$, $f_{si_{max}}$, $f_{si_{min}}$

Solution :

(i) We know that, $f_o = f_s + f_i$

$$\therefore \quad f_{o_{max}} = f_{s_{max}} + f_i \qquad (\because f_i \text{ is always constant})$$

$$\therefore \quad f_{o_{max}} = 30\text{ MHz} + 40.525\text{ MHz}$$

$$\boxed{f_{o_{max}} = 70.525\text{ MHz}}$$

Similarly,

$$\boxed{f_{o_{min}} = 43.525\text{ MHz}}$$

Also,

$$f_{si} = f_s + 2f_i$$

$$\therefore \quad f_{si_{max}} = f_{s_{max}} + 2f_i$$

$$\therefore \quad f_{si_{max}} = 30\text{ MHz} + 2(40.525)\text{ MHz}$$

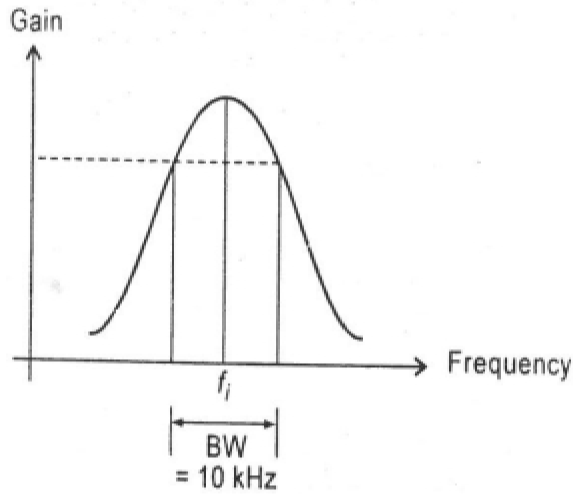
$$\boxed{f_{si_{max}} = 111.050\text{ MHz}}$$

Similarly,

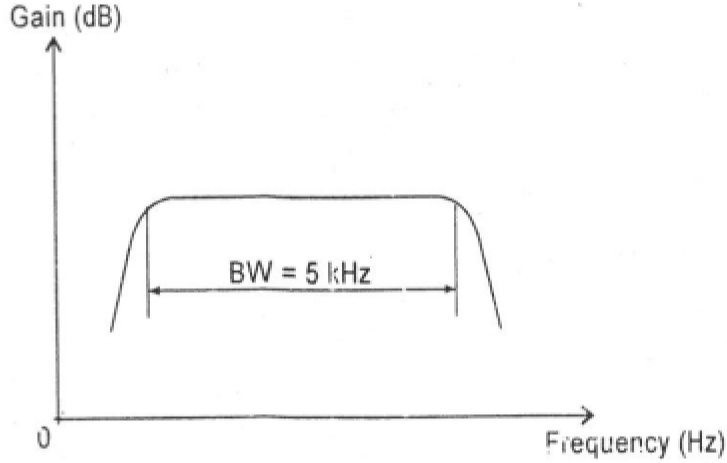
$$\boxed{f_{si_{min}} = 84.050\text{ MHz}}$$

(ii) Frequency Response of IF and AF Amplifiers

(a) IF Amplifier



(b) AF Amplifier



Problem 3 : *In an AM receiver having no RF stage the loaded Q of the arial coupling circuit is 125. IF is 465 kHz find*

(i) Image frequency and its rejection at 1 MHz and 30 MHz.

(ii) The IF required to make the image rejection ratio as good as 30 MHz as it is at 1 MHz.

Given : $f_i = 465$ kHz

$Q = 125$

To find : f_{si} and α in both cases.

Solution :

(i) For f_s = 1 MHz

$$f_{si} = f_s + 2f_i = 1 \text{ MHz} + 2(465) \text{ kHz}$$

$$= 1000 \text{ kHz} + 930 \text{ kHz}$$

$$\boxed{f_{si} = 1.93 \text{ MHz}}$$

Now, rejection ratio is given by

$$\alpha = \sqrt{1 + Q^2\rho^2}$$

$$\text{and } \rho = \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}}$$

$$\therefore \boxed{\rho = 1.4118}$$

$$\therefore \alpha = \sqrt{1 + Q^2\rho^2}$$

$$\boxed{\alpha = 176.486}$$

For f_s = 30 MHz

$$f_{si} = f_s + 2f_i = 30000 \text{ kHz} + 2(465) \text{ kHz}$$

$$\boxed{f_{si} = 30.93 \text{ MHz}}$$

Now, rejection ratio is given by

$$\alpha = \sqrt{1 + Q^2\rho^2}$$

$$\text{and } \rho = \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}}$$

$$\therefore \boxed{\rho = 0.061}$$

$$\therefore \boxed{\alpha = 7.69}$$

(ii) Required α = 176.486 and f_s = 30 MHz

We need to find f_i

Putting new value of α and Q in

$$\alpha = \sqrt{1 + Q^2\rho^2}$$

$$\therefore \quad 125^2 \rho^2 + 1 = \alpha^2$$

$$\therefore \quad 125^2 \rho^2 + 1 = 31146.3$$

$$\therefore \quad \rho^2 = 1.99336$$

$$\boxed{\rho = 1.41186}$$

$$\therefore \quad \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}} = 1.41186$$

$$\therefore \quad \frac{f_{si}^2 - f_s^2}{f_s f_{si}} = 1.41186$$

$$f_{si}^2 - f_s^2 = 1.41186\, f_s f_{si}$$

Put f_s = 30 MHz

$$\therefore f_{si}^2 - 42.355\, f_{si} - 900 = 0$$

Solving this we get

$$\boxed{f_{si} = 57.898 \text{ MHz}}$$

$$\therefore \quad f_s + 2f_i = 57.898 \text{ MHz}$$

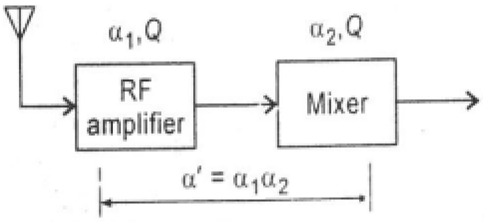
$$\therefore \quad 2f_i = 27.898 \text{ MHz}$$

$\therefore$ New IF frequency is

$$\therefore \boxed{f_i = 13.949 \text{ MHz}}$$

• ***Problem 4 :*** *Consider the following block diagram of an SHR receiver.*

Given that, overall rejection of image frequency is 100, find Q, both blocks have same Q. Also f_s = 1 MHz and f_i = 455 kHz.



Given :

$$\alpha' = 100$$

$$f_s = 1 \text{ MHz}$$

$$f_i = 455 \text{ kHz}$$

To find : Q

Solution : Image frequency is

$$f_{si} = f_s + 2f_i$$

$$\therefore \quad f_{si} = 1000 \text{ kHz} + 910 \text{ kHz}$$

$$\boxed{f_{si} = 1.91 \text{ MHz}}$$

Now, for RF amplifier

$$\alpha_1 = \sqrt{1 + Q^2\rho^2}$$

$$\text{and } \rho = \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}}$$

$$\therefore \boxed{\rho = 1.3864}$$

$$\therefore \boxed{\alpha_1 = \sqrt{1 + 1.922\, Q^2}}$$

Similarly,

$$\therefore \boxed{\alpha_2 = \sqrt{1 + 1.922\, Q^2}}$$

Now,

$$\alpha' = \alpha_1 \alpha_2 = 100$$

$$\therefore (1 + 1.922\, Q^2) = 100$$

$$\therefore Q^2 = 51.508$$

$$\therefore \boxed{Q = 7.17}$$

Problem 5 : *The broadcast superhetrodyne receiver has intermediate frequency 455 kHz and it is tuned for 1500 kHz. Calculate the image frequency and the quality factor of the tuned circuit having image frequency rejection ratio equal to 75.*

Given :

$$f_i = 455 \text{ kHz}$$

$$f_s = 1500 \text{ kHz}$$

$$\alpha = 75$$

To find : f_{si}, Q

Solution :

We know, $f_{si} = f_s + 2f_i$

$$\therefore f_{si} = 1500 \text{ kHz} + 2(455 \text{ kHz})$$

$$\therefore \boxed{f_{si} = 2410 \text{ kHz}}$$

Now, rejection ratio is given by

$$\alpha = \sqrt{1 + Q^2 \rho^2} \qquad .....(1)$$

$$\text{and } \rho = \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}}$$

$$\therefore \boxed{\rho = 0.984}$$

Put in equation (1), we get

$$\therefore \alpha = \sqrt{1 + Q^2 (0.984)^2}$$

$$\therefore 75 = \sqrt{1 + 0.968\, Q^2}$$

$$\therefore 1 + 0.968\, Q^2 = 5625$$

$$\therefore Q^2 = \frac{5624}{0.968}$$

$$\therefore \boxed{Q = 76.22}$$

❋❋❋

8 Analog Pulse Modulation

Topic	Theory imp	Oral imp
Introduction	☆	☆☆☆☆
Sampling and Sampling Theorem	☆☆	☆☆☆☆
Types of Sampling	☆	☆
Pulse Amplitude Modulation	☆☆☆	☆☆
Pulse Width Modulation	☆☆☆	☆☆
Pulse Position Modulation	☆☆☆	☆☆
Comparisons	☆☆	☆☆☆☆
FAQ's	☆☆	☆

8.0 Introduction

- In analog modulation carrier was a continuous periodic analog signal but in pulse modulation the carrier is a continuous periodic digital signal i.e. a train of pulses.
- Why we use pulse modulation ?

  The answer is digital signals are immune to noise.
- Why we then use analog modulation ?

  The answer is digital signal requires high bandwidth and there is conversion cost from analog to digital as most practical signals are analog signals.
- Hierarchy of pulse modulation

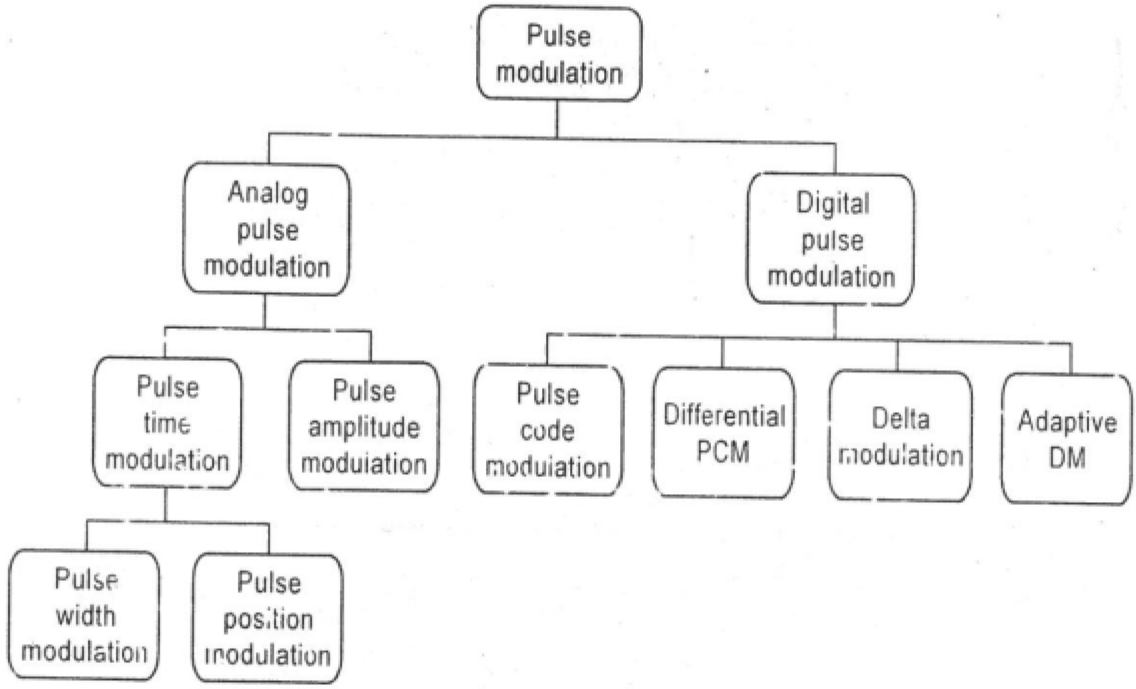


Fig. 8.1

Description

(a) Pulse Modulation

- Some parameter of a *pulse train* is varied in accordance with the message signal.

(b) Analog Pulse Modulation

- A periodic pulse train is used as the carrier wave and some characteristic feature of each pulse (i.e., amplitude, duration or position) is varied in a *continuous manner* in accordance with the corresponding *sample value* of the message signal.
- Information is transmitted basically in analog form, but the transmission takes place at discrete time instants.

(c) Digital Pulse Modulation

- Message signal is represented in a form that is discrete in both time and amplitude.
- The transmission is done in *digital form* as a sequence of *coded pulses*

8.1 Sampling and Sampling Theorem

Q.1. State and prove sampling theorem for low pass band signals.

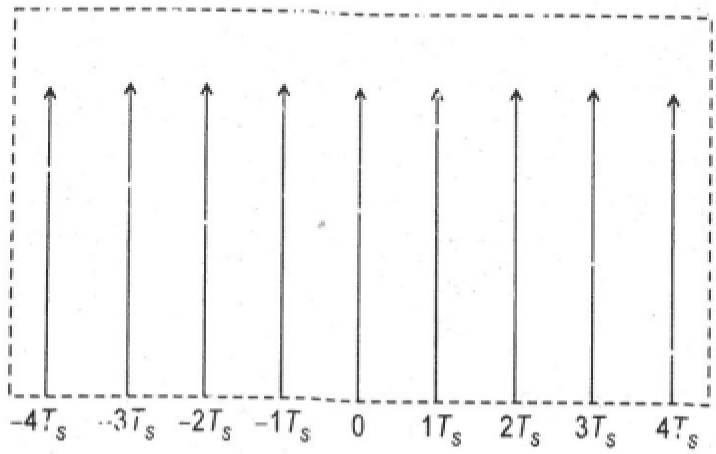
Q.2. State sampling theorem.

Q.3. State sampling theorem. Draw the spectrum of any signal which is properly sampled.

Q.4. Explain the term aliasing and its effects. How it can be eliminated ?

Q.5. State and prove sampling theorem.

- In signal processing, sampling is the conversion of a continuous signal to a discrete signal. A common example is the conversion of a sound wave (a continuous-time signal) to a sequence of samples (a discrete-time signal).
- A *sample* refers to a value or set of values at a point in time and/or space.
- A *sampler* is a subsystem or operator that extracts samples from continuous signal.
- A theoretical ideal sampler multiplies a continuous signal with a Dirac comb signal (an impulse train). This multiplication "picks out" values but the result is still continuous-valued. If this signal is then discretized (i.e., converted into a sequence) and quantized along all dimensions, it becomes a discrete signal.
- Sampling rate is the rate at which samples are taken.



where T_s = Sampling interval

$$f_s = \frac{1}{T_s}$$

f_s = Sampling rate

Fig. 8.2 : Dirac comb signal.

- The sampling theorem by C.E. Shannon is stated as follows :

  In order to recover the signal function $f(t)$ exactly, it is necessary to sample $f(t)$ at a rate greater than or equal to twice its highest frequency component.
- Practically speaking for e.g., to sample an analog signal having a maximum frequency of 2 kHz requires sampling at greater than or equal to 4 kHz to preserve and recover the waveform exactly.
- The minimum sampling required is called *nyquist rate* and inverse nyquist rate of it is called *nyquist interval*.

- The consequences of sampling a signal at a rate below its highest frequency component results in a phenomenon known as *aliasing*. This concept results in a frequency mistakenly taking on the identity of an entirely different frequency when recovered. In an attempt to clarify this, consider the ideal sampler of *figure 8.3(a)*, with a sample period of T shown in *figure 8.3(b)*, sampling the waveform $f(t)$ as pictured in *figure 8.3(c)*. The sampled data points of $f(t)$ are shown in *figure 8.3(d)* and can be represented as $f'(t)$.

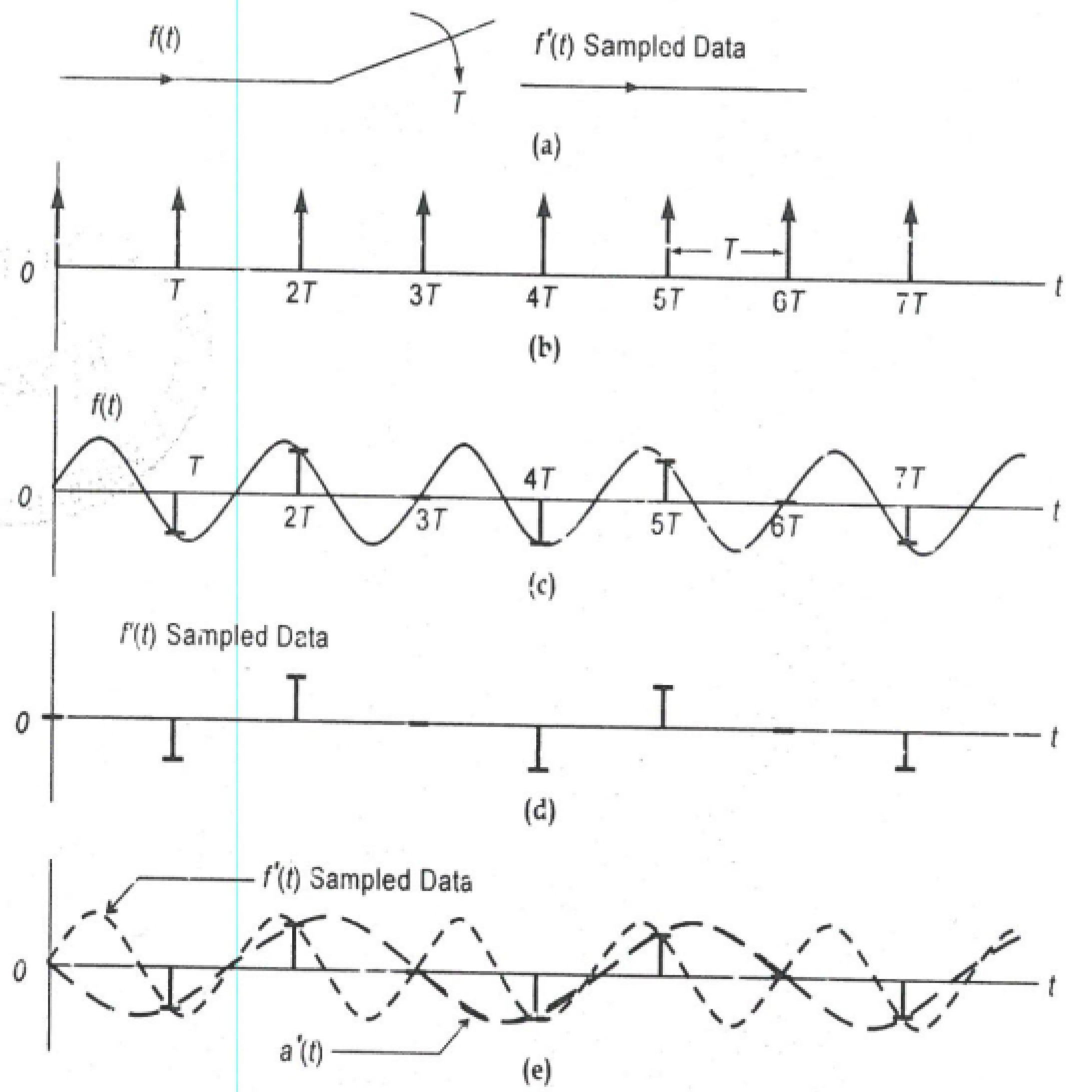


Fig. 8.3

- Note in *figure 8.3(e)* that another frequency component, i.e. another signal $a'(t)$, can be found that has the same sample set of data points as $f'(t)$ in *(d)*. Because of this it is difficult to determine which is the original signal. This is the **Aliasing Effect**.
- On the surface it is easily said that anti-aliasing designs can be achieved by sampling at a rate greater than twice the maximum frequency found within the signal to be sampled.

- In the real world, however, most signals contain the entire spectrum of frequency components. To recover such information accurately the system would require an unrealizable or infinite sampling rate, which is not possible.
- This difficulty can be easily overcome by preconditioning the input signal, the means of which would be a band-limiting or frequency filtering function performed prior to the sample data input. The prefilter, typically called *anti-aliasing filter* guarantees, for example in the low pass filter case, that the sampled data system receives analog signals having a spectral content no greater than those frequencies allowed by the filter.
- Sampling a continuous signal $m(t)$ with frequency f_s results in a new signal that can be represented as $m(nT_s)$ where $T_s = (1/f_s)$ and n varies from $-\infty$ to ∞.

8.2 Types of Sampling

- Sampling can be classified as follows

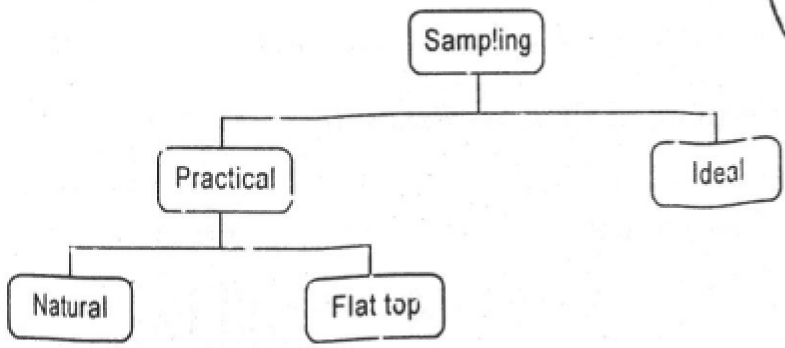


Fig. 8.4

(a) Ideal

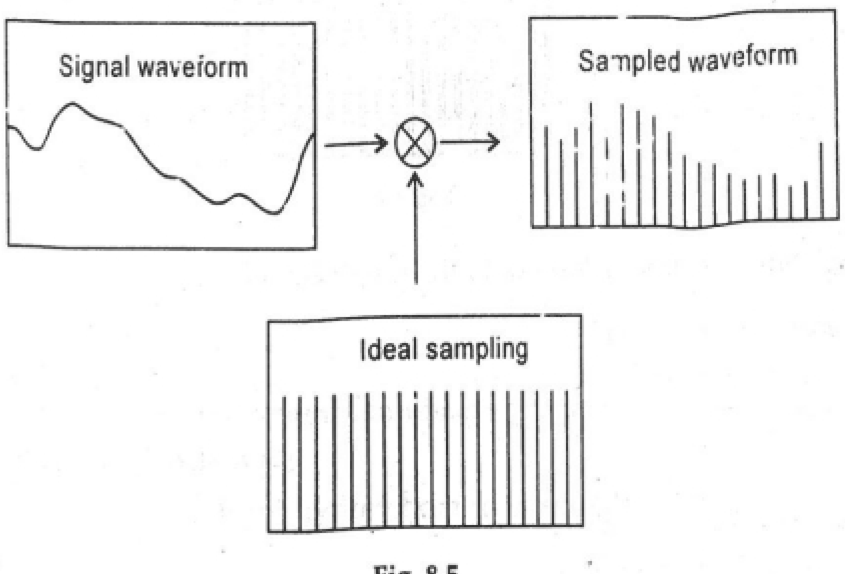


Fig. 8.5

- Sampling function is train of impulses.
- Sampling function mathematically is given by

$$S(t) = \sum_{-\infty}^{\infty} \delta(t - nT_s) \qquad T_s = \frac{1}{f_s}$$

where T_s = Sampling interval

f_s = Sampling rate or frequency

- The sampled signal is given by

$$X_\delta(t) = X(t) \times S(t)$$

$$X_\delta(nT_s) = \sum_{n=-\infty}^{\infty} X(nT_s) \cdot \delta(t - nT_s)$$

- It is a mathematical model to study sampling.
- Practically impossible to implement
- Can be reconstructed using ideal low pass filter.

(b) Natural Sampling or Chopper Sampling

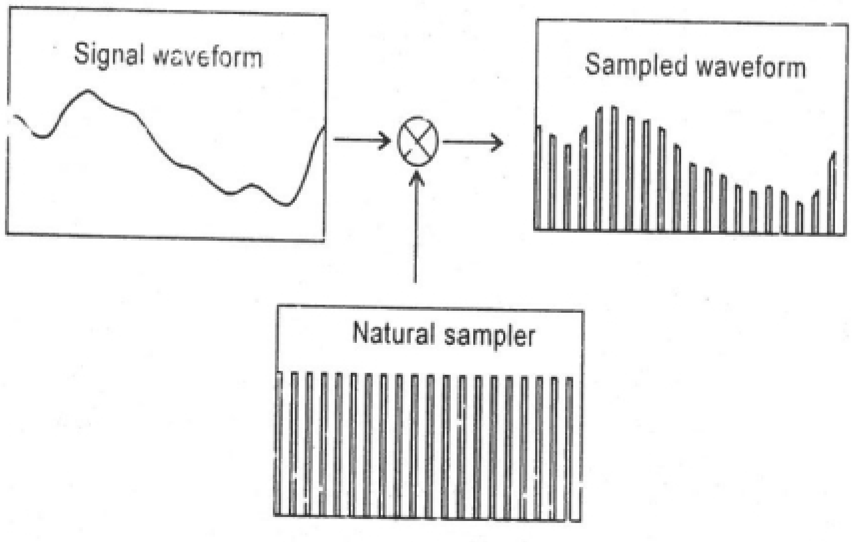


Fig. 8.6

- Sampling function is train of finite length impulses.
- **Generator Circuit**

$S(t)$ = Sampling waveform (Train of pulses : Pulse duration 'τ' are separated by sampling time T_s)

$X(t)$ = Baseband or modulating signal

$X_\delta(t) = S(t)\,X(t)$

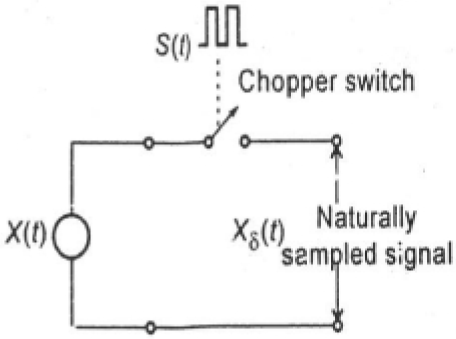


Fig. 8.7(a) : Chopper sampling.

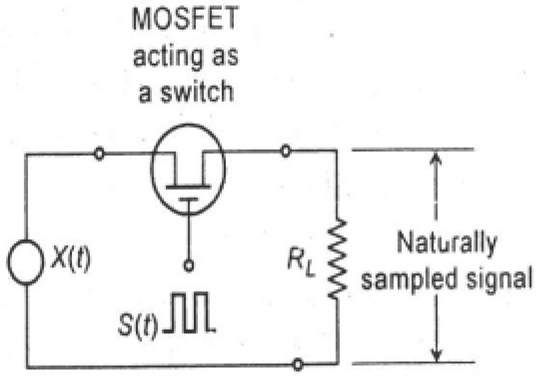


Fig. 8.7(b) : Chopper sampling with MOSFET acting as a switch.

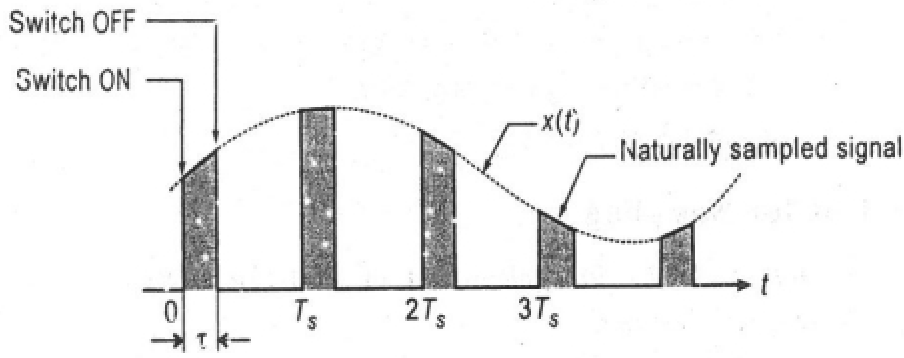


Fig. 8.7(c) : Naturally sampled signal.

Sampled signal is a train of pulses of width 'τ', whose amplitudes are varying.

- Natural sampling is sometimes called as *chopper sampling* because the waveforms of sampled signal appear to be chopped off from the continuous time signal $x(t)$.
- Reconstruction can be done passing through low pass filter. Passing through an ideal low pass filter will give an original signal.

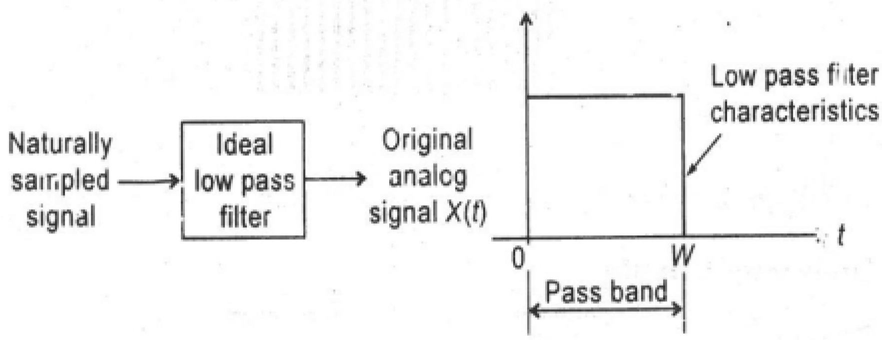


Fig. 8.8 : Reconstruction of original signal from naturally sampled signal.

- **Cross Talk :** It is interference of signals in same channel
  - With samples of finite duration it is not completely possible to eliminate crosstalk generated in the channel.
  - When N number of channels are multiplexed, maximum duration each sample can have is $\tau = T_s/N$.

- The τ value should be as large as possible to have signal power.
- To reduce cross talk, τ should be as low as possible.
- Hence τ is trade off factor between crosstalk and signal power

- **Advantages**
  - Generation is easy.
  - Low pass filter can be used for reconstruction.
  - SNR ratio is high
  - Bandwidth required is less
- **Disadvantages**
  - The amplitude of high frequency decreases, therefore distortion is introduced during reconstruction
  - Possibility of cross talk

(c) Flat Top Sampling

Q. *Sketch a circuit for generation of Flat-top sampled signal and explain briefly its working.*

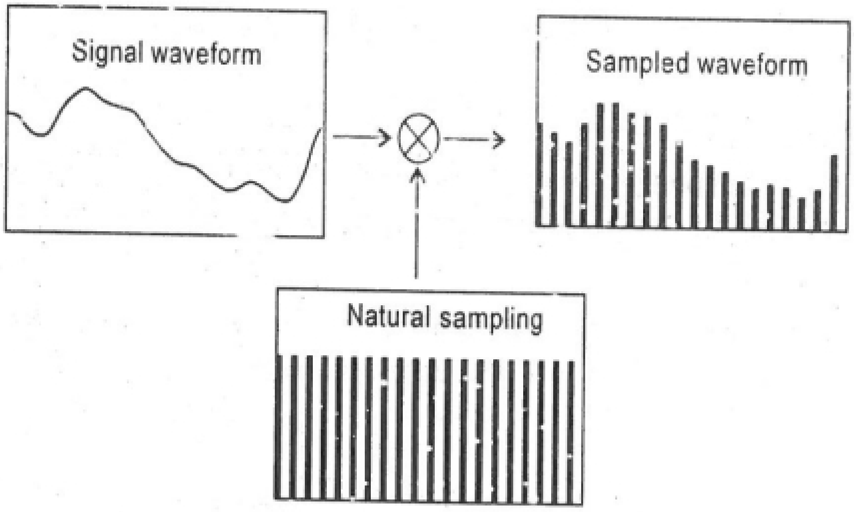


Fig. 8.9

- Sampling function is train of finite length impulses.
- **Generator Circuit**

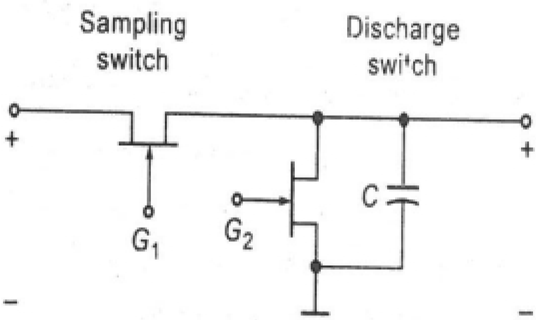


Fig. 8.10(a)

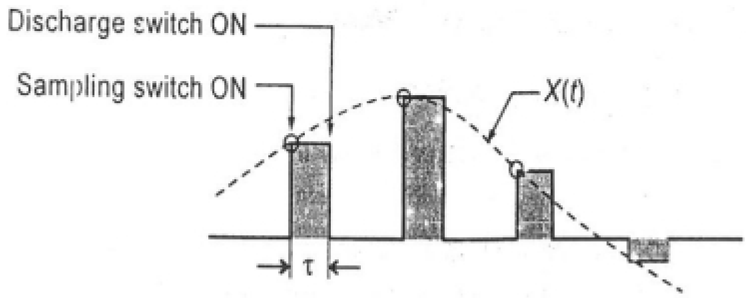
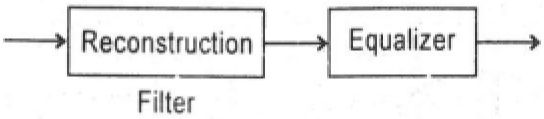


Fig. 8.10(b) : Operation of sample and hold circuit.

- G_1 and G_2 are FET switches.
- At start of sampling time τ, G_1 is switched ON for short period of time and the capacitor is charged with voltage V.
- There is no discharge path for capacitor, till we switch on G_2 i.e. at end of τ. Hence voltage across the output is equal to the voltage across capacitor during time τ which is constant i.e. we get flat top samples.

- Reconstruction can be done passing through low pass filter and equalizer
  - When we pass signal through low pass filter high frequency component are attenuated.
  - This loss in high frequency component is called *aperture effect*.
  - This aperture effect causes amplitude distortion and delay.
  - This can be removed by equalizer.



- **Advantages**
  - Generation is easy.
  - Low pass filter can be used for reconstruction.
  - SNR ratio is high.
  - Bandwidth required is less.

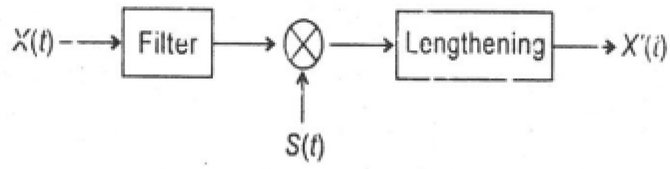
- **Disadvantages**
  - The amplitude of high frequency decreases, therefore distortion is introduced during reconstruction.
  - Possibility of cross talk.
  - Aperture effect introduces distortion.

8.3 Pulse Amplitude Modulation (PAM)

Q.1. Explain generation of PAM and PWM.

Q.2. Explain how PAM signal can be generated and demodulated.

- PAM is a pulse modulation system in which amplitude of the digital carrier is varied according to amplitude of the baseband signal
- Mathematical model of PAM modulator



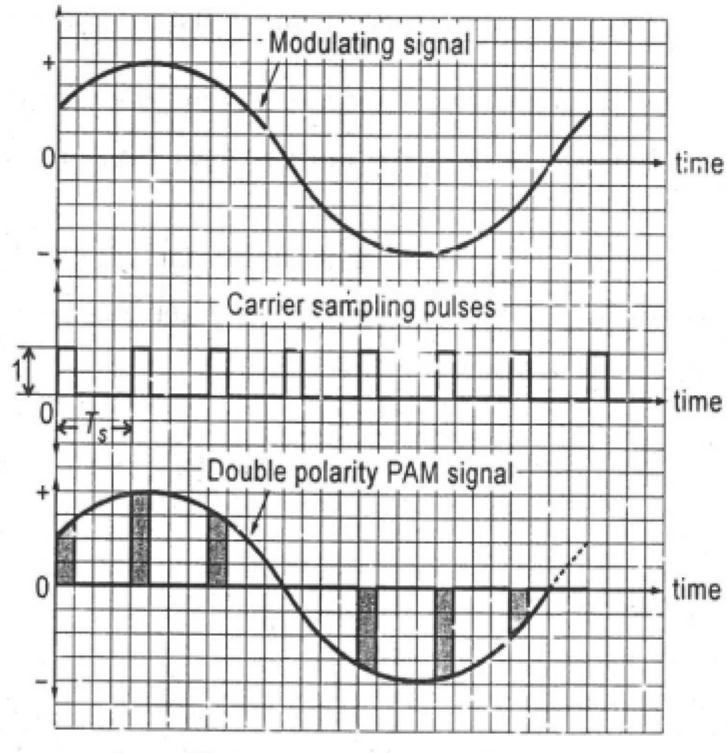


Fig. 8.11 : Waveform of PAM.

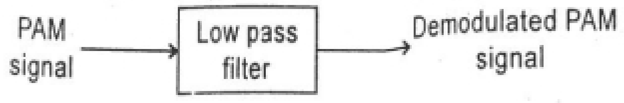
- **Operations Performed**

  (a) Low-pass filtering to prevent aliasing

  (b) Ideal sampling or multiplication - It multiplies the two signal.

  (c) Lengthening the duration of each sample. The use of lengthening is

  - To reduce the required transmission bandwidth (*BW*). Since it is inversely proportional to pulse duration.
  - To get the exact signal value.



PAM detector

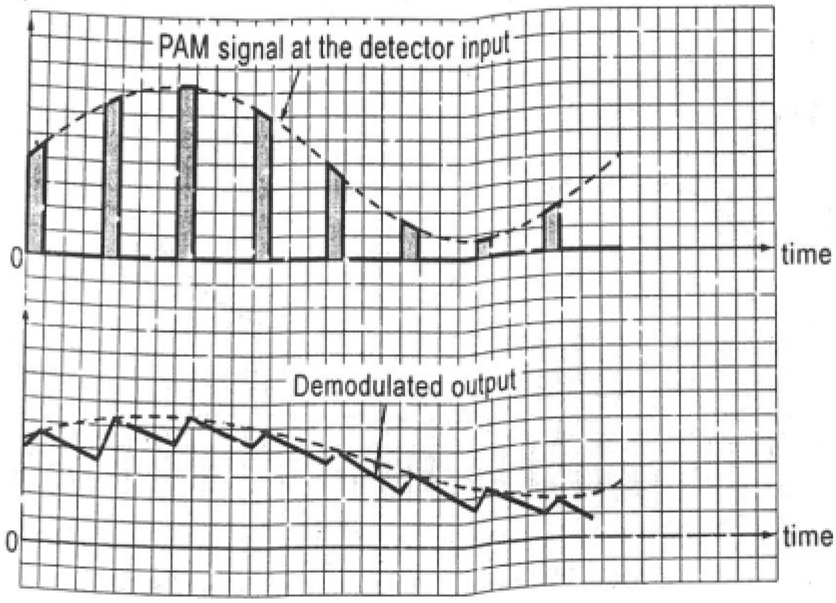


Fig. 8.12 : Detection of PAM and waveforms.

- **PAM Generation Circuits**

  (a) Natural sampling or (b) Flat top sampling.

- **PAM Demodulator**

  It is done by passing it through low pass filter.

- **Advantages and Disadvantages**
  - Very simple system.
  - Effect of noise is maximum on PAM systems as energy is stored in amplitude.
  - High bandwidth.
  - Transmitted power is not constant
  - No synchronisation between transmitter and receiver required.

8.4 Pulse Width Modulation (PWM)

Q.1. Explain the generation and detection of PWM.

Q.2. Explain generation and demodulation of PWM signal with the help of suitable diagrams.

- In pulse width modulation, width of pulse of digital carrier is varied in accordance with the amplitude of the baseband signal.

- Hence information is contained in the width.
- E.g.

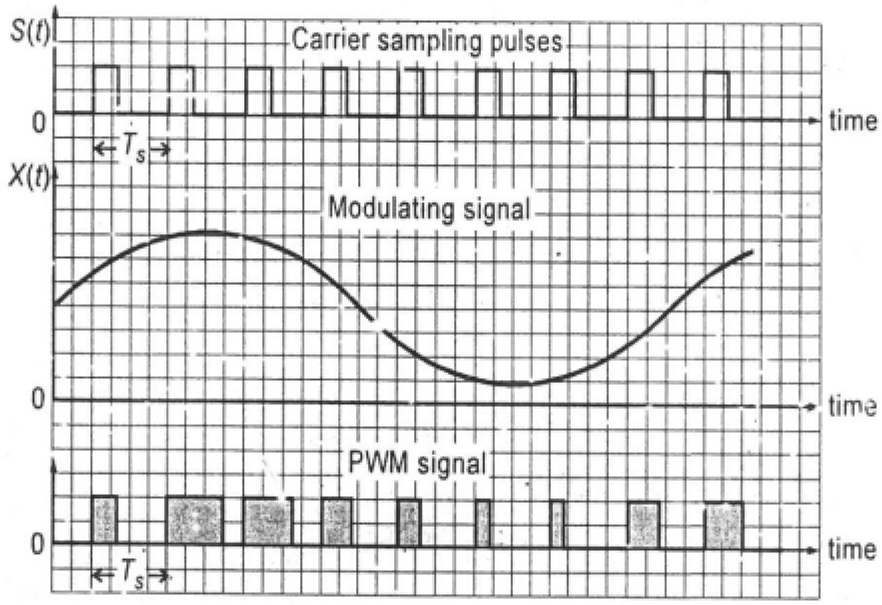


Fig. 8.13 : PWM signal [Trial edge modulated signal].

- **Generator Circuit**

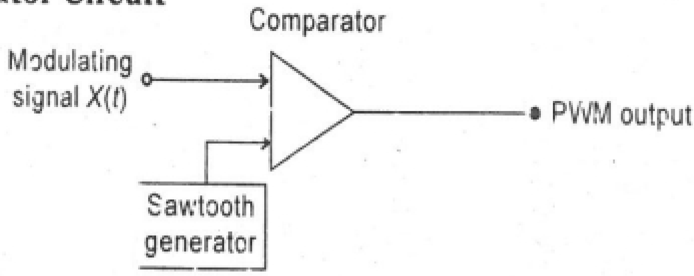


Fig. 8.14 : PWM generator.

- **Wave Forms**

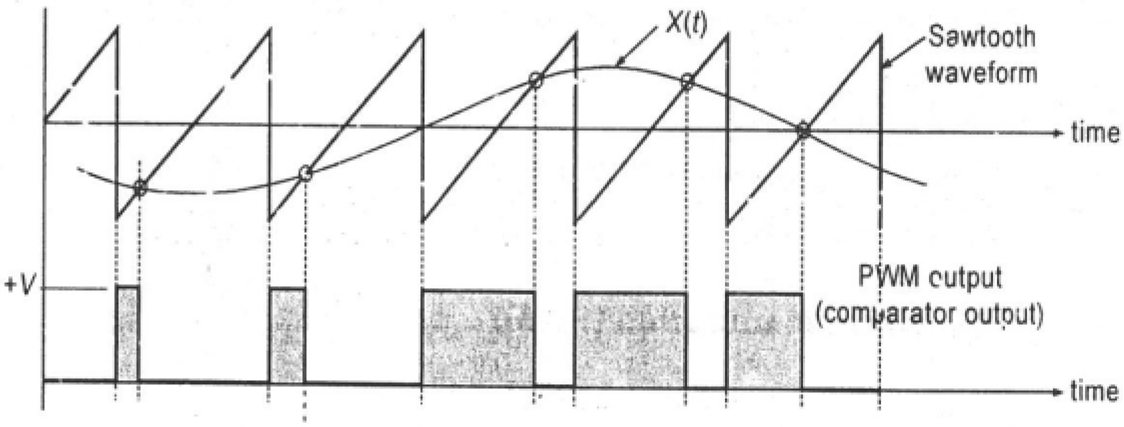


Fig. 8.15 : Waveforms.

  - Comparator compares the two signal.
  - If voltage $X(t)$ is higher than the saw tooth waveform voltage, it gives positive voltage V for that much time.
  - Hence we can see that pulses are of variable length.

- **Demodulator**

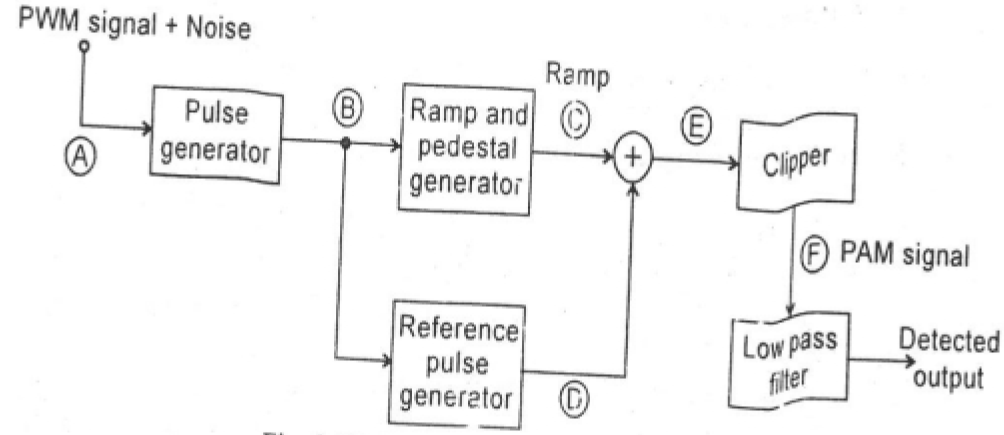


Fig. 8.16 : PWM detection circuit.

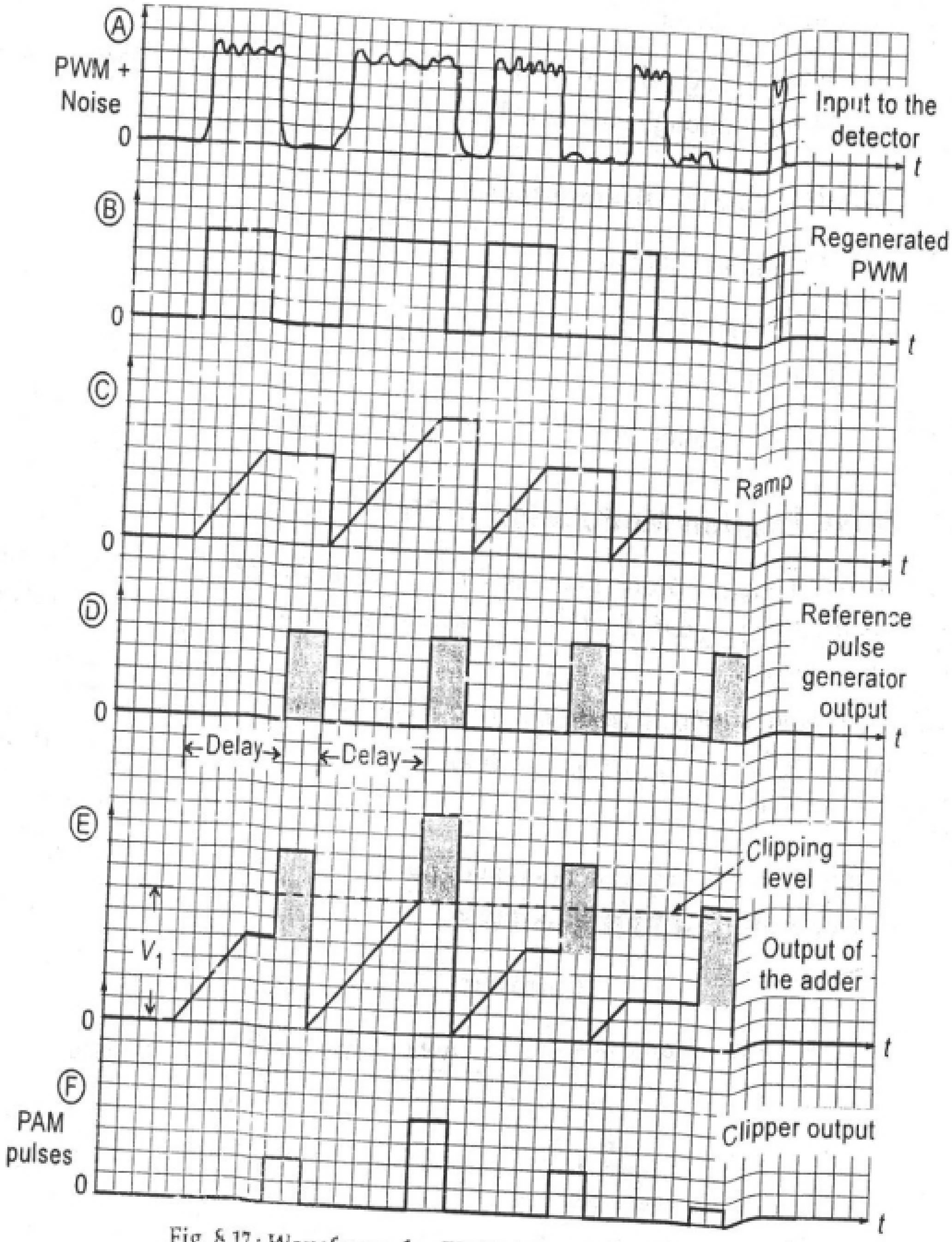


Fig. 8.17 : Waveforms for PWM detection circuit.

- **Pulse Generator**
  - It removes the effect of noise to give waveform B.
- **Ramp and Pedestal Generator**
  - Generates a flat top when B waveform voltage is low and a slant edge when B waveform voltage is high.
  - Reference Pulse generator generates a pulse after delay.
  - After adding C and D we get E waveform.
  - Clipper shifts the DC reference level to V_1 after clipping we get a PAM waveform signal.
  - Now the original signal can be easily recovered from PAM signal by passing it through low pass filter.
- **Advantages and Disadvantages**
  - Good noise immunity.
  - Variable power.
  - High bandwidth.
  - Synchronization between transmitter and receiver is required.

8.5 Pulse Position Modulation (PPM)

Q.1. Sketch a system block diagram that gives PWM and PPM outputs for an analog input signal. Explain its working giving waveforms at the output of each block.

Q.2. Explain how PPM is generated from PWM signal.

Q.3. What is pulsed position modulation ? How it can be generated and demodulate ?

- Information is stored in position of pulse.
- **Generation of PPM from PWM**

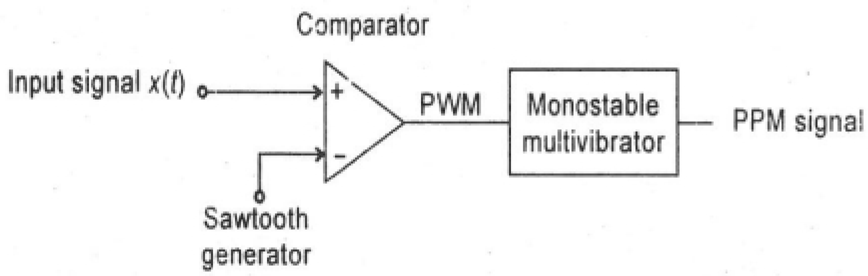


Fig. 8.18 : Generation of PPM signal.

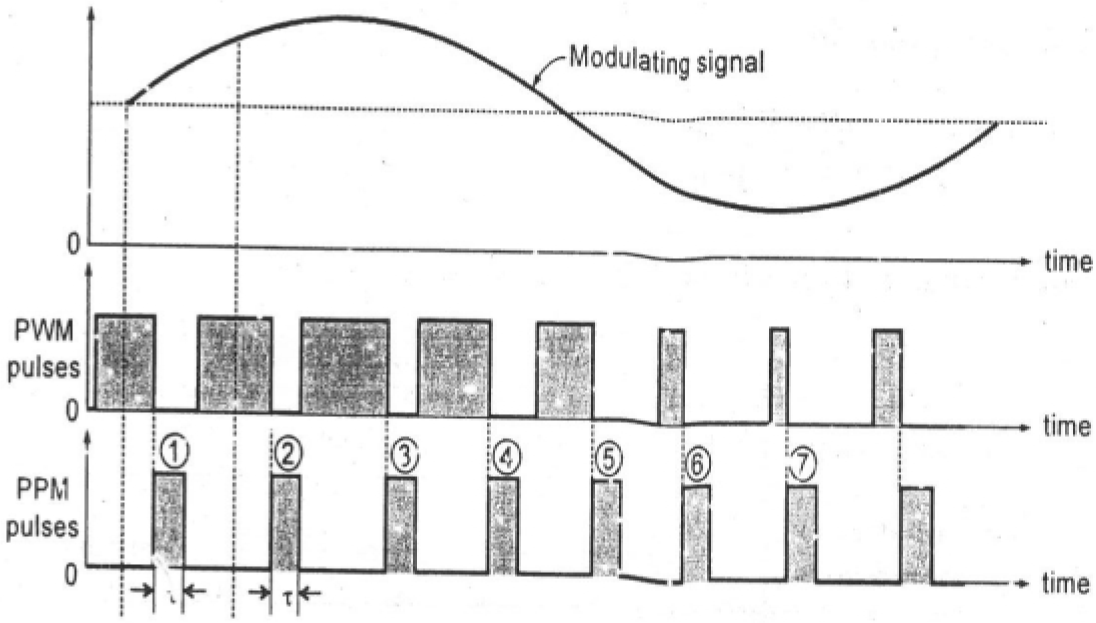


Fig. 8.19 : PPM pulses generated from PWM signal.

- PPM can be easily generated from PWM.
- Attach a monostable multivibrator to the end of PWM generator to get a PPM signal.
- Monostable multivibrator generates a pulse with width τ at every falling edge.
- From the waveform we can easily see that a PPM can be easily generated from PWM.

- **PPM Demodulator with the Help of PWM**

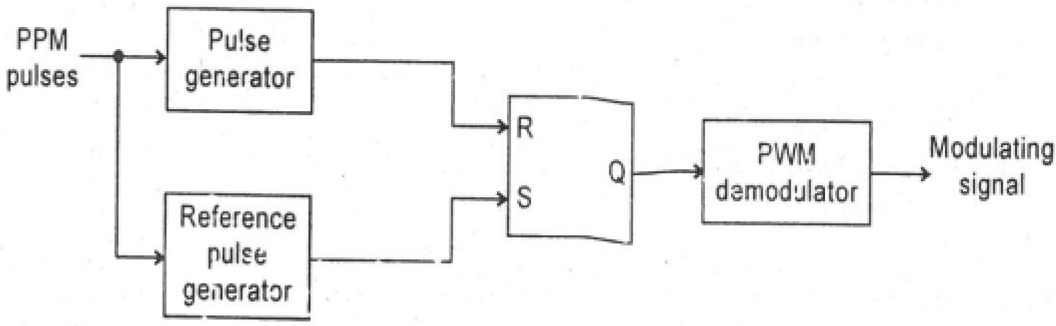


Fig. 8.20 : PPM demodulator circuit.

- The SR flip-flop, pulse generator and reference pulse generator converts the PPM signal to PWM signal which is then demodulated by PWM demodulator.

- **Advantages and Disadvantages**
  - High noise immunity.
  - Synchronization between transmitter and receiver is required.
  - High bandwidth.
  - Constant power

8.6 Comparison

Q.1. Compare PAM and PWM.

Q.2. Compare PWM, PAM and PPM.

Comparison Between PAM, PWM and PPM

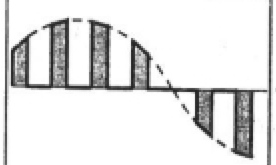
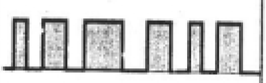
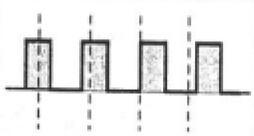
	PAM	PWM	PPM
Type of carrier	Train of pulse	Train of pulse	Train of pulse
Information is stored in	Amplitude	Width	Position
Bandwidth	Low	High	High
Noise immunity	Low	High	Highest
Need to be sent with synchronized pulse	No	Yes	Yes
Transmitted power	Varies with amplitude of pulses	Varies with variation in width	Remains constant
Complexity of generation and detection	Complex	Easy	Complex
Similarity with other modulation systems	Similar to AM	Similar to FM	Similar to PM
Variable characteristics of the pulsed carrier	Amplitude	Width	Position
Output waveforms			

Table 8.1

8.7 Frequently Asked Questions

Q.1. What is the standard sampling frequency for speech signals ?

Ans. The standard sampling frequency for speech signals is 8 kHz.

Q.2. Explain the term - Aperture Effect ?

Ans.

- When we pass the signal through a low pass filter for reconstruction the original signal is not generated.
- This is due to the fact that the original signal also had some high frequency component which is filtered by low pass filter.
- Hence aperture effect can be defined as *the loss of information due to filtering of high frequency component.*
- This can be reduced by reducing the pulse width 'τ' or by an equalizer.

Q.3. A continuous time signal

$$X(t) = 5 \sin 2000 \pi t + 7 \sin 4000 \pi t$$

is to be sampled. Calculate the minimum sampling frequency ?

Ans. There are two frequency components present

$$X(t) = a \sin \omega_1 t + b \sin \omega_2 t$$

Comparing above equation with given equation.

$$\omega_1 = 2000\pi = 2\pi f_1 \Rightarrow f_1 = 1000 \text{ Hz}$$

$$\omega_2 = 4000\pi = 2\pi f_2 \Rightarrow f_2 = 2000 \text{ Hz}$$

$$\text{Max}(f_1, f_2) = \text{Max}(1000, 2000) = 2000$$

According to sampling theorem, sampling rate should be twice the maximum frequency component present in a signal.

$$\therefore \text{ Sampling rate} = 2 \times 2000 = 4000 \text{ Hz} = 4 \text{ kHz}$$

Q.4. List the advantages and disadvantages of PAM, PWM and PPM.

Ans. Refer sections 8.3, 8.4 and 8.5.

Q.5. Explain aliasing error and aperture effect.

Ans. For aliasing refer sections 8.1 and for aperture effect refer same section Q.2.

Q.6. The information signal x(t) having a spectrum shown in figure given below is sampled at frequencies (i) 75 kHz and (ii) 100 Hz. Plot the spectrum of sampled signal.

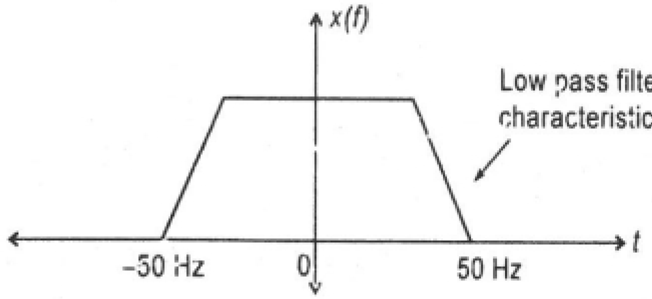


Fig. 8.21

Ans. **Case (i)** $f_s = 75$ kHz

Explain Aliasing from section 8.1.

Spectrum :

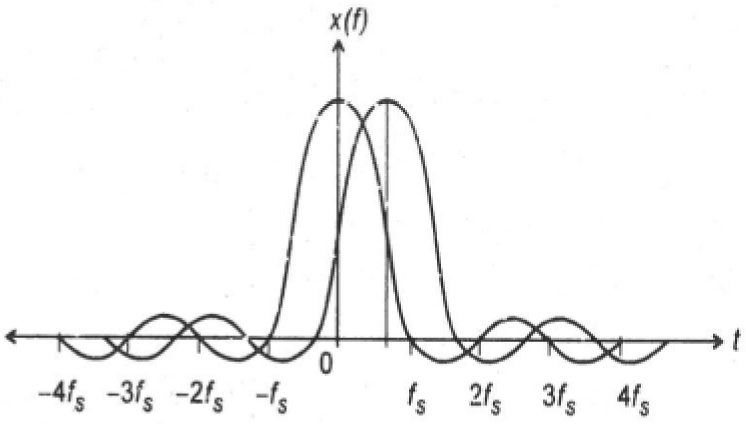


Fig. 8.22

Case (ii) $f_s = 100$ kHz

Spectrum :

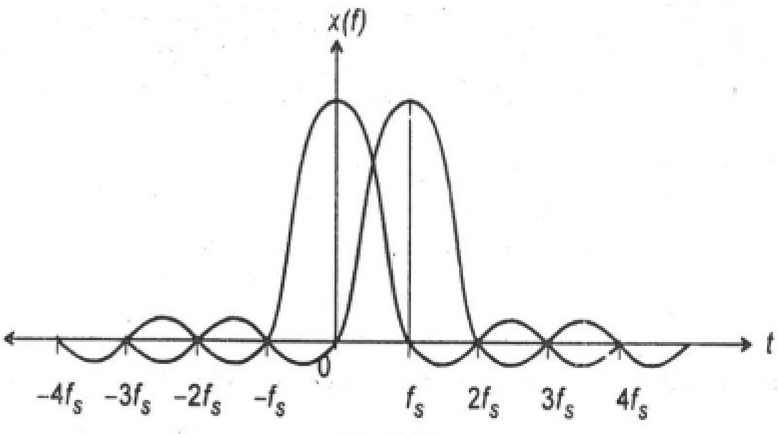


Fig. 8.23

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9 Digital Pulse Modulation

Topic	Theory imp	Oral imp
Introduction	☆	☆☆☆☆
Quantization	☆☆☆	☆☆☆
Quantization Error	☆☆	☆
Types of Quantizer	☆☆	☆☆
PCM	☆☆☆	☆☆☆
DPCM	☆☆	☆☆
DM	☆☆☆	☆☆
ADM	☆☆	☆☆
Comparisons	☆☆☆	☆☆☆☆
FAQ's	☆☆	☆☆

9.0 Introduction

- As in the last chapter, we studied about analog pulse modulation techniques (PAM, PWM and PPM).
- In this chapter we will study about digital pulse modulation techniques.
- Digital pulse modulation
  - Message signal is represented in a form that is discrete in both time and amplitude.
  - The transmission is done in digital form as a sequence of coded pulses.
- In digital pulse modulation, the amplitude of the signal is sampled and then transformed into codes and then transmitted.
- There are four different digital modulation techniques
  - **(a)** Pulse Code Modulation (PCM)
  - **(b)** Differential PCM
  - **(c)** Delta Modulation (DM)
  - **(d)** Adaptive DM
- In the case of digital modulation, addition to sampling, the samples of message signal must be converted into "symbols" by means of quantization.

9.1 Quantization

Q. What is quantization ? Explain with the help of a diagram.

- Quantization basically means approximating the original signal.
- Quantization means mapping the continuous amplitude variation of a signal to discrete set of points.
- It can be said as discretization of magnitude of signal.

> **Note :** *Refresh the concept of sampling from the last chapter especially the last point of section 8.1 in chapter 8.*

- A process of transforming the sampled amplitude $m(nT_s)$ of a message signal $m(t)$ at time $t = nT_s$ into a discrete amplitude $v(nT_s)$ taken from a finite set of possible amplitudes is called *Quantization*.
- The sample $m(nT_s)$ of message signal is mapped into (represented by) v_k if it lies inside the interval

  $m_k < m(nT_s) < m_k+1$; where $k = 1, 2, \ldots L$

  where L is the total number of output levels used in the quantizer.
- The levels m_k; $k = 1, 2, \ldots, L$ are called *decision levels* or *decision thresholds*.
- Output levels v_k where $k = 1, 2, \ldots, L$ are called *representation levels* or *reconstruction levels*.
- The spacing between two adjacent representation levels is called *a quantum* or *step size*.
- Quantizers can be of a *uniform* or *nonuniform* type. In a uniform quantizer, the representation levels are uniformly spaced.
- Consequently, quantizer causes a distortion that is modeled as a *quantization noise*.

9.2 Quantization Error

Q. Explain what is meant by quantization noise.

- Quantization introduces an error defined as a difference between the input signal $x(t)$ and the output signal $x_q(t)$

  $$q(t) = x(t) - x_q(t)$$
- Derivation of signal-to-noise ratio at the quantizer output is given in the next section

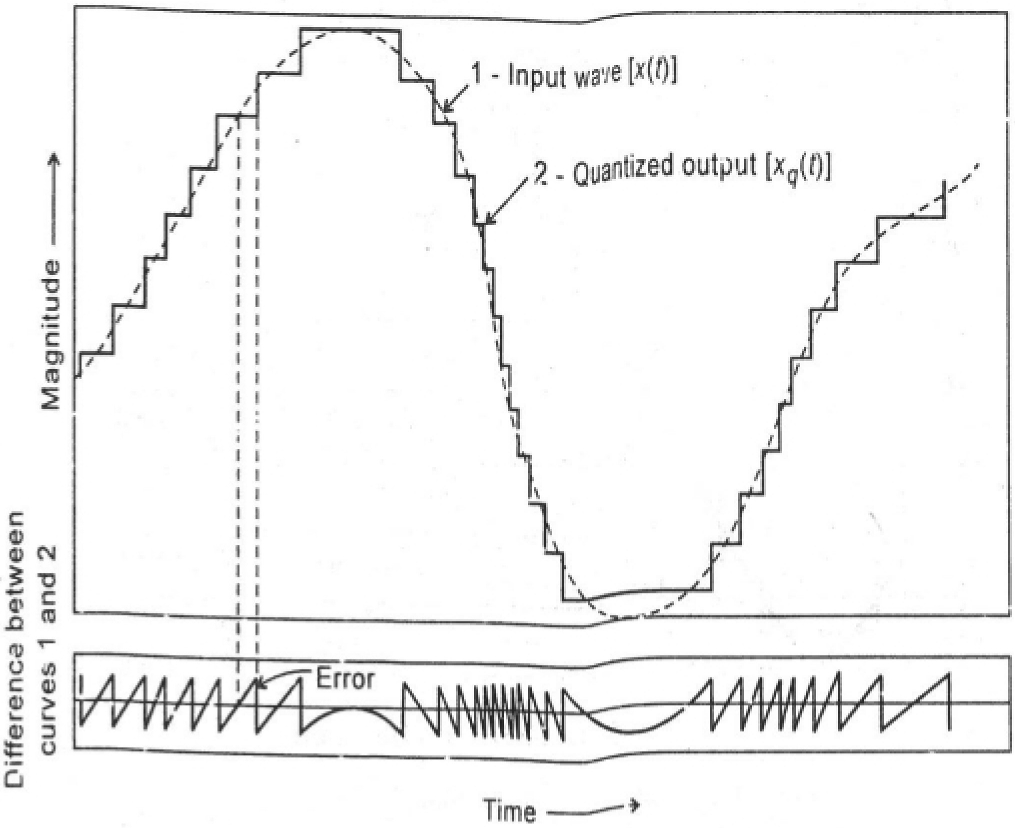


Fig. 9.1

9.2.1 Derivation of Expression for the Quantization Error

(1) In the *figure 9.2* the input signal $x(t)$ varies between the voltage levels V_H and V_L. Therefore the total variation in amplitude is given by,

$$\text{Total variation in amplitude} = V_H - V_L$$

(2) If we assume that $V_H = V$ and $V_L = -V$ then

$$\text{Total change in amplitude} = 2V \text{ volts}$$

(3) If this range is divided into Q levels of quantization then the step size is given,

$$S = \frac{V_H - V_L}{Q} = \frac{2V}{Q}$$

If we assume that $V_H = +1$ volt, $V_L = -1$ volt then $S = \dfrac{2}{Q}$.

(4) If the step size is assumed to be sufficiently small then the quantization error can be assumed to have distributed uniformly and we can say that the quantization error is a random variable with "uniform distribution".

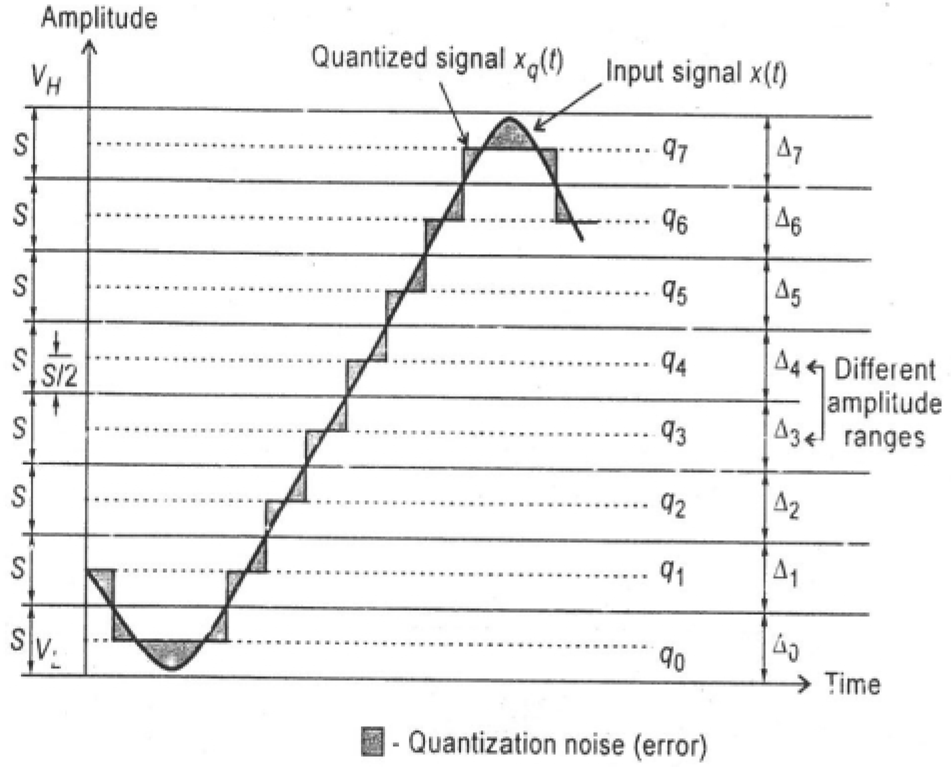


Fig. 9.2 : Process of quantization.

As already seen, the maximum quantization error is "$\pm S/2$". Therefore we can say that over the range $+S/2$ to $-S/2$, quantization error is uniformly distributed random variable.

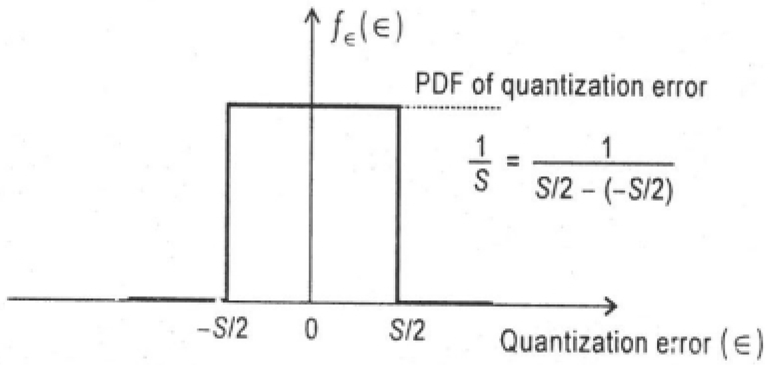


Fig. 9.3 : Uniform distribution for quantization error.

(5) The uniform distribution of quantization error is as shown in *figure 9.3*. The probability density function PDF for the quantization error "ϵ" is defined as,

$$
\begin{aligned}
f_\epsilon(\epsilon) &= 0 \quad \text{for } \epsilon \le -\frac{S}{2} \\
&= \frac{1}{S} \quad \text{for } -\frac{S}{2} \le \epsilon \le \frac{S}{2} \\
&= 0 \quad \text{for } \epsilon > \frac{S}{2}
\end{aligned}
$$

(6) The mean value or average value of the quantization error is zero.

The noise power is given by,

$$\text{Noise power} = \frac{V_n^2}{R}\ , \quad \text{where } V_n^2 = \text{Mean square noise voltage.}$$

(7) We have defined the quantization noise as a random variable, with a probability density function (PDF) equal to $f_\epsilon(\epsilon)$, we can find the mean square value of noise voltage as,

$$\text{Mean square value} = \int_{-\infty}^{\infty} \epsilon^2 f_\epsilon(\epsilon)\, d\epsilon$$

(8) But from *figure 9.3* it is clear that PDF $f_\epsilon(\epsilon)$ exists only over the range $-S/2$ to $+S/2$/. Also absolute $f_\epsilon(\epsilon) = 1/S$ and change the limits of integration to get

$$\bar{\epsilon}^2 = \int_{-S/2}^{S/2} \epsilon^2 \times \frac{1}{S}\, d\epsilon = \frac{1}{S}\left[\frac{\epsilon^3}{3}\right]_{-S/2}^{S/2} = \frac{1}{S}\left[\frac{S^3}{24} + \frac{S^3}{24}\right]$$

$$= \frac{S^2}{12}$$

$$\text{Thus,} \quad V_n^2 = \text{Mean square value of noise signal} = \frac{S^2}{12}$$

(9) If we substitute $R = 1$ ohm in equation

$$\text{Noise power} = \frac{V_n^2}{R},$$

then the noise power is called *normalized* noise power.

$$\therefore \ \text{Normalized noise power} = N_q = \frac{V_n^2}{1}$$

Normalized quantization noise power,

$$\boxed{N_q = \frac{S^2}{12}} \quad \text{; for linear quantization}$$

This is the required expression.

(10) Signal to Noise Ratio

$$\frac{S_i}{N_q} = \frac{\text{Normalized signal power}}{\text{Normalized noise power}}$$

$$S_i = \frac{(\text{rms voltage})^2}{R} = \frac{\left(\frac{V}{\sqrt{2}}\right)^2}{R} = \frac{V^2}{2R}$$

$$R = 1$$

$$\text{Normalize, } S_i = \frac{V^2}{2}$$

$$\frac{S_i}{N_q} = \frac{\frac{V^2}{2}}{\frac{S^2}{12}} = \frac{6V^2}{S^2}$$

$$\text{But } S = \frac{2V}{Q} \qquad \text{.....(Refer section 9.2.1)}$$

$$\frac{S_i}{N_q} = \frac{3}{2} Q^2 \qquad \text{Now, } Q = 2^N$$

where N is the number of bits used to represent the different quantization levels.

$$\frac{S_i}{N_q} = \frac{3}{2} 2^{2N}$$

$$\therefore \left[\frac{S_i}{N_q}\right]_{db} = 10 \log_{10}\left(\frac{3}{2} 2^{2N}\right)$$

$$= 1.76 + 6N$$

$$\doteq (1.8 + 6N) \text{ dB}$$

- Output SNR of a quantizer increases exponentially with increasing the number N of bits per sample.
- But, increase in N requires a proportionate increase in transmission bandwidth.
- Signal-to-quantization noise ratio for varying number of representation levels is given by

Number of Representation Levels, L	Number of Bits/Sample, R	Signal-to-Noise Ratio (dB)
32	5	31.8
64	6	37.8
128	7	43.8
256	8	49.8

Table 9.1

9.3 Types of Quantizers

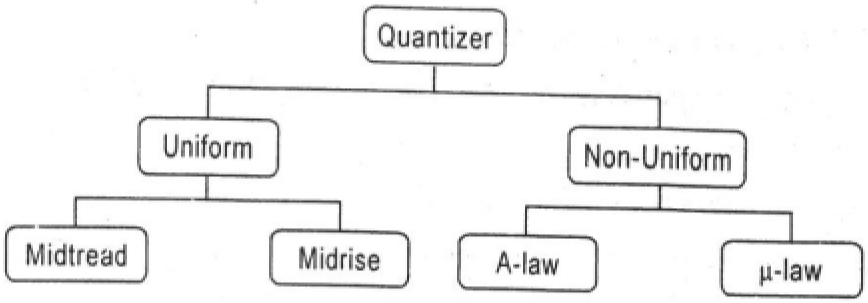


Fig. 9.4

(a) Uniform Quantizer

- A uniform quantizer is one where step size remains constant.
- Depending on the shape of quantization characteristic at the origin, we distinguish a *midtread type* and *midrise type* quantizer.

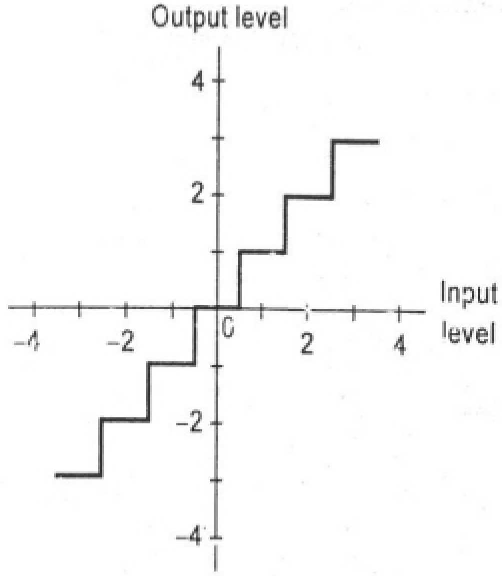


Fig. 9.5(a) : Midtread.

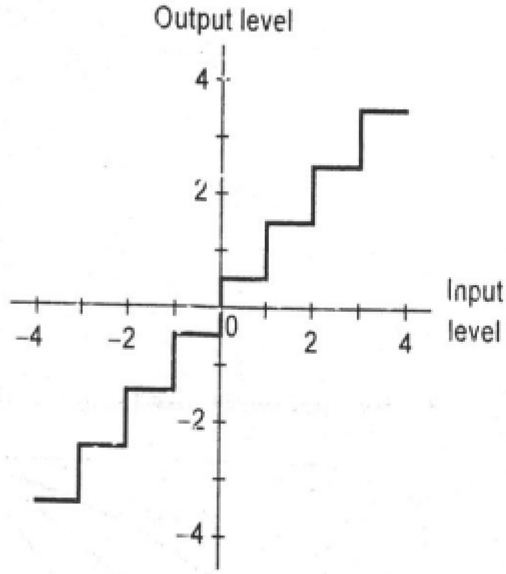


Fig. 9.5(b) : Midrise.

(b) Non Uniform Quantization and Companding

- Effectively we can say that in non uniform quantization step size is not equal. It is larger for weak signals and smaller for strong signals.
- Non uniform quantization means that the signal is passed through a *compressor* and then the compressed signal is uniformly quantized.
- The compressor has a nonlinear transfer characteristic that increases the level of weak signals.
- At the receiver in order to restore the signal samples to their correct level, an *expander* is used with a characteristic complementary to the compressor.
- Ideally, the compression and expansion laws are exactly inverse.
- As a result, the signal-to-noise ratio of quantizer becomes constant for any input signal level.

9.3.1 Companding

Q.1. Write short note on companding.

Q.2. Show how companding reduces the quantization error. Give compander characteristics.

- Companding is non-uniform quantization.
- Companding = Compression at transmitter + Expansion at receiver

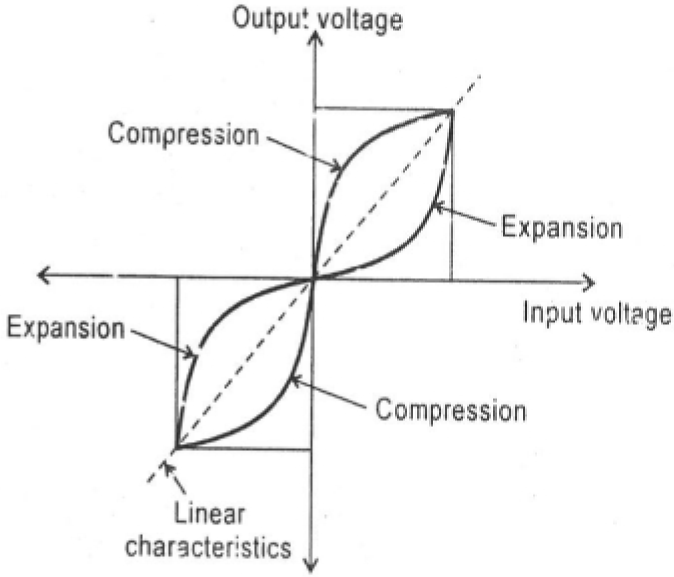


Fig. 9.6 : Companding curves for PCM.

- Two frequently used compression laws

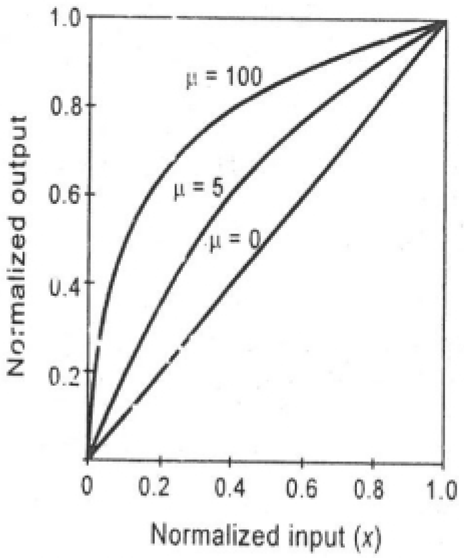


Fig. 9.7 : Compressor characteristic of a μ-law compressor.

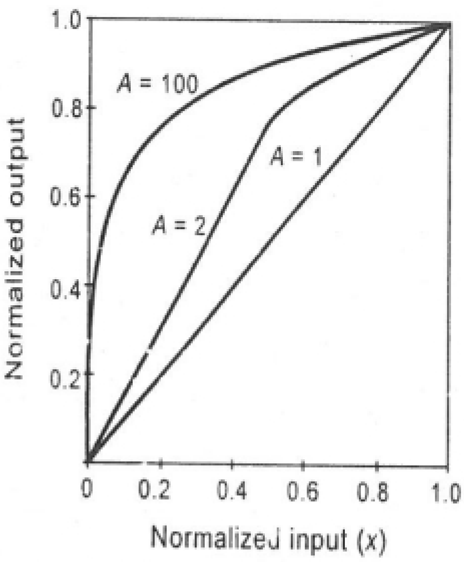


Fig. 9.8 : Compressor characteristic of a A-law compressor.

Ideally Compression Characteristics are

(1) Linear for Small Amplitude

(2) Logarithmic Elsewhere

> **Note :** *μ-law and A-law are given just for your reference. They are less important from exam point of view.*

(a) μ-law

- In the μ-law, compressor characteristics is continuous.
- Approximately linear for small amplitude.
- Logarithmic for high input level elsewhere.

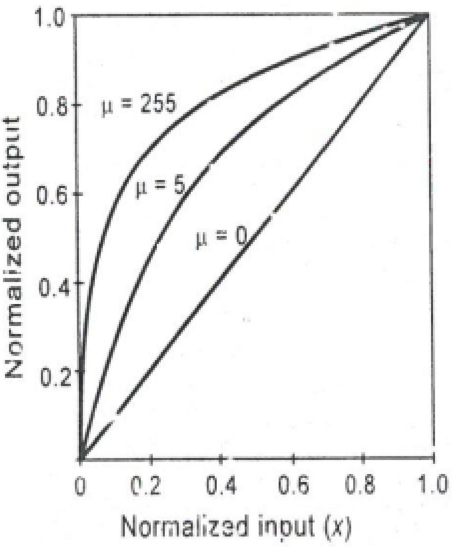


Fig. 9.9(a) : Compressor characteristic of a μ-law compressor.

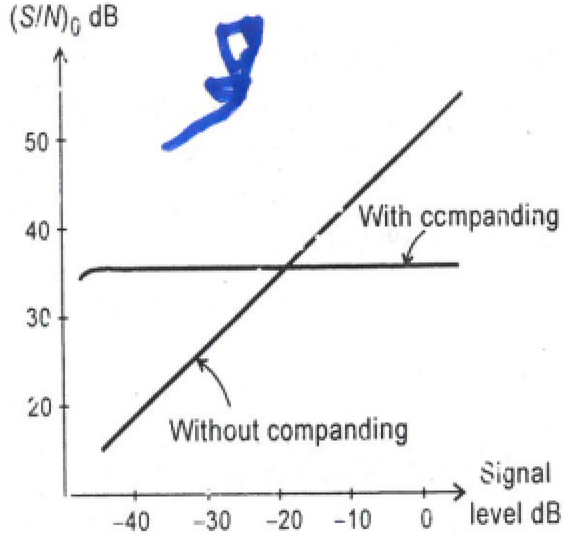


Fig. 9.9(b) : PCM performance with μ-law companding.

(b) A-law

- In A-law companding, the compressor characteristics is piecewise of linear segment or low input level and logarithmic for high input level.

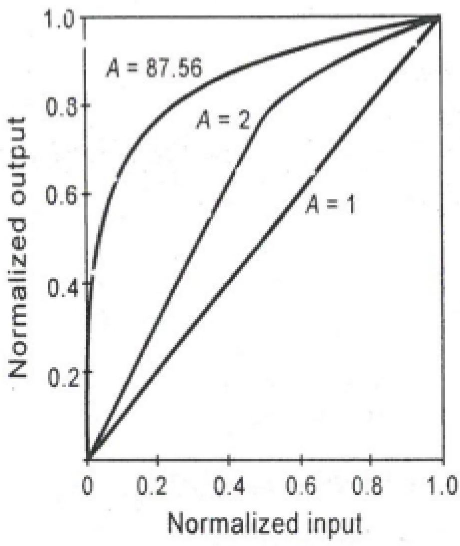


Fig. 9.10 : Compressor characteristic of a A-law compressor.

9.4 Pulse Code Modulation (PCM)

Q.1. *With a the neat sketch of waveforms and PCM system explain its working.*

Q.2. *State advantages and disadvantages of digital transmission. With neat block diagram explain the operation of single channel, simplex PCM transmission system.*

- It is a digital system.
- It converts analog input signal into digital signal before transmission.
- Digital data is transmitted in the form of pulses over the channel.
- Pulses are detected and then digital data is converted into analog signal by D/A converter at the receiver.

Block Diagram

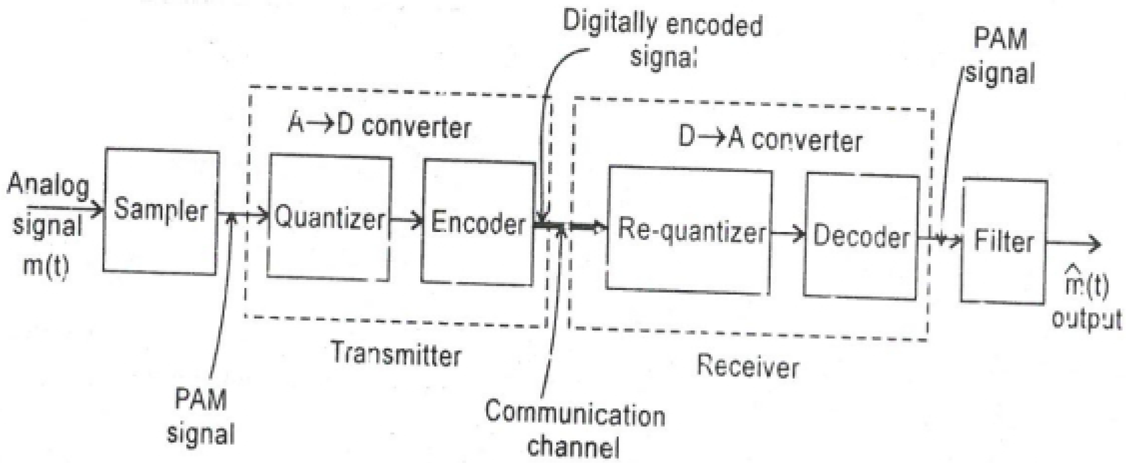


Fig. 9.11

The different blocks in above diagram are explained as follows :

9.4.1 Transmitter Section

Q. *Explain the various steps involved in the generation of PCM signal with a neat block diagram and signals.*

- Sampler circuit
- Quantizer
- Encoder

(1) Sampler

- The analog signal is converted into different pulses.
- Amplitude of the pulses varies according to the modulating signal (PAM).
- Sampling allows time division multiplexing of different signals.

(2) Qantization

- By quantizing the signal $m(t)$ we create a new signal $m_q(t)$ which is the approximation to original analog signal $m(t)$.
- Operation of quantization is explained further :

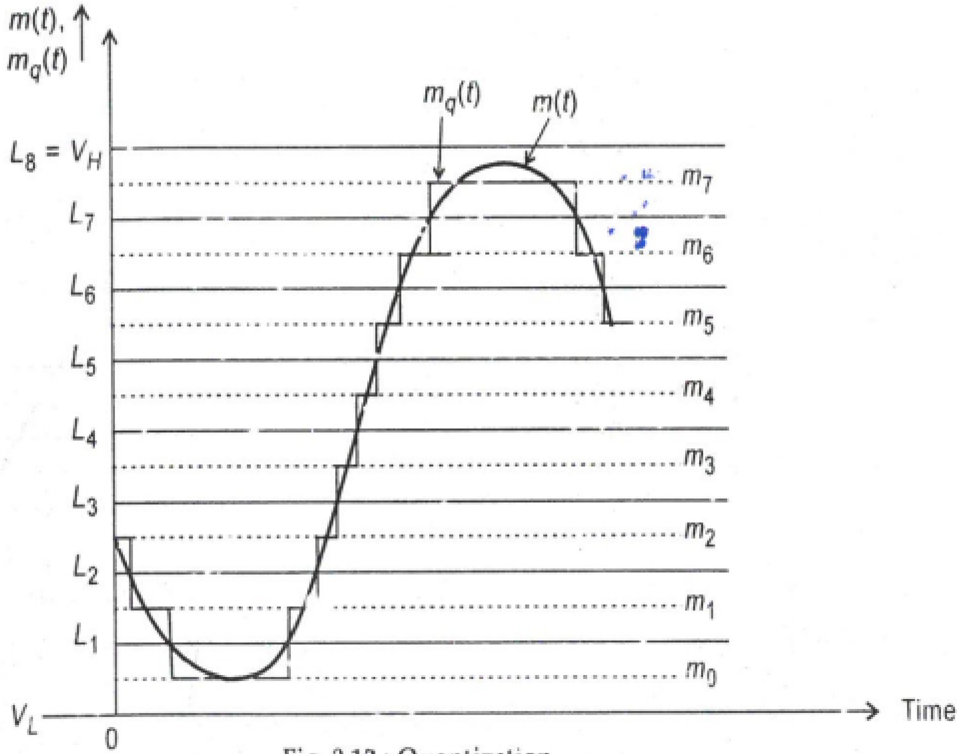


Fig. 9.12 : Quantization.

- As shown in the *figure 9.12* the signal varies between V_H and V_L. The total range is divided into $m = 8$ different intervals each of step size S.

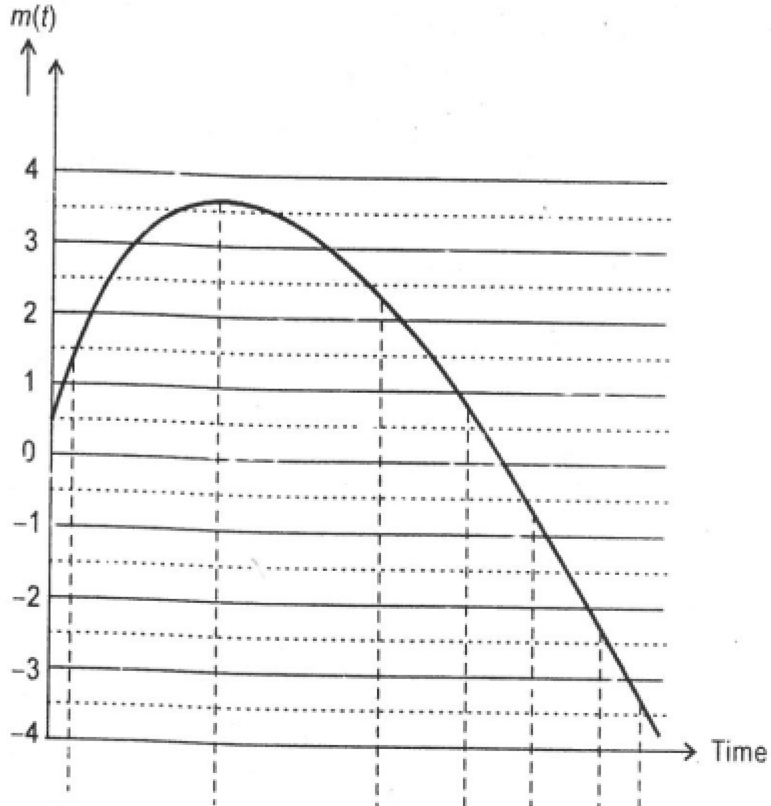
$$S = \frac{V_H - V_L}{8}$$

- The quantization levels are $m_0, m_1, m_2, \ldots\ldots, m_7$. As shown in *figure 9.12*, $m_q(t)$ is close approximation to $m(t)$. The quality of approximation can be improved by reducing the step size 'S'; hence increasing number of levels.
- Around 256 levels can be used to obtain the quality signal in commercial TV signal transmission.
- The difference between the original signal $m(t)$ and its approximation $m_q(t)$ is called as *Quantization error*. The quantization error can be reduced by increasing number of steps or reducing step size.

(3) Encoder

- The encoder is used to code the different quantization levels into digital data as follows :

Code no.	Quantization levels
7	3.5
6	2.5
5	1.5
4	0.5
3	–0.5
2	–1.5
1	–2.5
0	–3.5



Sample value	1.3	3.6	2.3	0.7	–0.7	–2.4	–3.4
Nearest quantization level	1.5	3.5	2.5	0.5	–0.5	–2.5	–3.5
Code number	5	7	6	4	3	1	0
Binary representation	101	111	110	100	011	001	000

Fig. 9.13 : Quantization.

- Thus the encoder converts the standard quantization levels into Binary code. The encoded data is transmitted in the form of pulses or voltage levels as shown below.

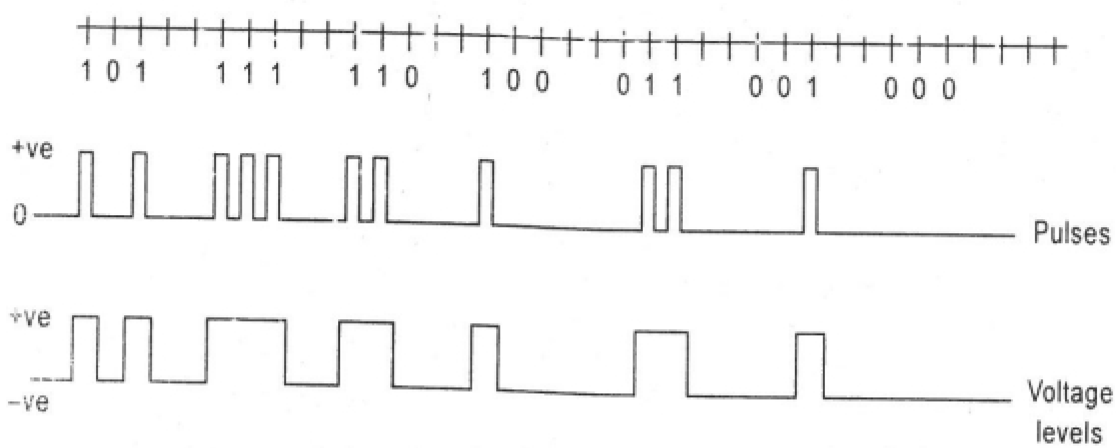


Fig. 9.14

9.4.2 Receiver Section

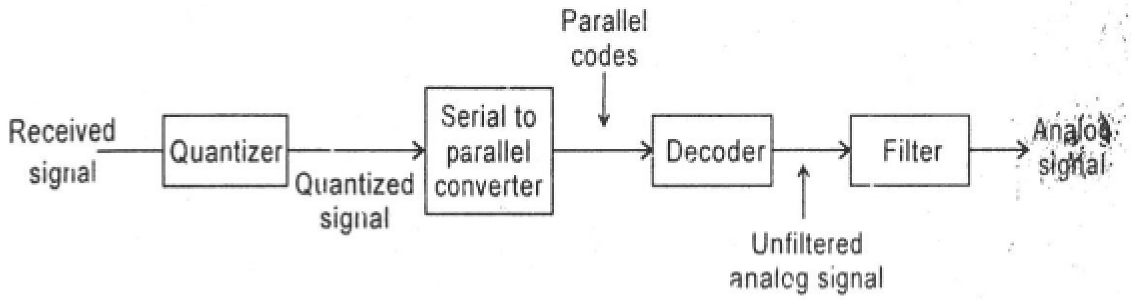


Fig. 9.15

- Quantizer (2 - level)
- Serial to parallel converter
- Decoder
- Filter

(1) Quantization

Quantization is done to remove noise added through the channel.

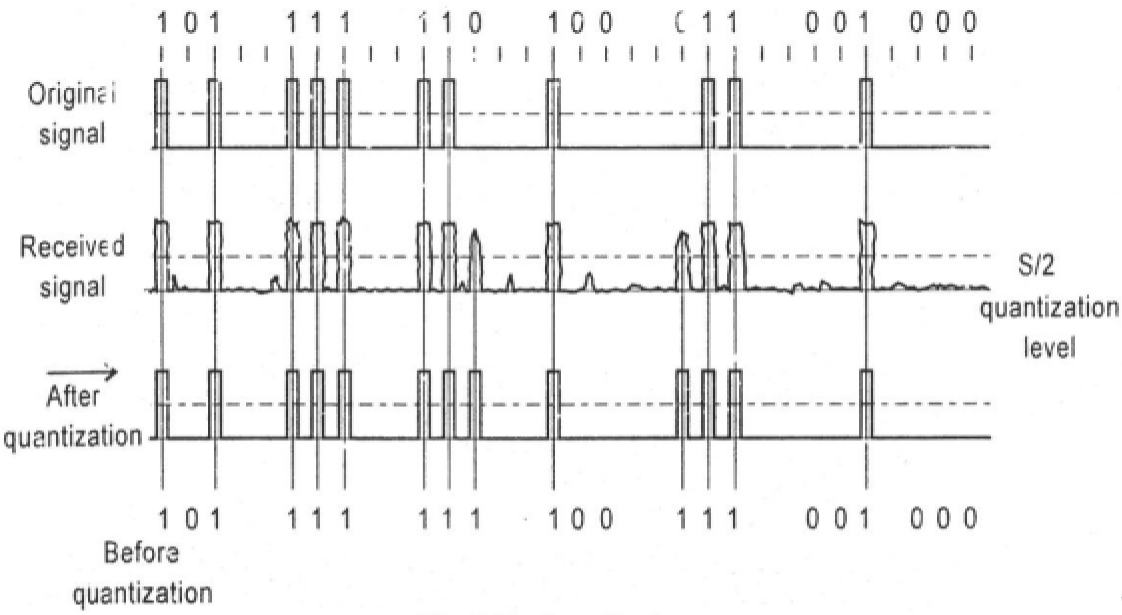


Fig. 9.16 : Quantization.

- The *figure 9.16* shows that received signal is requantized into two different levels by removing channel noise.
- If noise is greater than $(S/2)$, it is not removed by quantizer.
- The probability of error occurring due to noise can be reduced by increasing step size; but in this case, quantization error increases.

  Hence the received signal is not a perfect replica of the transmitted signals.

(2) Serial to Parallel Converter

It converts the sequence of codes to N digit PCM.

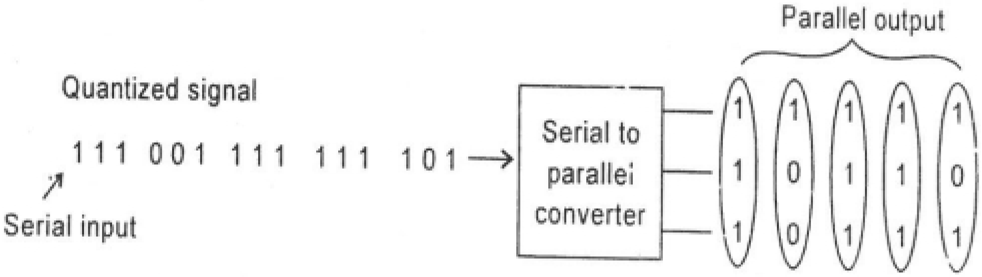


Fig. 9.17 : Serial to parallel converter.

(3) Decoder

The decoder performs the inverse operation of encoder. The coded signal arrived at the receiver is again converted into quantized sample values.

Thus quantized PAM signal is obtained it is called as *digital to analog* (D/A) conversion.

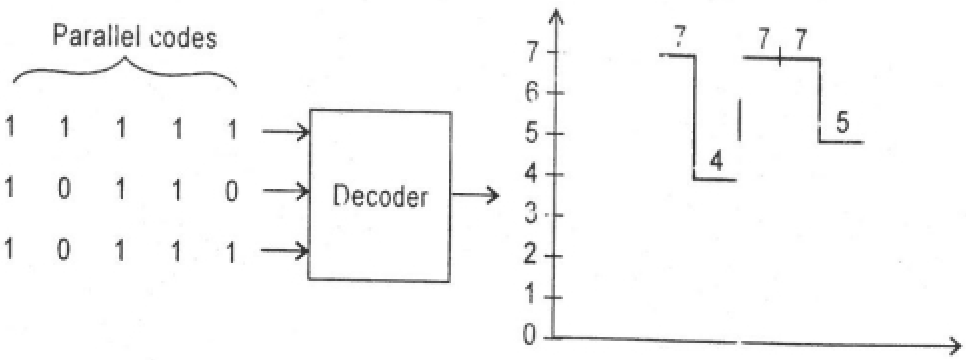


Fig. 9.18 : Decoder.

(4) Filter

The quantized PAM signal is passed through '*low pass filter*' to obtian the final output $m'(t)$ which is approximately identical to the input signal $m(t)$ at the transmitter.

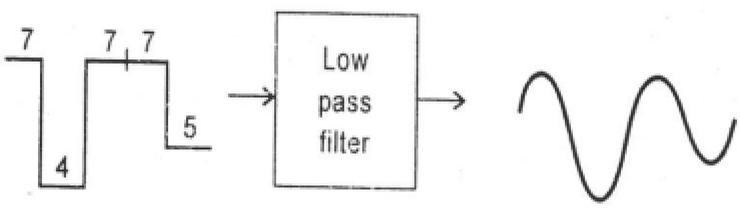


Fig. 9.19 : Filter.

9.4.3 Advantage of PCM Over Analog Transmission

(1) Noise Immunity

- PCM is digital system and almost free from effect of noise in the channel.
- Transmission is in the form of group of pulses which indicates the signal amplitude.
- At the receiver we have to detect only the presence of the pulse, irrespective of its amplitude and shape.

Hence the PCM system is free from channel noise.

The only error in PCM is *quantization error* which can be reduced by increasing number of quantization level or reducing step size.

(2) Inbuilt Secrecy

In PCM the modulating signal is converted into digital form before transmission. The digital data can be coded into any form.

The code is transmitted in the form of pulses.

Hence, by designing the complex codes, which only few people can decode, secret messages can be transmitted over the channel.

(3) Multiplexing of Signals

By sampling analog signals one after another and converting each sample to a group of pulses, it is possible to send many signals over the channel by using Time Division Multiplexing (TDM).

(4) Regeneration of Digital Signals

In case of analog signals transmitted over telephone cable, the quality of signal deteriorates over a few km as the power reduces due to resistance of the wire. Hence repeaters should be inserted every few kilometers to increase power of the signal.

Cross talk disturbance is more in analog transmission.

In PCM, as the signal is transmitted in the form of pulses representing certain codes, the receiver has to detect the presence of the pulse, shape or position or width of pulse.

Hence PCM is far better than other analog systems like PAM, PPM and PWM.

Hence it is used in :

(i) City telephones, ISD systems.

(ii) Music systems and compact disks (CDs).

(iii) Military communication.

(iv) Digital TV transmission.

9.5 DPCM (Differential Pulse Code Modulation)

Q. Compare PCM and DPCM.

Block Diagram

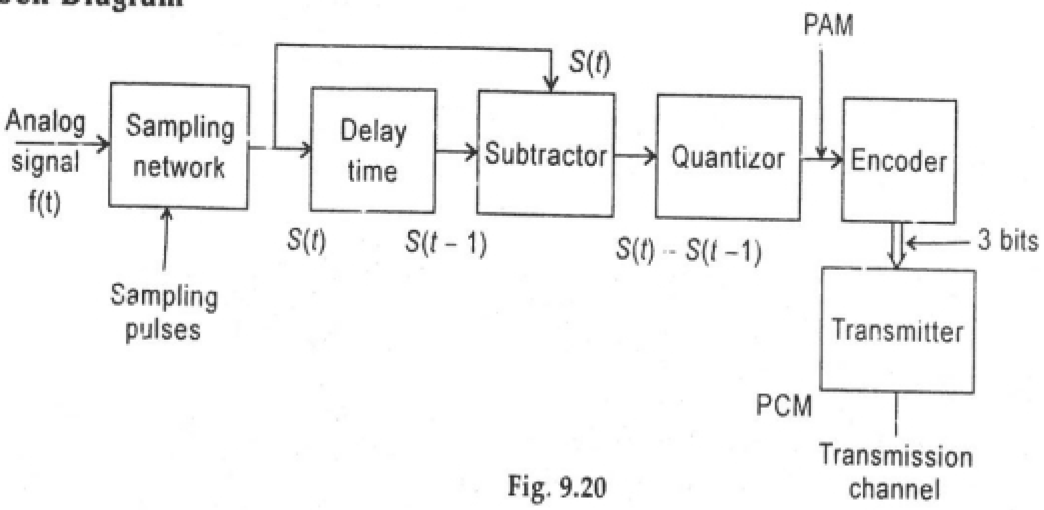


Fig. 9.20

- In case of PCM, if amplitude of analog signal is increased then number of required quantization levels will be more and hence we need more number of bits to transmit it.
- This requires large bandwidth.
- This drawback can be rectified by transmitting difference between 2 adjacent samples instead of transmitting sampled amplitudes.
- This requires less bits and bandwidth reduces drastically.

Advantages DPCM

- Transmission bit rate is reduced.
- Signal to noise ratio is increased. Hence better reception than PCM.

Drawbacks

- DPCM is optimised for voice signals and not applicable with data signals.
- DPCM is more affected by noise compared to PCM.

Comparison Between PCM and DPCM

	PCM	DPCM
(i) Number of quantization levels (Q)	64	8
(ii) Code words	8 bits	3 bits
(iii) Bandwidth for transmission	$8(f_m)$	$3(f_m)$

where f_m = Modulating frequency

Table 9.2

9.5.1 Delta Modulation (LDM)

Q.1. Draw a neat block diagram of delta modulator. Explain the working with waveforms at output at each block.

Q.2. Delta modulation diagram working and limitations.

Q.3. Write a short note on Delta Modulation.

Q.4. Explain : Delta Modulation

- Delta Modulation is a modulation technique in which the difference $m(t) - \hat{m}(t)$ is encoded into just a single bit. Refer the *figure 9.21*.
- A single bit (providing just 2 possibilities) is used to increase (+ve) or decrease (−ve) the estimate of the original signal.

Block Diagram

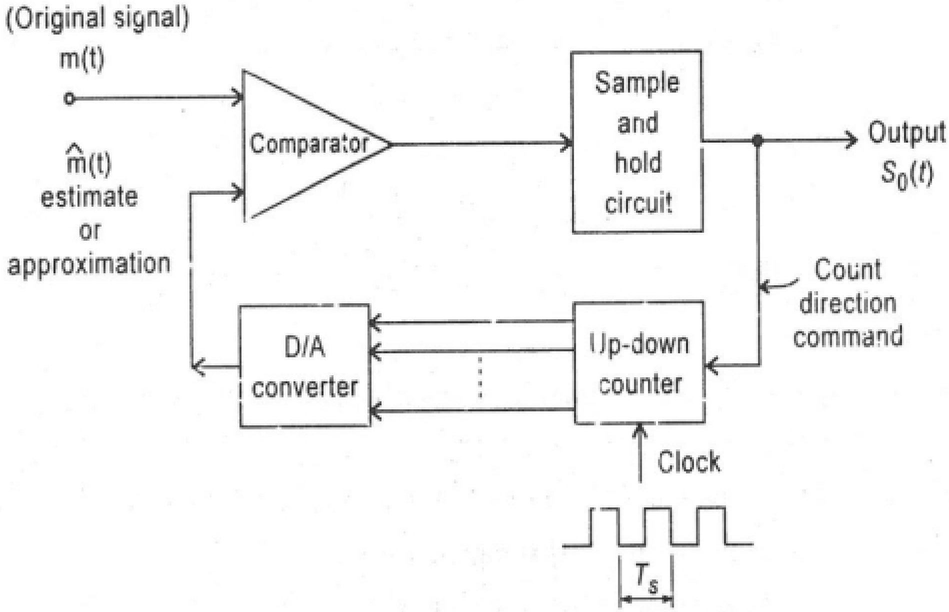


Fig. 9.21(a) : Delta modulator.

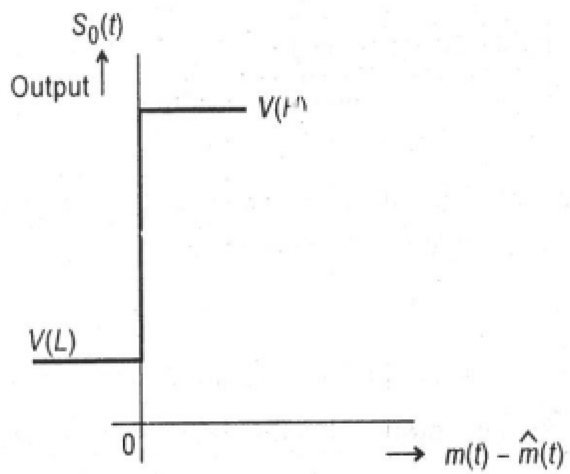


Fig. 9.21(b)

Waveform

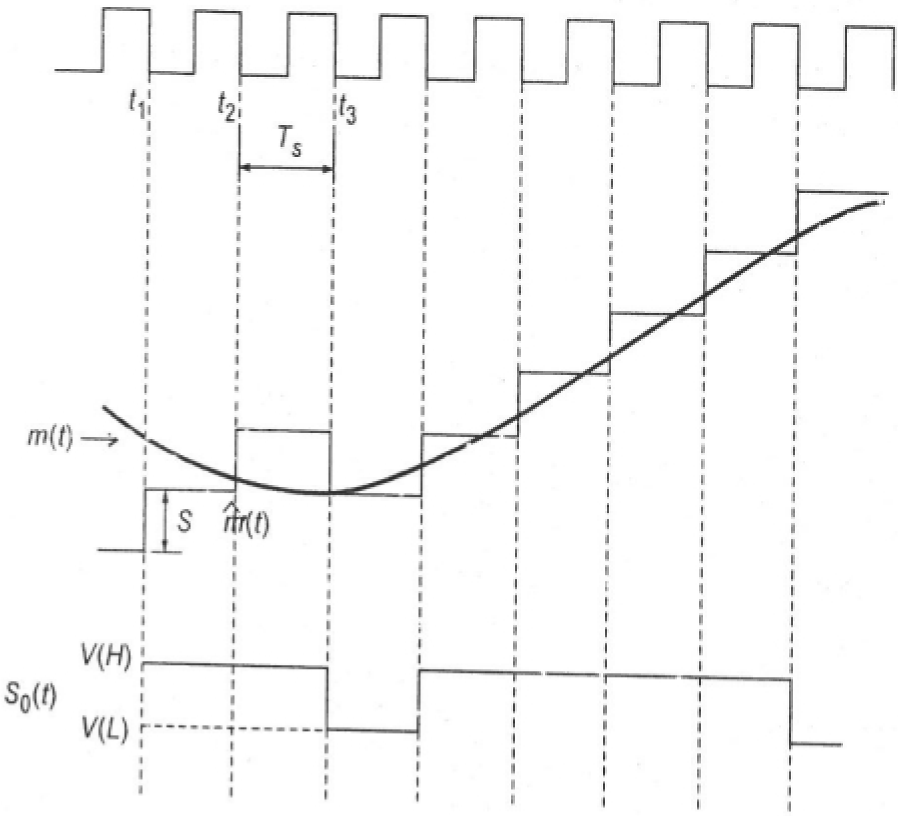


Fig. 9.22

Functions of Different Blocks

(1) Comparator

The modulating signal $m(t)$ and its quantized approximation $\hat{m}(t)$ are inputs to the comparator.

Comparator compares the two input levels and produces difference output.

(i) when $m(t) > \hat{m}(t)$, then output $S_0(t)$ is high i.e. $V(H)$

(ii) when $m(t) < \hat{m}(t)$, then output $S_0(t)$ is low i.e. $V(L)$

(2) Sample and Hold Circuit

- The difference signal produced by comparator is analog signal which is converted into digital by analog to digital (A/D) convertor, i.e. sample and hold circuit.
- The output of the circuit $S_0(t)$ is either high or low depending upon difference between the input signal and quantized signal.

(3) Up-down Counter

The counter counts up or down depending upon count direction command.

(i) $m(t) > \hat{m}(t)$, then the counter counts up at the trailing edge of clock pulse. Hence signal $\hat{m}(t)$ jumps upwards through step size 'S'.

(ii) if $m(t) < \hat{m}(t)$, then counter counts down at the trailing edge of clock pulse. Hence signal $\hat{m}(t)$ jumps down through step size 'S'.

(4) D-A Converter

This block is used to generate quantized analog approximation of original signal $\hat{m}(t)$ from digital data.

9.5.2 Problems with Delta Modulation

(1) Hunting

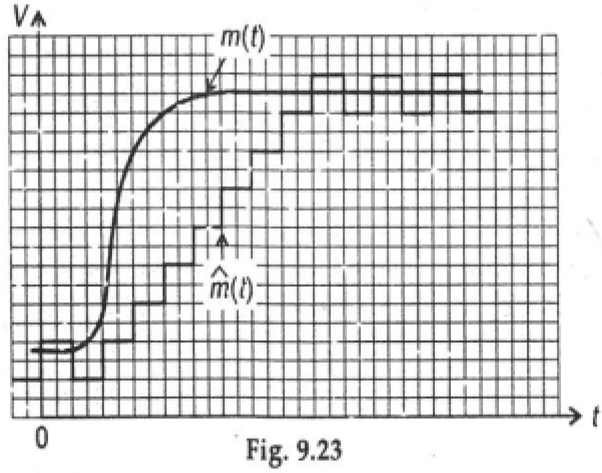


Fig. 9.23

From the waveforms of $m(t)$ and $\hat{m}(t)$ we can see that, $\hat{m}(t)$ tries to catch up with $m(t)$, but it never succeeds.

Initially there is more error between the two input signals but as $\hat{m}(t)$ approaches $m(t)$, the quantization error decreases. Even if $\hat{m}(t)$ catches up with $m(t)$, then also it keeps on hunting around $m(t)$.

This is also known as ***Granular noise***.

(2) Slope Overload Error

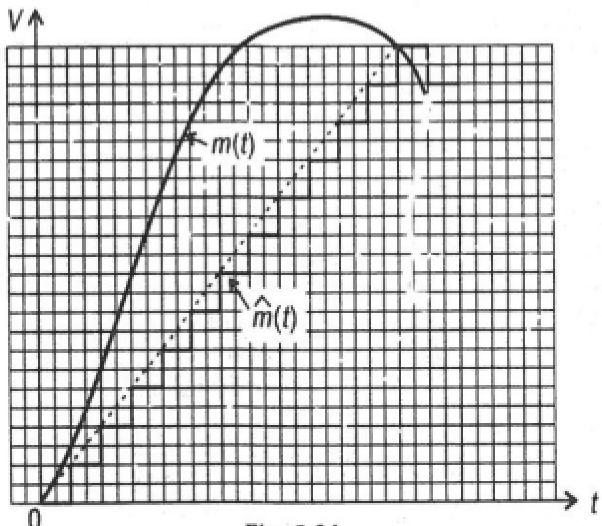


Fig. 9.24

- As shown in *figure 9.24*, $m(t)$ has *large slope extending over more time*. Hence $\hat{m}(t)$ can not keep up with it. Hence, the error $\hat{m}(t) - m(t)$ goes on increasing. The difference between $\hat{m}(t)$ and $m(t)$ is called as *slope overload error*.

- To avoid slope overload error, Adaptive Delta Modulation (ADM) is used in which step size gradually increases, whenever slope-overload occurs.
- The linear form of $\hat{m}(t)$ justifies the name Linear Delta Modulation (LDM)

9.6 Adaptive Delta Modulation (ADM)

Q.1. Sketch block diagram of Adaptive Delta Modulator and explain the working. How is the step size calculated.

Q.2. How is adaptive delta modulation better than linear delta modulation ? Draw the block diagram of adaptive delta modulator and explain each block in detail. Also, explain how the step size is determined.

Block Diagram

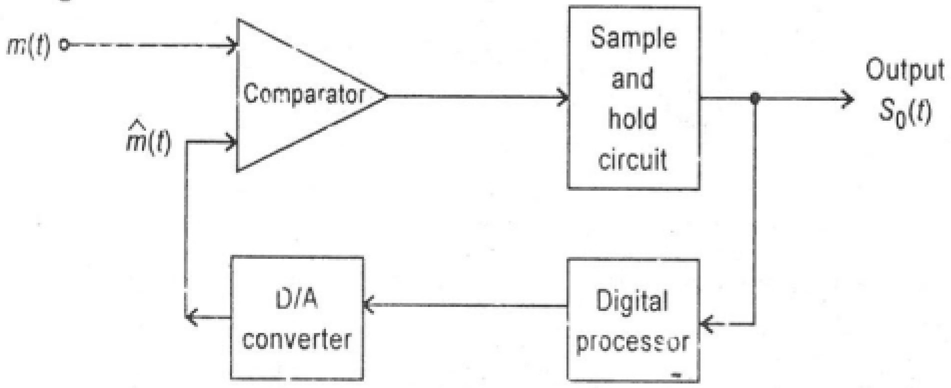


Fig. 9.25 : Adaptive delta modulator.

Waveform

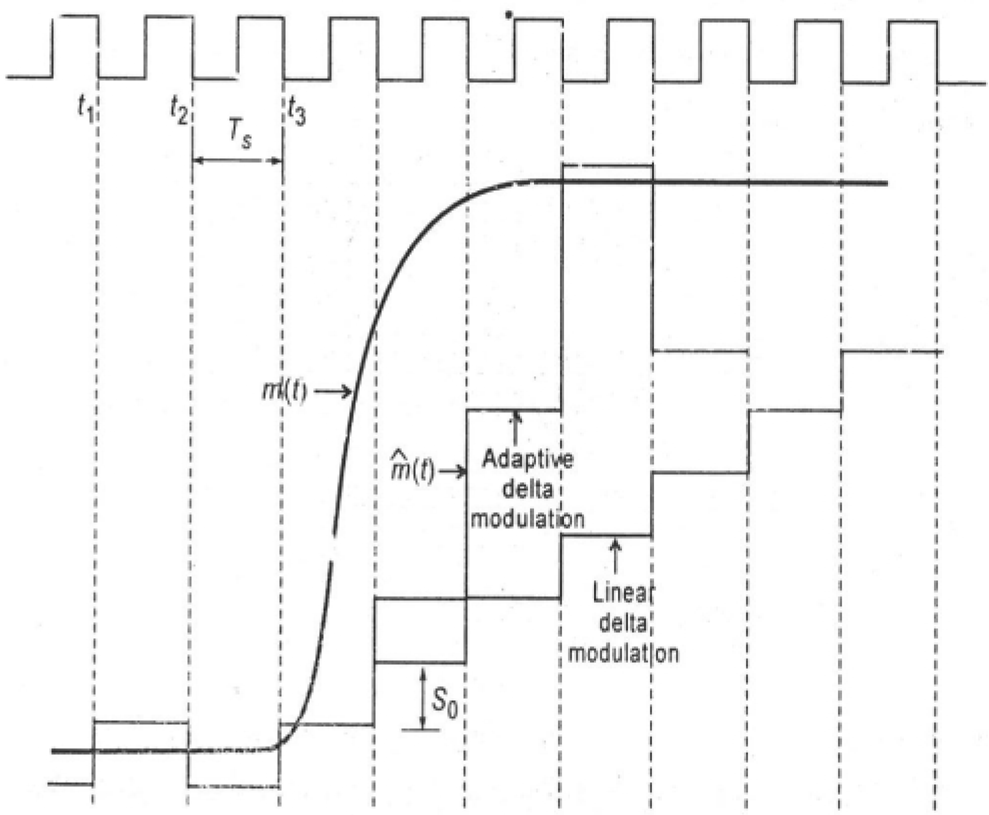


Fig. 9.26

- This technique is modification over linear delta modulation to reduce slope-overload error.
- In linear delta modulation, step size is fixed, hence $\hat{m}(t)$ approaches towards modulating signal $m(t)$ linearly, but in ADM, step size becomes progressively large to reduce the slope-overload error.

Functions of Different Blocks

(1) Comparator

The modulating or baseband signal $m(t)$ and its quantized approximation $\hat{m}(t)$ are inputs of comparator. It compares the two input levels and produces difference output, which is error between the two input signals.

The output of the circuit $S_0(t)$ is either high or low depending upon difference between $m(t)$ and $\hat{m}(t)$.

(2) Sample and Hold Circuit

The difference signal produced by the comparator is analog signal which is converted into digital by A/D converter i.e. sample and hold circuit.

if $m(t) > \hat{m}(t)$, the output $S_0(t)$ is high i.e. $V(H)$

if $m(t) < \hat{m}(t)$, the output $S_0(t)$ is low i.e. $V(L)$

(3) Digital Processor

The processor has an accumulator and at each active pulse (trailing edge) generates a step size 'S' which augments or diminishes the accumulator. The step size is variable and multiple of basic step size S_0.

In response to k^{th} active clock edge, the processor generates a step equal in amplitude to the step generated in response to $(k-1)^{th}$ clock pulse.

if $m(t) > \hat{m}(t)$, then this step size is increased by the accumulator by amount S_0, otherwise decreases the step size by amount S_0.

As shown in the waveform, $\hat{m}(t)$ catches upto $m(t)$ very fast compared to linear delta modulation.

(4) D/A Convertor

This block is used to generate quantized analog approximation of original signal $\hat{m}(t)$ from digital data.

Disadvantage

- Adaptive delta modulation reduces the slope-overload error compared to linear delta modulation but increases quantization error.

9.7 Comparisons

Q. Difference between Pulse Code Modulation and Delta Modulation.

Sr.	PCM (Pulse Code Modulation)	DM (Delta Modulation)
(1)	PCM transmits amplitude of analog signal in digitally coded form.	Delta modulation does not convey any information about actual value of the sample.
(2)	The data is transmitted in the form of no. of coded bits (8 bits)	Data is transmitted in the form of single bit indicating slope of signal either +ve or −ve.
(3)	For reproducing original data, sampling frequency should not be very large. system can be practically implemented.	The sampling frequency should be large compared to PCM. Hence system is not practicable. Even if bandwidth is less (1 bit), the benefit can not be practically implemented.
(4)	More complex than DM.	Simple to implement.

Table 9.3 : Comparison between PCM and DM.

Sr.	LDM (Linear Delta Modulation)	ADM (Adaptive Delta Modulation)
(1)	The step size by which the approximation $\hat{m}(t)$ tries to catch up with $m(t)$ (actual signal) is fixed.	The step size goes on changing according to requirement to avoid slope-overload error.
(2)	Quantization error is less but slope overload error is more.	Slope overload error is less but quantization error is more.

Table 9.4 : Comparison between LDM and ADM.

Sr.	PCM (Digital)	PAM (Analog)
(1)	Certain levels of signals are quantized and then transmitted in coded form by using pulses. Hence it is digital system. The code can be binary, gray or any other.	The actual value of sample (amplitude) is transmitted. Hence it is analog system.
(2)	Bandwidth required is more because each PAM sample is converted into many pulses of PCM.	Bandwidth required is not large.

(3)	Channel noise can be reduced by requantization. The system is less noisy.	The channel noise can distort the amplitude of pulse. The distortion appears in final output. Hence it is more noisy.
(4)	In PCM, the receiver has to detect only the presence or absence of pulse. Hence even if pulse is distorted, it does not affect the output. System is noise free. Signal quality does not deteriorate with distance.	Signal quality deteriorates with distance. Hence repeaters (amplifiers) are required over the channel to reamplify the loss of signal.
(5)	Expensive system.	Inexpensive system.

Table 9.5 : Comparison between PCM and PAM.

Q. Compare Analog and digital modulation.

Sr.	Digital Modulation	Analog Modulation
(1)	Transmitted signal is digital in nature.	Transmitted signal is analog in nature.
(2)	Amplitude, width or position of transmitted pulses is constant.	Amplitude, frequency and phase of transmitted signal varies.
(3)	Information is transmitted in the form of code words.	Information is transmitted as variations in any property or attribute (amplitude, phase or frequency of signal).
(4)	Coding techniques are used to detect and correct errors.	No such concept for error detection or correction.
(5)	Noise immunity is very high.	Noise immunity is poor for AM but slightly better for FM and PM
(6)	Repeaters can be used, as we can separate signal from noise.	Repeaters cannot be used as separation of signal and noise is impossible.
(7)	High channel bandwidth is needed due to high bit rates.	Lower bandwidth required.

(8)	Time Division Multiplexing (TDM) is used for Multiplexing digital signals.	Frequency division Multiplexing (FDM) is used for multiplexing analog signals.
(9)	It is suitable for military applications and sending of secret messages as we use coding techniques which are reliable to detect and correct any error.	It is not suitable for transmission of secret information in military application.
(10)	**Examples :** PCM (Pulse Code Modulation) DM (Delta Modulation) ADM (Adaptive Delta Modulation) DPCM (Differential PCM)	**Examples :** AM (Amplitude Modulation) FM (frequency Modulation) PM)Phase Modulation) PAM, PWM, etc.

Table 9.6 : Comparison between Digital and Analog Modulation.

9.8 Frequently Asked Questions

Q.1. ***Differentiate between Pulse modulation Techniques and Digital Pulse Modulation Techniques. Why is PCM more resistant than other form of pulse modulation ?***

Ans. Comparison from section 9.7.

PCM is more noise resistant because in PCM information is not stored in width, position or amplitude of pulse. PCM receiver we just have to detect there is pulse or not hence more noise resistant.

Q.2. ***What are the drawbacks of delta modulation ? How can they be minimized ? Explain slope overload error and hunting error in delta modulation. Derive the condition to avoid slope overload error.***

Ans. Explain slope overload error and hunting error from section 9.5.2 and explain adaptive delta modulation from section 9.6.

Q.3. ***What are advantages and disadvantages of digital modulation ? Also draw the block diagram of PCM and explain it. Why is PCM more noise resistant than other form of pulse modulation ?***

Ans. Advantages and disadvantages of digital modulation from Q.5. in FAQs.

Refer PCM section 9.4.

PCM is more noise resistant because in PCM information is not stored in width, position or amplitude of pulse. In PCM at receiver, we just have to detect there is pulse or not hence more noise resistant.

Q.4. *How is adaptive delta modulation better than linear delta modulation ? Draw the block diagram of Delta Modulation and explain each block in detail.*

Ans. Explain slope overload error and hunting error from section 9.5.2 and explain adaptive delta modulation from section 9.6.

Q.5. *Advantages and disadvantages of digital modulation.*

Ans. Advantages

(1) More immune to noise.

(2) As it is digital form, encryption is easy.

(3) Line codes can be applied for noise immunization.

(4) Error detection and correction is easy.

Disadvantages

(1) More hardware is required.

(2) Cost is more.

(3) More complex.

(4) Usually synchronization is required between transmitter and receiver.

Q.6. *Draw neat block diagram of delta modulator and explain its working. What are drawbacks of delta modulator and how are they overcome by ADM ?*

Ans. Delta modulation - refer section 9.5.1.

Error - refer section 9.5.2.

ADM - refer section 9.6.

Q.7. *If voice signal is PCM encoded, using 8 bit PCM, what is the output bit rate ?*

Ans. The voice signal is always sampled at 8 kHz.

$\therefore$ Sampling frequency = f_s = 8 kHz

i.e. we get 8000 samples/sec.

Now, each sample is encoded using 8 bits

$\therefore$ Output bit rate $= 8000 \times 8$

$= 16$ kbps

Q.8. State disadvantages of Uniform Quantization in PCM. What is the remedy ?

Ans. Refer section 9.3 (b) and section 9.3.2.

Q.9. Advantages of Digital Communication.

Ans. Refer section 9.4.3.

Q.10. Compare Analog and Digital communication.

Ans. Refer section 9.4.3.

❋❋❋

e remedy ?

***

10 MULTIPLEXING

Topic	Theory imp	Oral imp
TDM	☆☆	☆☆
FDM	☆☆	☆☆
Comparison	☆☆☆☆	☆☆☆☆
FAQ's	☆☆☆☆	☆

10.1 Time Division Multiplexing (TDM)

Q. *What is multiplexing in communication system ? Draw the block diagram of a TDM-PAM system to transmit four voice channel. Sketch one frame of transmitted signal.*

- According to sampling theorem a signal is uniquely specified by its values at intervals $(1/2f_m)$ second ; where f_m is frequency of modulating signals.
- At receivers the complete signal can be reconstructed from the knowledge of the signal at these instants alone.
- We therefore need to transmit only the samples of the signals at these instants and rest of time is idle period.
- During this idle period we may transmit the samples of other signals.
- We can thus interweave the samples of several signals on the channel.
- At the receiving end, these samples can be separated by a proper synchronous detector. This is known as *time division multiplexing*.

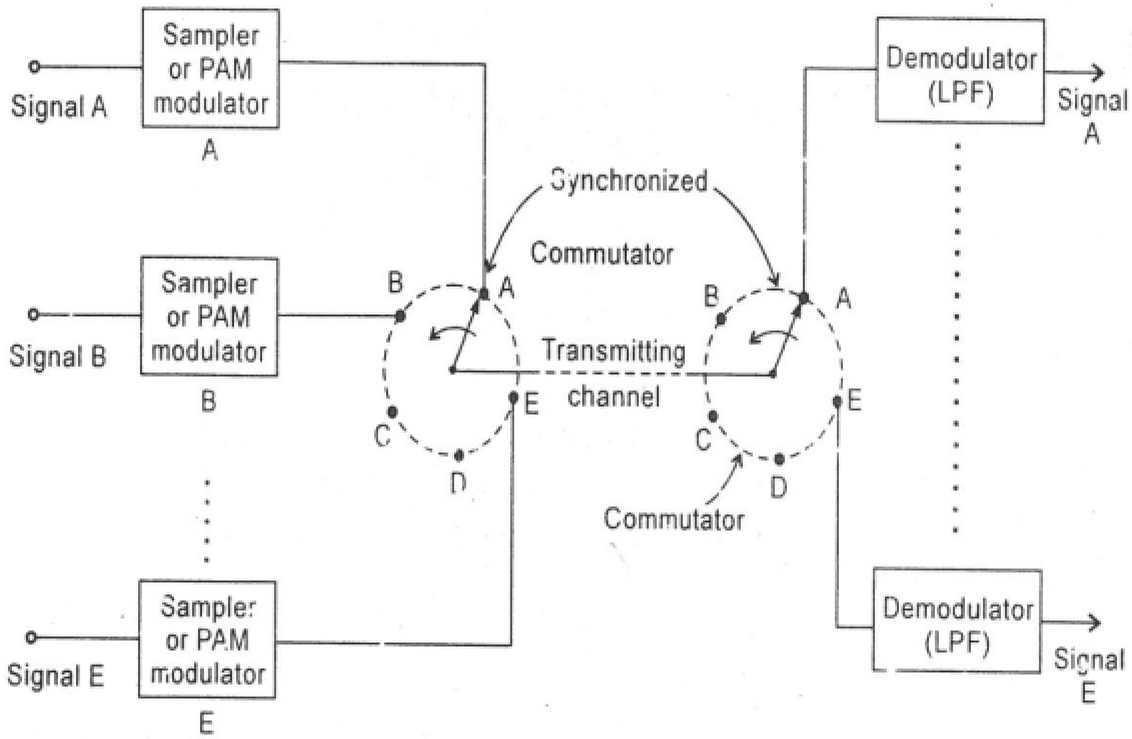


Fig. 10.1

10.2 Frequency Division Multiplexing (FDM)

Transmission of one signal at a time on a channel is a highly wasteful situation. This difficulty can be overcome if we can shift the frequency spectrum of various signals so that they occupy different frequency ranges without overlapping. This shifting is possible with modulation.

Therefore it is possible to transmit a large number of signals at the same time on one channel by using modulation techniques.

This *process of shifting signals to different frequency range* is known as frequency Division Multiplexing (FDM).

At receiving end the various signals can be separated by using appropriate filters. The frequency shifting or translation of many signals is not done in one step (This is for the sake of flexibility, economy and simplicity). Instead standardized grouping of channels are used and several steps of frequency translation takes place before all the channels have been placed in their location in frequency spectrum that is transmitted in a particular link.

Group Formation

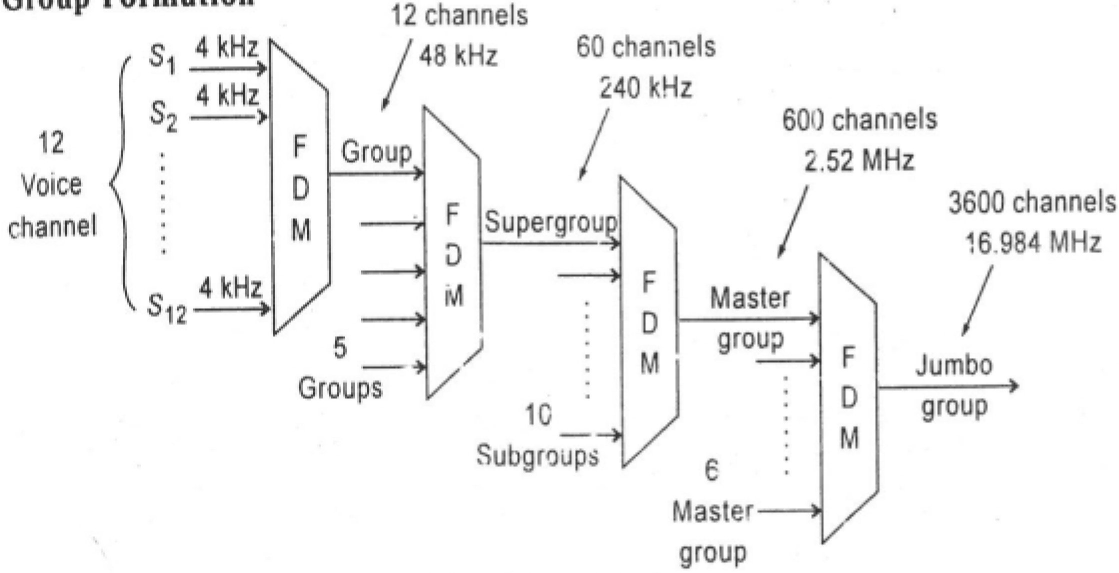


Fig. 10.2 : FDM hierarchy.

> **Note :** *Group formation is less important from exam point of view. It is just for your reference.*

> **Note :** *Multiplexing explained in chapter 1 is FDM.*

Following block diagram shows basic idea to form group of 3 channels of 0 - 4 kHz.

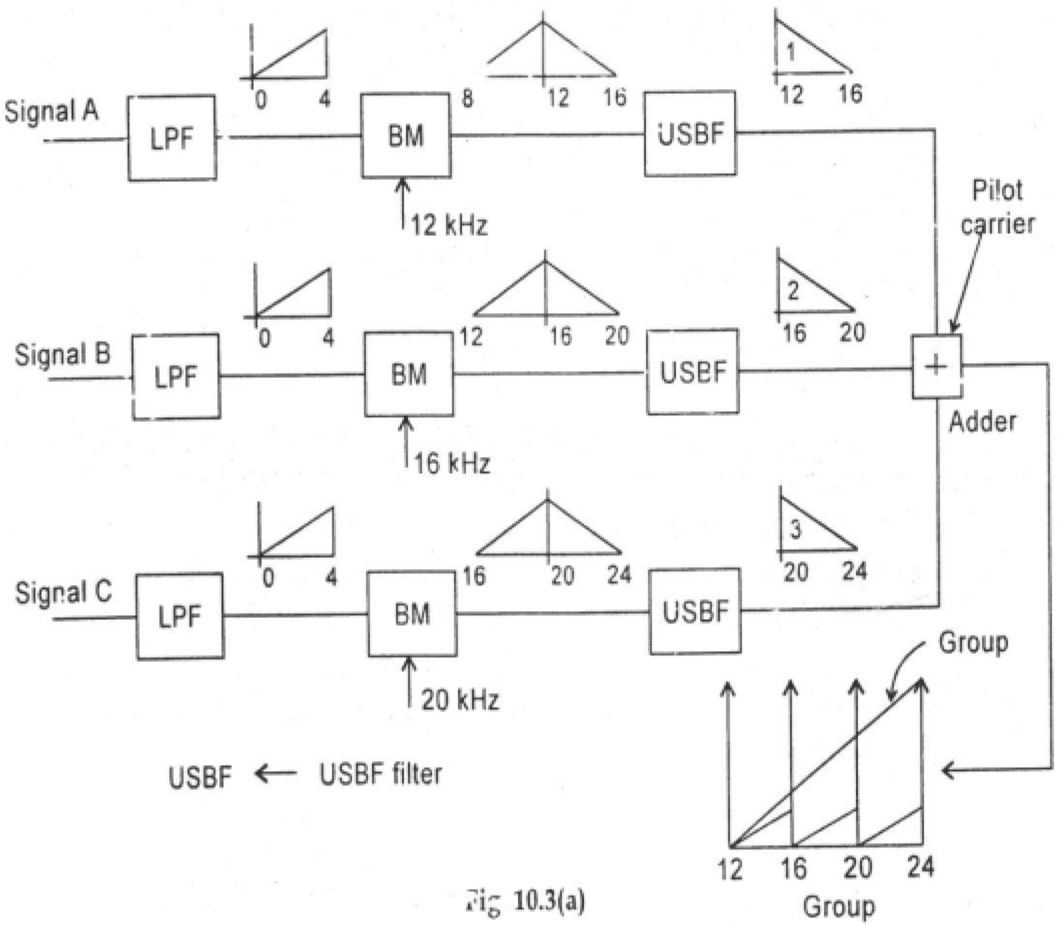


Fig 10.3(a)

Recovery at receiver is done as follows

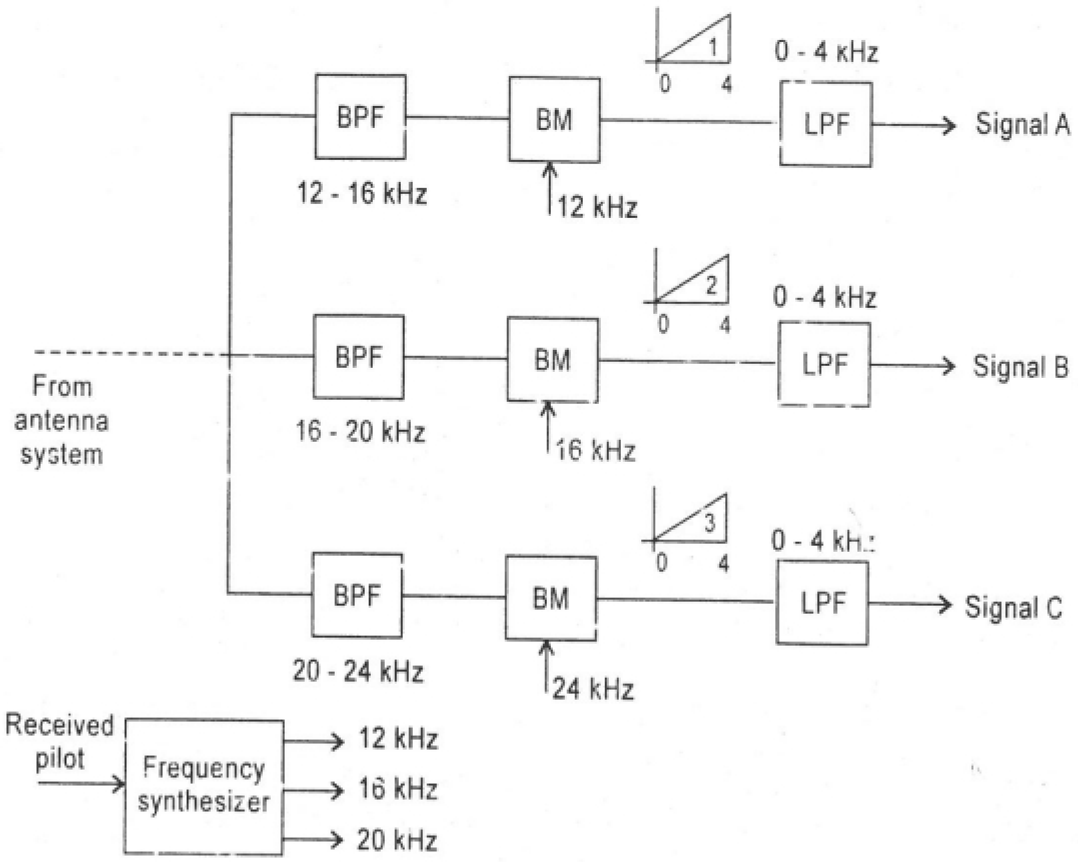
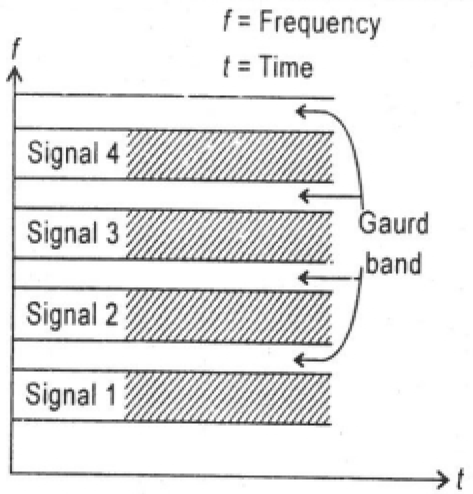
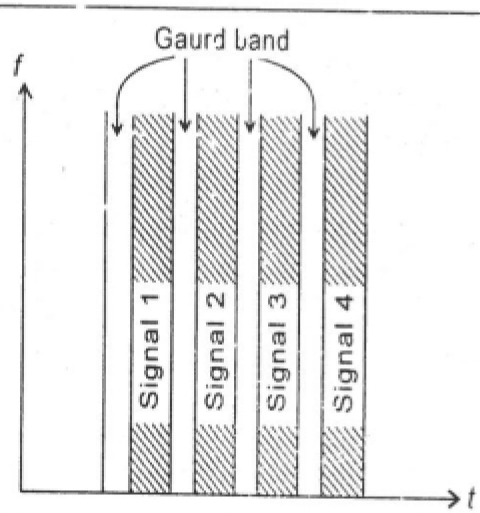


Fig. 10.3(b)

10.3 Comparison Between TDM and FDM

Q. Compare TDM and FDM.

<table>
<tr><th></th><th>FDM</th><th>TDM</th></tr>
<tr><td>(1)</td><td>

Communication Space Representation</td><td>

Communication Space Representation</td></tr>
</table>

(2)	All signals to be transmitted are continuous and are mixed in time domain.	The samples of each signal remain distinct and can be recognized and separated in time domain.
(3)	Frequency spectrum of various modulated signals occupy different bands in the frequency domain.	Frequency spectrum of various sampled signals occupy the same frequency region and are all mixed beyond recognition.
(4)	Signals are mixed in the time domain but maintain their identity in frequency domain.	Spectrum identity is maintained in time domain.
(5)	Complicated circuitry is used.	Simplified circuitry is used.
(6)	We need to generate different carriers for each channel. Each channel occupies a different frequency band and hence needs a different BPF design.	Requires identical circuits for each channel, consisting of simple synchronous commutator switches. Filters used are only LPF (identical) for each channel.
(7)	More interchannel crosstalk.	Less interchannel crosstalk.
(8)	Non-linearity in various ways produces harmonic distortion due to frequency mixing (Balanced modulator). Therefore, it will introduce interference within channels (inter channel cross talk).	Signals from different channels are not applied to the system simultanously but are allotted different time intervals. Hence non-linearity requirement in TDM are same as for a single channel.
(9)	Not preferred and less used system.	Commonly used.

Table 10.1

10.4 Frequently Asked Questions

Q.1. *What is multiplexing in communication system ?*

Draw block diagram of each TDM-PCM system and explain each block. Calculate bit rate at output of this system.

Ans.

- Multiplexing is transmission of one or more signal at one time in a single channel, such that they can be received or separated at receiver.

- It is possible either by TDM or FDM.
- Define TDM and FDM system from section 10.1 and 10.2.

TDM - PCM Systems

- From name we understand that it is Time Division Multiplexing of PCM systems.

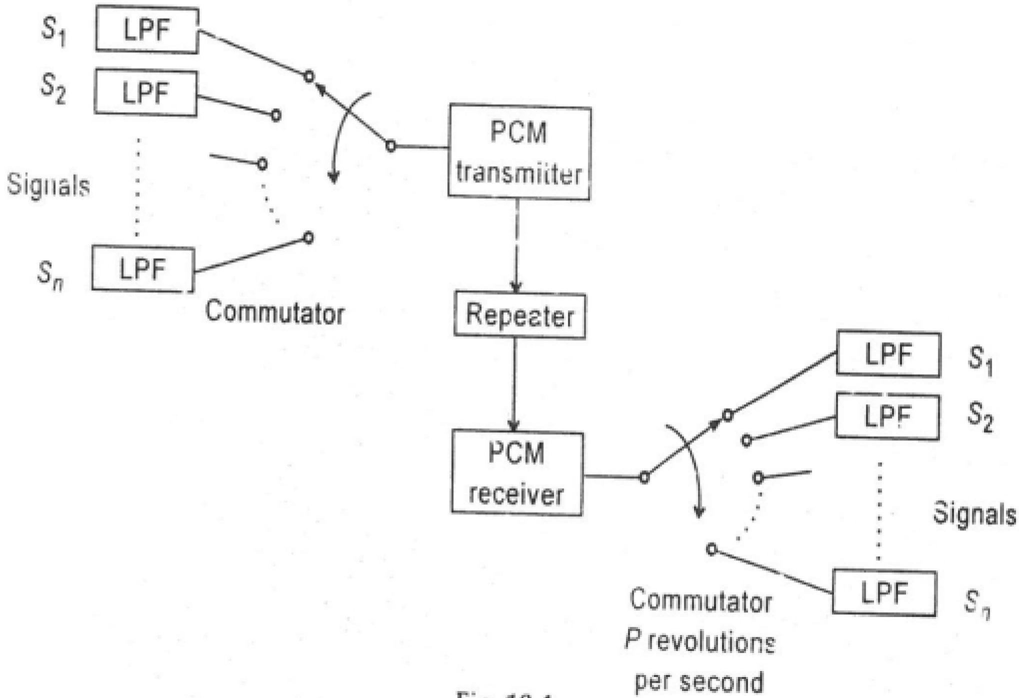


Fig. 10.4

- As from block diagram (*figure 10.4*) we can see that we first time multiplex the signal with P samples per second; and then we use PCM transmitter to transmit the multiplexed signal.
- LPF is used to band limit the signal at transmitter.
- The multiplexed signal is then converted to codes by Pulse Code Modulation.
- At receiver the reverse process is done.
- Repeater is used to sustain the power of the signal.
- LPF at receiver is used to recover the original signal.

Advantages of TDM Over PCM

- Bandwidth utilization is more than in PCM system.

T_1 System

- It is basic PCM-TDM where 24 signals, which are band limited to 3.3 kHz, are time division multiplexed.
- Hence sampling is done at 8 kHz i.e. $\geq 2 \times 3.3$ kHz.

- Calculation of bit rate of T_1 system.

Number of bits/frame

(i) Commutators sweeps continuously at 8000 revolution per sec, i.e. will generate 8000 samples per second.

i.e. Sampling rate = 8 kHz

(ii) Each sample is converted into 8 bit digital code.

(iii) In one frame i.e. one complete revolution of commutator there are 24 signals

$$\therefore \text{ Number of bits/frame } = 24 \times 8 \text{ bits}$$
$$= 192 \text{ bits}$$

(iv) After frame, 1 bit is for synchronization

$$\therefore \text{ Total bits/frame } = 192 + 1 = 193 \text{ bits/frame.}$$

- **Bit Rate**

Now commutator makes 8000 revolutions per sec.

i.e. 8000 frames are generated per second

$$\therefore \text{ Bit rate } = \text{Number of bits per sec}$$
$$= \text{Number of frames/sec} \times \text{Number of bits/frame}$$
$$= 8000 \times 193$$
$$= 1.5 \text{ Mb/sec (Mega bits per sec)}$$

- **Bandwidth** $= \frac{1}{2}$ Bit rate $= \frac{1}{2} \times 1.5$ Mb/sec $= 750$ kHz

- **Duration of Each Pulse**

We have 193 bits in $\frac{1}{8000}$ sec

$$\therefore \text{ 1 bit } = \frac{1}{8000 \times 193}$$
$$= 6.47 \times 10^{-7} \text{ sec}$$

Q.2. Explain TDM and FDM.

Ans. Write in brief from section 10.1 and 10.2.

Q.3. What do you mean by TDM ? Compare TDM with FDM.

Ans. Write in brief from section 10.1 and complete 10.3.

Q.4. ***Why is multiplexing used in communication systems ? Give one practical system where (i) FDM and (ii) TDM is used.***

Ans. Refer Q.1. from FAQ's.

Practical example where FDM is used : Telephone System

Practical example where TDM is used : T_1 System.

Q.5. ***One analog waveform $W_1(t)$, is bandlimited to 3 kHz and another $W_2(t)$ is bandlimited to 9 kHz. These two signals are to be sent by TDM over a PAM type system :***

(i) Determine the minimum sampling frequency for each signal.

(ii) Design a TDM commutator and decommutator to accommodate the signals.

(iii) Draw some typical waveforms for $W_1(t)$ and $W_2(t)$ and sketch the corresponding TDM-PAM waveform.

Ans. (i) For signal S_1

$$f_{S_1} = 2f_{m_1}$$

$$\therefore\ f_{S_1} = 6 \text{ kHz}$$

For signal S_2

$$f_{S_2} = 2f_{m_2}$$

$$\therefore\ f_{S_2} = 18 \text{ kHz}$$

where f_{S_1} and f_{S_2} = Sampling frequencies.

(ii) TDM commutator and decommutator to accommodate the two signals :

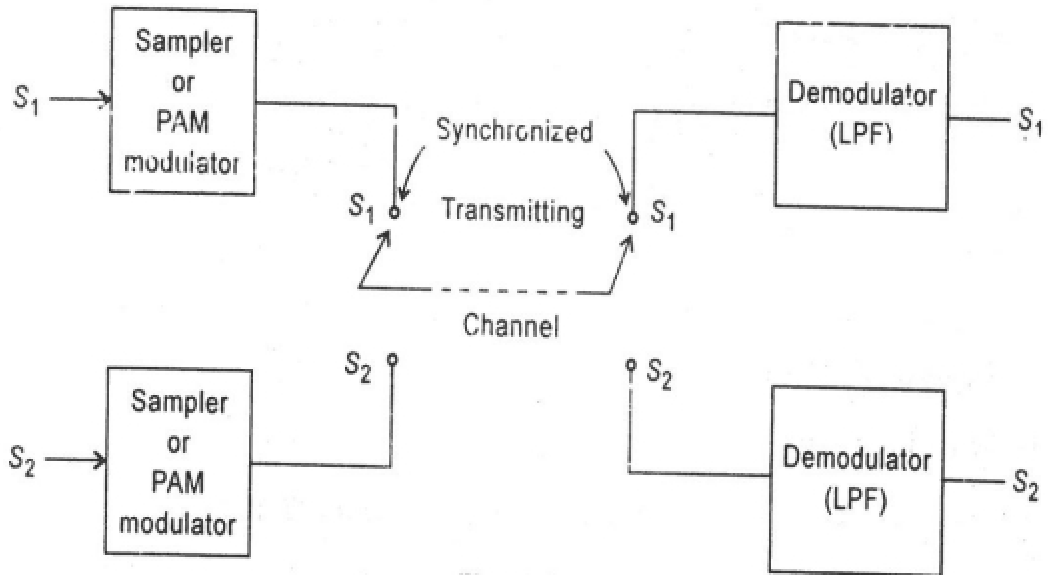


Fig. 10.5

(iii) Waveforms :

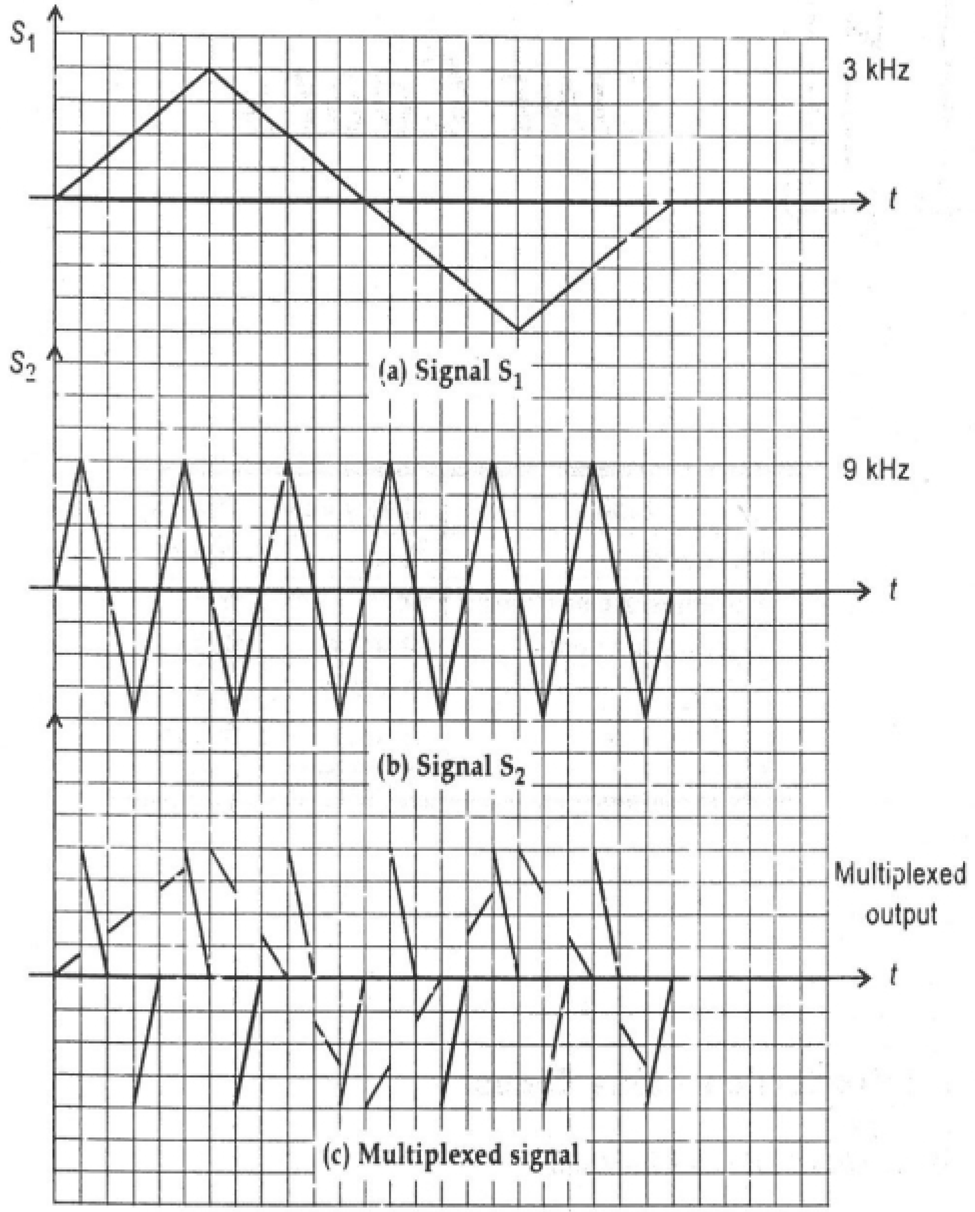


Fig. 10.6

Note : *Triangular waves are just taken for simplicity*

***

11 LINE CODES

Topic	Theory imp	Oral imp
Line Codes	☆☆	☆☆☆
Properties of Line Codes	☆	☆
Types of Line Codes	☆☆☆☆	☆☆
PSD	☆☆☆☆	☆☆
FAQ's	☆☆☆	☆☆☆
Problems	☆☆☆☆	☆☆

11.1 Introduction - Line Codes

Q. Write short note on Line Codes.

The information from sources such as computers is discrete in nature.

If such discrete signal is transmitted over a communication channel, the signal gets dispersed. Due to this pulses overlap and it results in distortion.

To avoid this distortion, the data is first converted to suitable format before transmission over a communication channel. The various formats used are called as *Line Codes.*

The formats or line codes are :

- Non-return to zero (NRZ) and return to zero (RZ) unipolar format.
- NRZ and RZ polar format.
- NRZ bipolar format.
- Manchester format.
- Polar quaternary NRZ format.

Parameters / Properties of Line Codes

The following are some important properties of line codes :

(1) The cable systems and other communication systems do not allow transmission of a dc signal. Therefore the signal transmitted must have a zero average value. NRZ bipolar format usually satisfy this requirement.

(2) As the code adds redundancy, the code efficiency should be as high as possible.

(3) The signal should undergo a sufficient number of zero crossings for synchronization at the receiver.

(4) The amount of energy in the signal at low frequencies should be small in order to minimize the crosstalk between channels.

11.2 Different Types of Line Codes

We have earlier seen what are the different formats/types of Line codes. Now we will discuss them in detail.

(1) Unipolar RZ Format

- The unipolar return to zero format is as shown in *figure 11.1*.

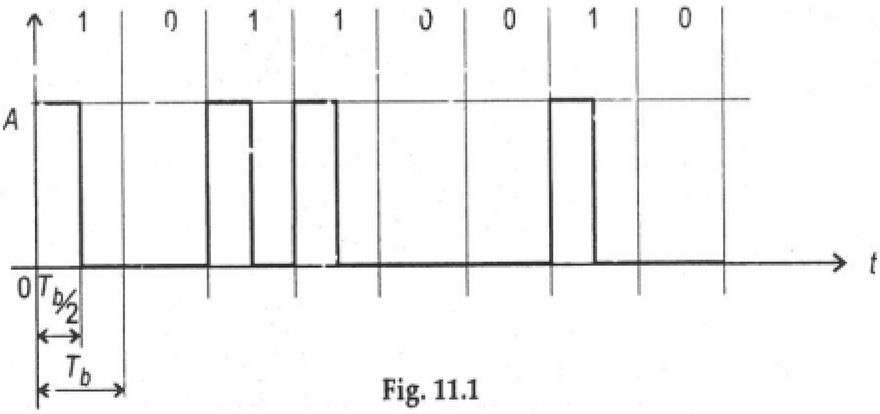


Fig. 11.1

- In this format each logic "0" is represented by an off pulse i.e. no change in level.
- Each logic "1" is represented by a pulse with amplitude A for duration of $T_b/2$ where T_b is bit duration.
- After $T_b/2$. the pulse returns to zero level, hence called *return to zero format* or *RZ format*.
- As the voltage level is either $+A$ or zero i.e. with only one polarity, it is called as a *unipolar format*.

(2) Unipolar NRZ Format

- Unipolar Non-return to zero format is shown in below *figure 11.2*.

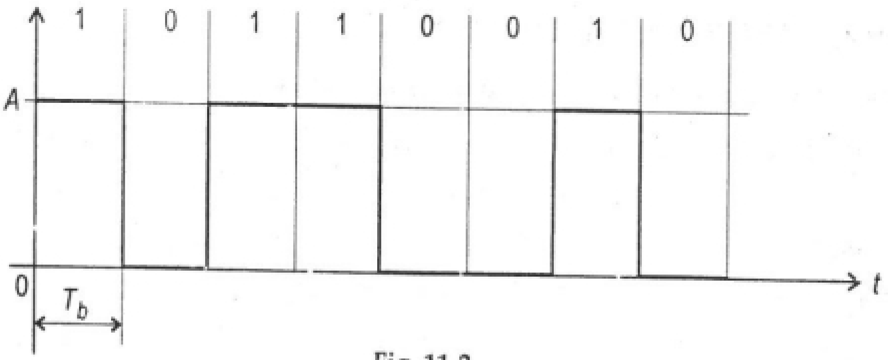


Fig. 11.2

- In this format each logic "0" is represented by an off pulse i.e. no change in the voltage level.
- Each logic "1" is represented by a pulse with amplitude A for complete duration T_b.
- In this the pulse does not return to zero after half bit period, hence the name non return to zero.
- As the pulse has voltage level either $+A$ or zero, it is called as *unipolar format*.

(3) Polar RZ Format

- The Polar return to zero format is as shown in *figure 11.3*.

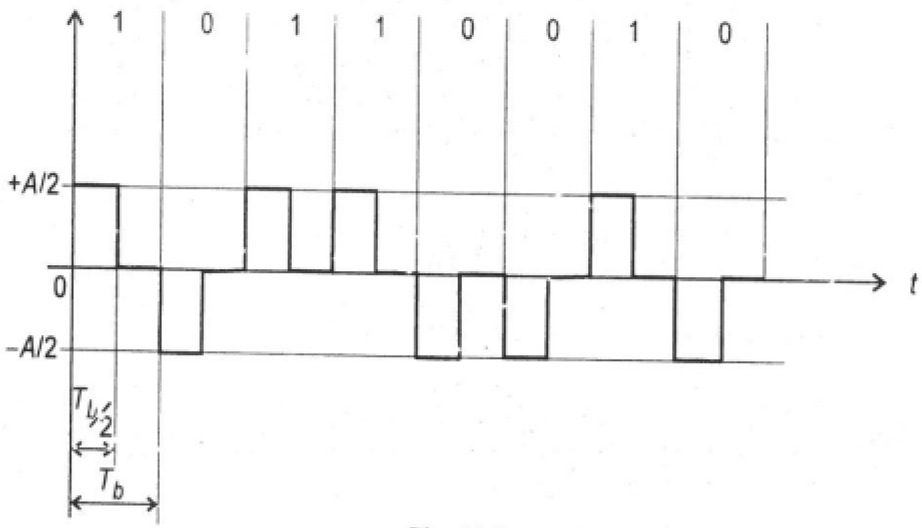


Fig. 11.3

- In this format each logic "0" is represented by a pulse with amplitude $-A/2$ for duration of $T_b/2$.
- Logic "1" is represented by a pulse with amplitude $+A/2$ for duration $T_b/2$.
- This format shows two opposite polarity phases with amplitudes $\pm A/2$, hence called *Polar format*.
- As the pulses return to zero level after $T_b/2$, it is called as *RZ format*.

(4) Polar NRZ Format

- The Polar non-return to zero format is as shown in *figure 11.4*.

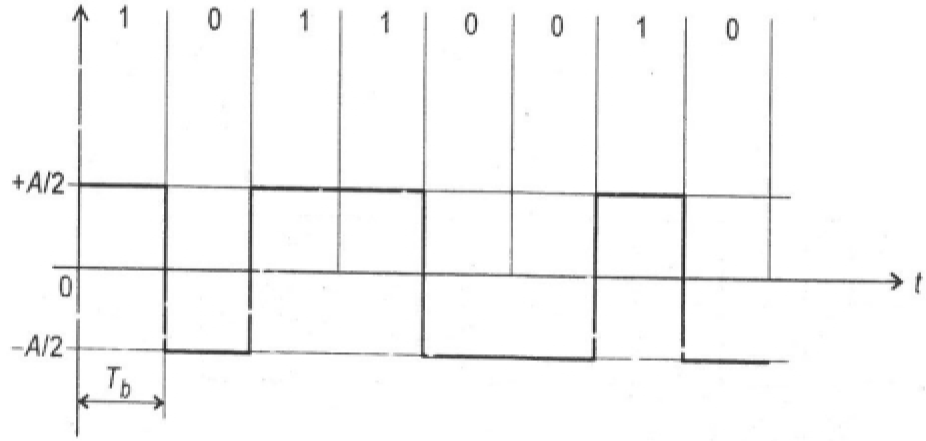


Fig. 11.4

- In this format each logic "0" is represented by pulse of amplitude $-A/2$ for duration of T_b.
- Logic "1" is represented by a pulse of amplitude $+A/2$ for complete duration T_b.

(5) Bipolar NRZ Format (AMI)

- The bipolar non-return to zero format is as shown in *figure 11.5*.

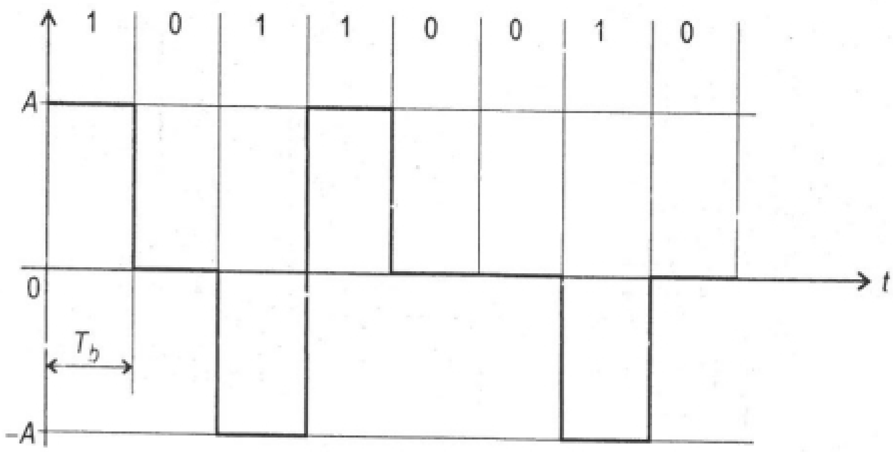


Fig. 11.5

- In this format each logic "0" is represented by an off pulse i.e. no pulse.
- Successive "1's" are represented by pulses with alternating polarity.

  In this format there are three voltage levels 0, $+A$, and $-A$. Hence called *pseudoternary* or *Alternative Mark Inversion (AMI) format*.

(6) Split Phase Manchester Format :

- The split phase Manchester format is as shown in *figure 11.6*.

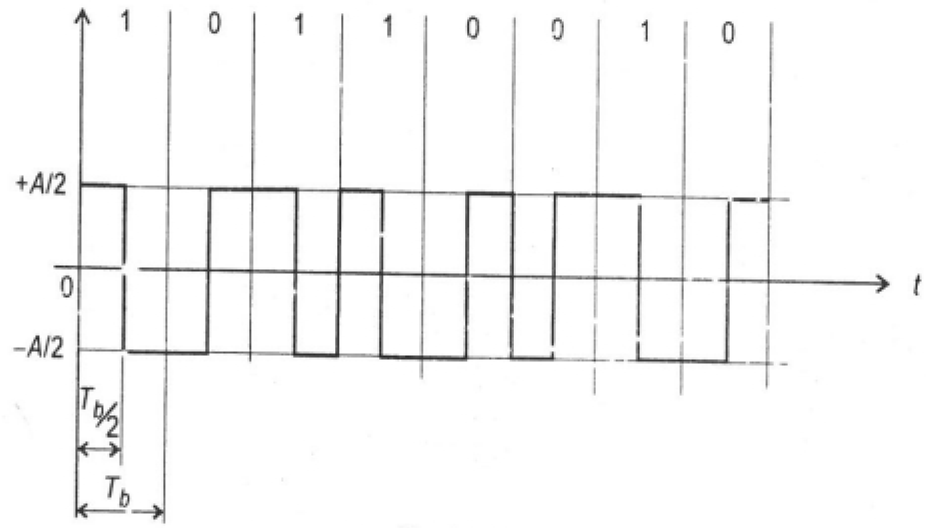


Fig. 11.6

- In this format each logic "1" is represented by a pulse of amplitude $+A/2$ for a half bit duration i.e. $T_b/2$ followed by a pulse of amplitude $-A/2$ for remaining bit duration.
- Thus, for transmitting "1" the shape of pulse will be as shown in *figure 11.7*.

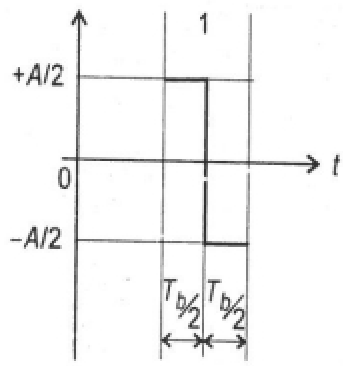


Fig. 11.7

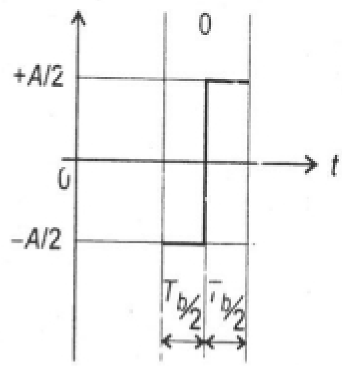


Fig. 11.8

- For representing logic "0" the pulses are used in reverse order.
- Thus, for transmitting "0" the shape of pulse will be as shown in *figure 11.8*.

(7) Polar Quaternary Format

- The *figure 11.9* shows polar quaternary non-return to zero format.

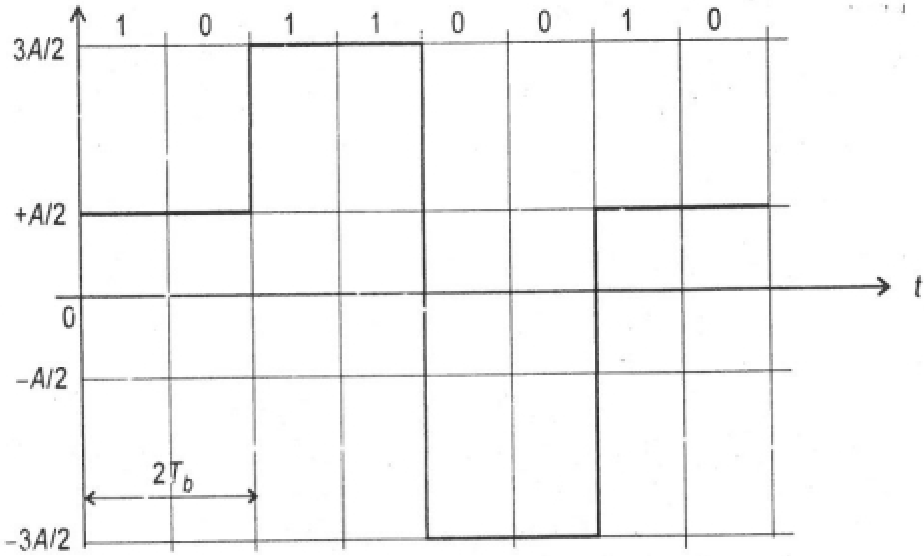


Fig. 11.9

Message Grouping	Level
0 0	-3A/2
0 1	-A/2
1 0	A/2
1 1	3A/2

- This format groups the message bits in the blocks of two.
- Using the table given above, the corresponding amplitude level is assigned to the group of 2 bits.
- There are 4 combinations available, so there are 4 levels.
- The number of levels can be calculated using following relation.

$$m = 2^k$$

where m = Number of levels

k = Number of bits in a group.

11.3 Power Spectral Density (PSD)

Definition : Power Spectral Density of any signal describes how the signal power is distributed with frequency.

Explaination : Power Spectral Density or power spectrum is just the graph of power versus frequency.

We all know that every signal can be expressed in terms of addition of many sine waves. Each sine wave has some amplitude and frequency, and since power is proportional to the square of amplitude, we can say that each sine wave has some power and frequency. PSD is just the graph of this power plotted against its frequency.

Example :

Consider the equation of the AM wave

$$v_{AM} = V_c \sin 2\pi f_c t + \frac{mV_c}{2} [(\cos 2\pi (f_c - f_m)t - \cos 2\pi (f_c + f_m)t]$$

The frequencies involved in this signal and their corresponding power are as follows :

	Frequency	Power
Carrier	f_c	$V_c^2/2R$
LSB	$f_c - f_m$	$m^2V_c^2/8R$
USB	$f_c + f_m$	$m^2V_c^2/8R$

∴ The power spectrum or PSD is

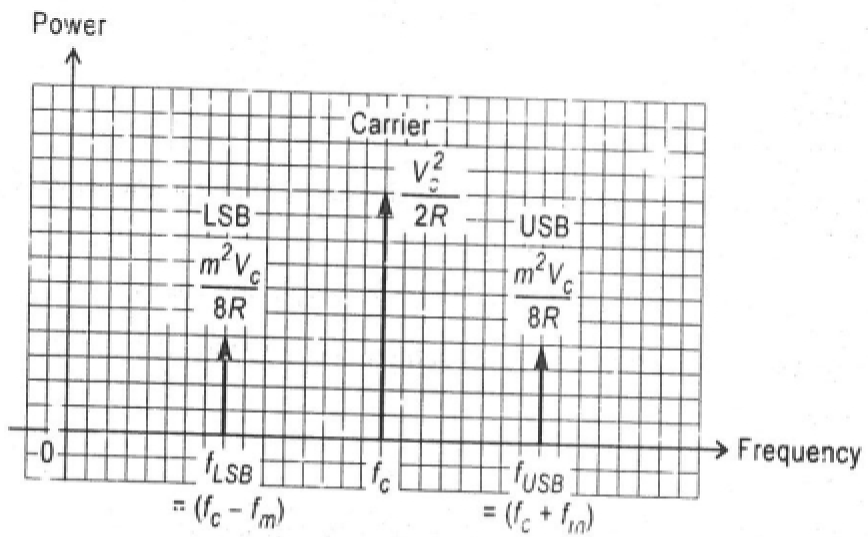


Fig. 11.10

Hence, we can say that, PSD of any signal describes how the signal power is distributed with frequency.

11.3.1 Significance of PSD

(i) It describes how much amount of power is carried by which frequency.

(ii) The total power of the signal can be calculated by integrating the PSD.

$$\therefore \; P_{Total} = \int_{-\infty}^{\infty} PSD$$

If we want to find the power contributed by a range of frequencies $[f_1, f_2]$ then,

$$P_{(f_1 - f_2)} = \int_{f_1}^{f_2} PSD$$

11.4 Power Spectral Density of Line Codes

Q. ***Derive and plot the power spectrum for polar NRZ signal. What is the first null bandwidth ?***

> **Note :** *For Null Bandwidth, refer Q.4. FAQ's.*

To find the PSD of any line code like NRZ polar, Manchester code etc. follow the following procedure

(i) Express the given signal as a mathematical function of time i.e. $x(t)$.

(ii) Find the Fourier transform of $x(t)$ i.e. $X(f)$.

$$X(f) = \int_{-T_b/2}^{T_b/2} x(t)\, e^{-j2\pi ft}\, dt$$

where T_b = Bit interval/duration.

(iii)
$$PSD = \frac{|X(f)|^2}{T_b}$$

PSD of NRZ Polar Signal

> **Note :** *NRZ polar is also called NRZ bi-polar.*

The NRZ polar signal is either $+A$ or $-A$ during any bit interval.

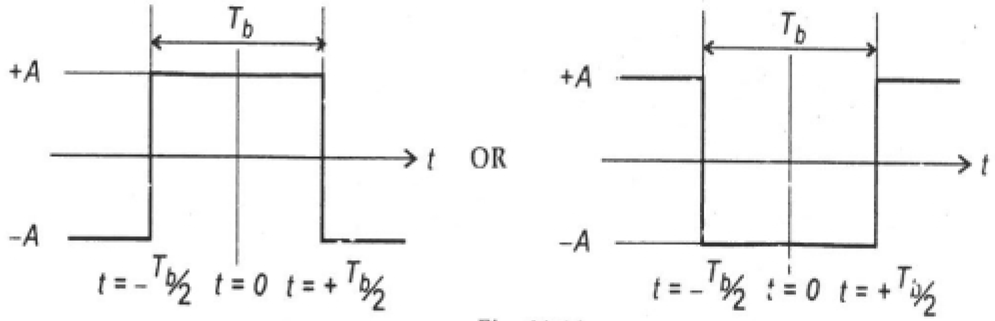


Fig. 11.11

Step (i) : Find $x(t)$

From the *figure 11.11* we can conclude

$$x(t) = \pm A \qquad -\frac{T_b}{2} \le t \le \frac{T_b}{2}$$

Step (ii) : Obtain $X(f)$

$$X(f) = \int_{-T_b/2}^{T_b/2} x(t)\, e^{-j2\pi ft}\, dt$$

$$= \int_{-T_b/2}^{T_b/2} \pm A\, e^{-j2\pi ft}\, dt = \pm A \left. \frac{e^{-j2\pi ft}}{-j2\pi f} \right|_{-T_b/2}^{T_b/2}$$

$$= \frac{\pm A}{-j2\pi f}\left[e^{-j2\pi f\, T_b/2} - e^{+j2\pi f\, T_b/2}\right]$$

$$= \frac{\pm A}{-j\pi f}\left[\frac{e^{-j\pi f\, T_b} - e^{+j\pi f\, T_b}}{2}\right]$$

$$= \frac{\pm A}{\cancel{-}\pi f}(\cancel{-}1)\left[\frac{e^{j\pi f\, T_b} - e^{-j\pi f\, T_b}}{2j}\right]$$

$$\boxed{X(f) = \frac{\pm A}{\pi f}\sin \pi f\, T_b} \qquad \left(\because \frac{e^{j\theta} - e^{-j\theta}}{2j} = \sin\theta\right)$$

Step (iii) : PSD

$$\text{PSD} = \frac{|X(f)|^2}{T_b}$$

$$\therefore \text{ PSD } = \frac{1}{T_b} \times \frac{A^2}{(\pi f)^2} \sin^2 \pi f T_b$$

$$= A^2 T_b \left(\frac{\sin \pi f T_b}{\pi f T_b} \right)^2$$

Now,

$$\frac{\sin \pi x}{\pi x} = \text{sinc } x, \quad \text{sinc function of } x$$

$$\therefore \boxed{\text{PSD } = A^2 T_b (\text{sinc } f T_b)^2}$$

PSD of NRZ Unipolar Signal

NRZ unipolar signal is either one or zero during any bit interval. It is as shown in *figure 11.12.*

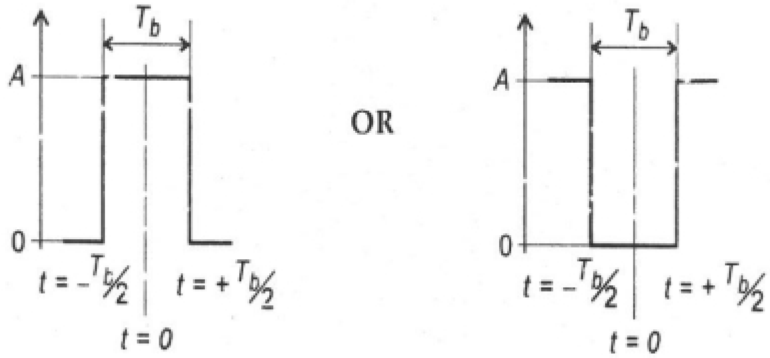


Fig. 11.12

The above signal can be redrawn as shown in *figure 11.13.*

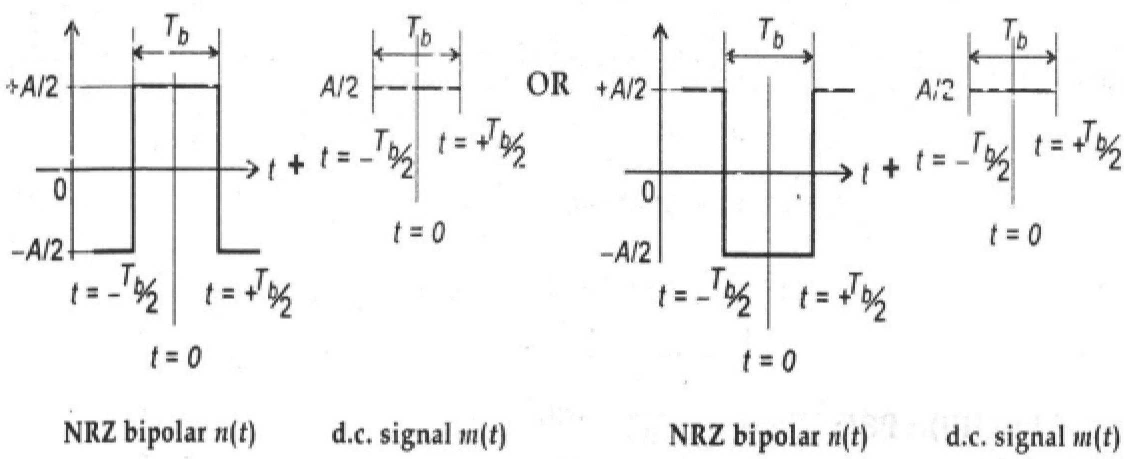


Fig. 11.13

Graphically

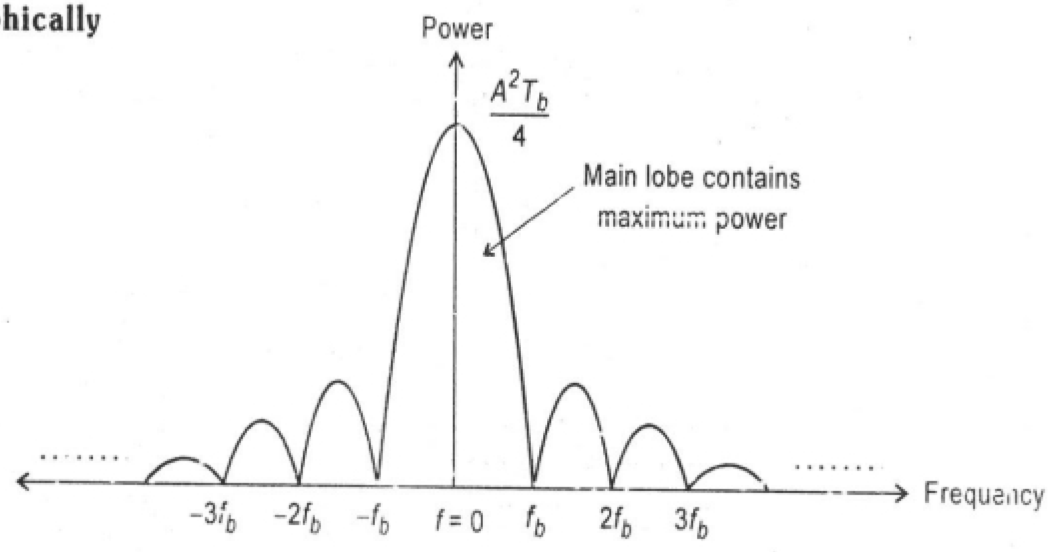


Fig. 11.14 : PSD of NRZ polar signal.

Step (i) : Find $x(t)$

From the above *figure 11.14* we can say that NRZ unipolar signal can be written as a summation of NRZ polar signal and a d.c. signal.

$$\therefore \underbrace{x(t)}_{\text{NRZ unipolar}} = \underbrace{n(t)}_{\text{NRZ bipolar}} + \underbrace{m(t)}_{\text{d.c. signal}}$$

$$\text{Here } m(t) = \frac{A}{2}, \text{ is a d.c. signal}$$

Step (ii) : Obtain Fourier Transform

This step is skipped since we already know Fourier transform of NRZ polar signal and Fourier transform of d.c. signal is not required because we will directly get its PSD in the next step.

Step (iii) : PSD

$$\because x(t) = n(t) + m(t)$$

$$\therefore \text{PSD}(x(t)) = \text{PSD}(n(t)) + \text{PSD}(m(t)) \quad \text{..... (1)}$$

Since, $n(t)$ is an NRZ polar signal with amplitude $A/2$, its PSD is

$$\boxed{\text{PSD}(n(t)) = \frac{A^2}{4} T_b(\text{sinc } f T_b)^2} \quad \text{..... (2)}$$

.. {from previous derivation}

Now, since $m(t)$ is a d.c. signal thus the frequency involved in this d.c. signal is $f = 0$ and the power carried by this frequency is just the square of amplitude.

Just recall the definition of PSD and to cut a long story short, just remember that the PSD of a d.c. signal is an impulse function with magnitude equal to square of the amplitude.

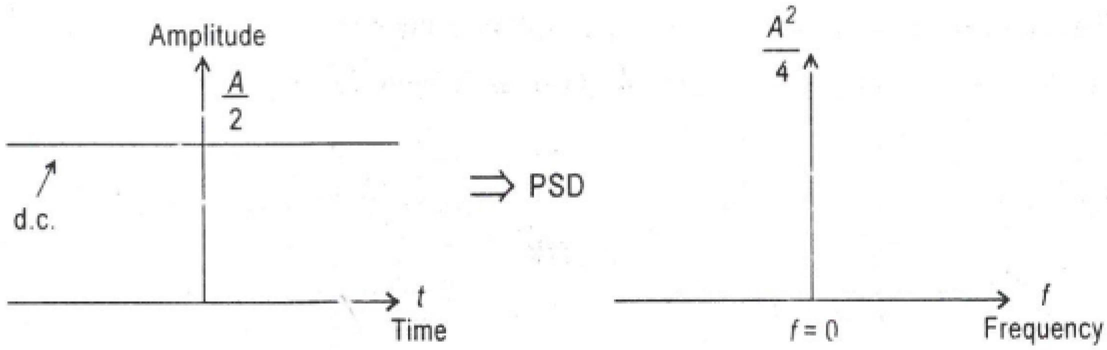


Fig. 11.15

> **Note :** *An impulse function is given by*
> $$\delta(f) = 1 \qquad f = 0$$
> $$= 0 \qquad f \neq 0$$

$$\therefore \quad \boxed{\text{PSD }(m(t)) = \frac{A^2}{4}\,\delta(f)} \qquad \text{..... (3)}$$

From equation (1), (2) and (3)

$$\therefore \quad \boxed{\text{PSD }(x(t)) = \frac{A^2}{4}\,T_b(\text{sinc } f\,T_b)^2 + \frac{A^2}{4}\delta(f)}$$

Graphically

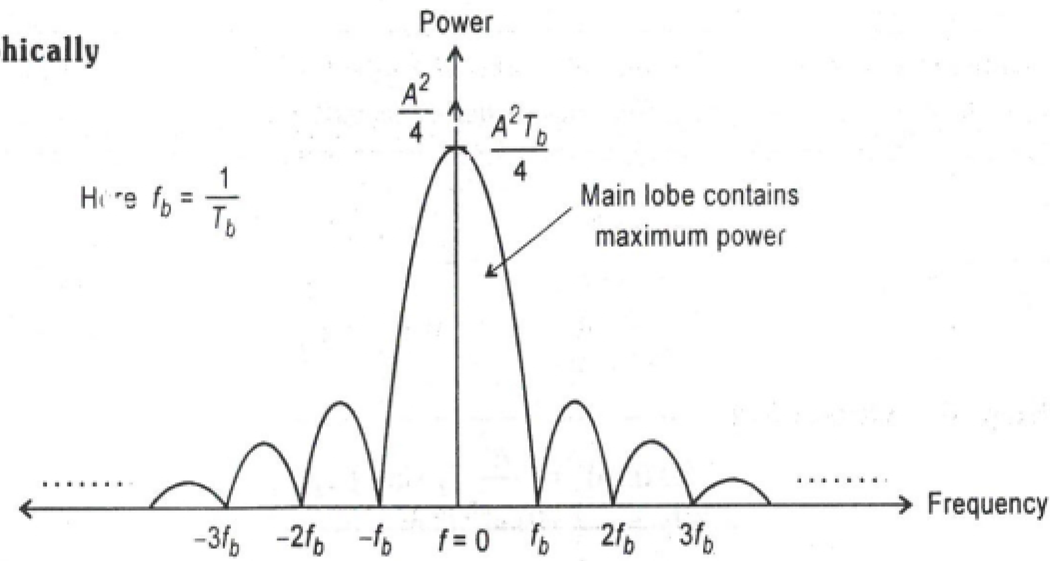


Fig. 11.16 : PSD of NRZ unipolar signal.

PSD of Manchester Code

In Manchester code, logic 1 is represented as in *figure 11.17(a)* and logic 0 is represented as in *figure 11.17(b)*.

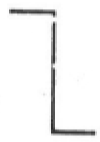


Fig. 11.17(a)

Fig. 11.17(b)

Thus we can see that always there is a transition in the middle.

A Manchester code can be graphically given as in *figure 11.18*.

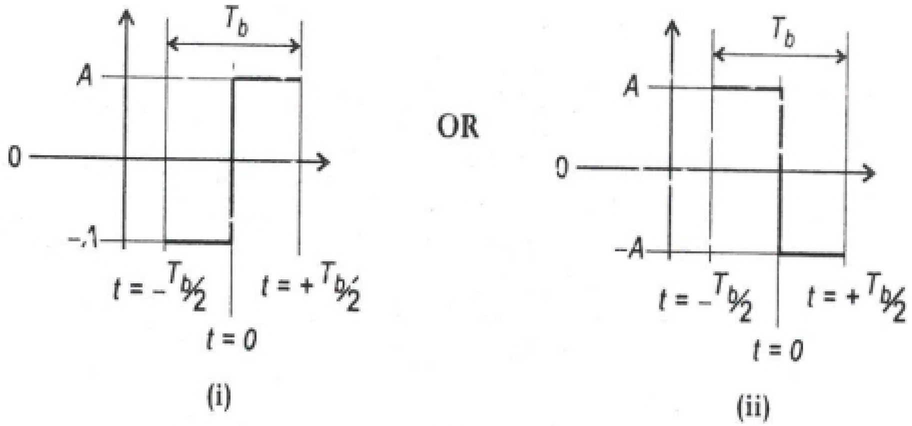


Fig. 11.18

Step (i) : Find $x(t)$

Looking at the *figure 11.18* we can say that, $x(t)$ is given by

Case (i)

$$x(t) = -A \qquad -\frac{T_b}{2} \le t \le 0$$
$$= +A \qquad 0 \le t \le \frac{T_b}{2}$$

OR

Case (ii)

$$x(t) = A \qquad -\frac{T_b}{2} \le t \le 0$$
$$= -A \qquad 0 \le t \le \frac{T_b}{2}$$

> **Note :** *Just consider case (ii) and solve, since it does not matter whether we use case (i) equation or case (ii) equation, final answer will be the same*

$\therefore$ Let

$$x(t) = A \qquad -\frac{T_b}{2} \le t \le 0$$
$$= -A \qquad 0 \le t \le \frac{T_b}{2}$$

Step (ii) : Obtain $X(f)$

$$X(f) = \int_{-T_b/2}^{T_b/2} x(t)\, e^{-j2\pi ft}\, dt$$

$$\therefore\ X(f) = \int_{-T_b/2}^{0} A\, e^{-j2\pi ft}\, dt + \int_{0}^{T_b/2} (-A)\, e^{-j2\pi ft}\, dt$$

$$= A\left[\frac{e^{-j2\pi ft}}{-j2\pi f}\right]_{-T_b/2}^{0} - A\left[\frac{e^{-j2\pi ft}}{-j2\pi f}\right]_{0}^{T_b/2}$$

$$= \frac{A}{-j2\pi f}\left[e^{0} - e^{j2\pi f\, T_b/2}\right] - \frac{A}{-j2\pi f}\left[e^{-j2\pi f\, T_b/2} - 1\right]$$

$$= \frac{A}{-j2\pi f}\left[1 - e^{j\pi f\, T_b} - e^{-j\pi f\, T_b} + 1\right]$$

$$= \frac{A}{-j2\pi f}\left[2 - (e^{j\pi f\, T_b} + e^{-j\pi f\, T_b})\right]$$

$$= \frac{-A}{j\pi f}\left[1 - \left(\frac{e^{j\pi f\, T_b} + e^{-j\pi f\, T_b}}{2}\right)\right]$$

$$\boxed{X(f) = \frac{jA}{\pi f}\left[1 - \cos \pi f\, T_b\right]} \qquad \left(\because \frac{e^{j\theta} + e^{-j\theta}}{2} = \cos\theta\right)$$

Step (iii) : PSD

$$\text{PSD} = \frac{|X(f)|^2}{T_b}$$

Note : $\because X(f) = \dfrac{jA}{\pi f}\left(1 - \cos \pi f\, T_b\right)$

$\therefore\ |X(f)| = \dfrac{A}{\pi f}\left(1 - \cos \pi f\, T_b\right)$

$$= \frac{A^2 T_b}{(\pi f\, T_b)^2}\left(1 - \cos \pi f\, T_b\right)^2$$

$$\boxed{\text{PSD} = A^2 T_b \left[\frac{1 - \cos \pi f\, T_b}{\pi f\, T_b}\right]^2}$$

This equation can be simplified as

$$\text{PSD} = A^2 T_b \left[\frac{2 \sin^2\left(\frac{\pi f\, T_b}{2}\right)}{\pi f\, T_b}\right]^2 \qquad \left(\because 1 - \cos 2\theta = 2 \sin^2\theta\right)$$

$$\boxed{\text{PSD} = A^2 T_b \left[\frac{\sin^2\left(\frac{\pi f T_b}{2}\right)}{\left(\frac{\pi f T_b}{2}\right)} \right]^2}$$

Graphically

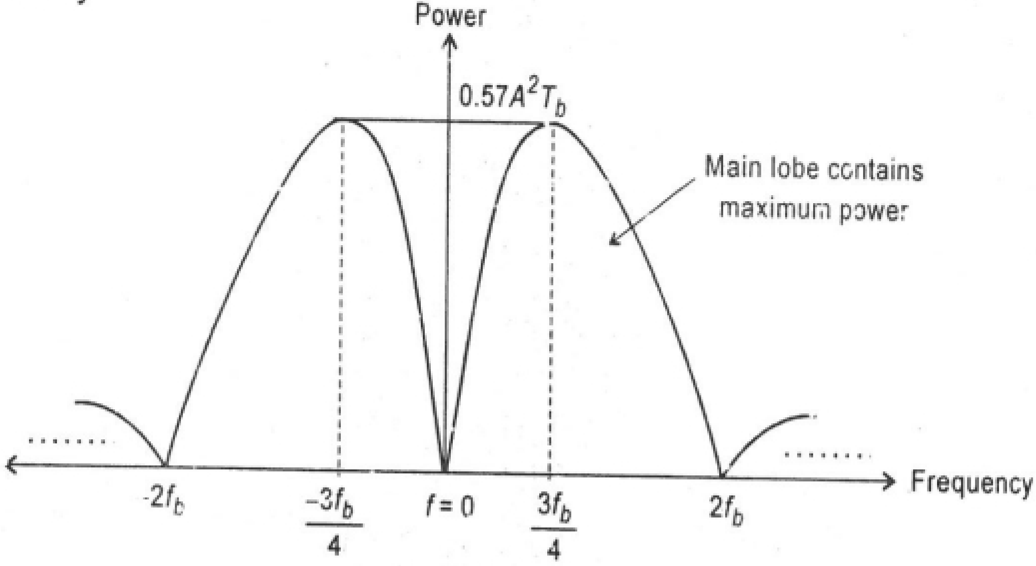


Fig. 11.19

11.5 Frequently Asked Questions

Q.1. Compare all the line codes ?

Ans.

Sr. No.	Parameter	Polar RZ	Polar NRZ	AMI	Manchester	Polar Quaternary NRZ
(1)	Transmission of d.c. component	Yes	Yes	No	No	Possible
(2)	Signaling rate	$1/T_b$	$1/T_b$	$1/T_b$	$1/T_b$	$1/2\ T_b$
(3)	Noise immunity	Low	Low	High	High	High
(4)	Synchronizing capability	Poor	Poor	Very good	Very good	Poor
(5)	Bandwidth required	$1/T_b$	$1/2\ T_b$	$1/2\ T_b$	$1/T_b$	$1/2\ T_b$
(6)	Crosstalk	High	High	Low	Low	Low

Table 11.1 : Comparison of line codes.

Q.2. Explain ISI.

Ans. Intersymbol Interference (ISI) :

In a communication system when the data is being transmitted in the form of pulses (bits) the output produced at the receiver due to the other bits or symbols interferes with the output produced by the desired bit.

This is called as *intersymbol interference (ISI)*. The intersymbol interference will introduce errors in the detected signal.

Cause of Intersymbol Interference :

The ISI arises due to the imperfections in the overall frequency response of the system. When a short pulse of duration T_b seconds is transmitted through a bandlimited system, then the frequency components contained in the input pulse are *differentially attenuated* and more importantly *differentially delayed* by the system. Due to this the pulse appearing at the output of the system will be **dispersed** over an interval which is longer then T_b seconds. Due to this dispersion, the symbols each of duration T_b will interfere with each other when transmitted over the communication channel. This will result in the intersymbol interference (ISI).

Effect of ISI :

In the absence of ISI and noise, the transmitted bit can be decoded correctly at the receiver. The presence of ISI will introduce errors in the decision device at the receiver output. Thus the receiver can make an error in deciding whether it has received a logic 1 or a logic 0.

Remedy to Reduce the ISI :

It has been proved that the function which produces a zero intersymbol interference is a *sinc function*. Thus instead of a rectangular pulse if we transmit a sinc pulse then the ISI can be reduced to zero. This is known as *Nyquist Pulse Shaping*.

Q.3. Prove following properties of Fourier transform :

(i) Linearity

(ii) Time shifting

(iii) Convolution in the time domain.

Note : *FT = Fourier transform*

Ans. The properties of Fourier transforms are :

(i) Scaling and Linearity :

If

$$x_1(t) \xleftrightarrow{FT} X_1(f)$$

$$x_2(t) \xleftrightarrow{FT} X_2(f)$$

then

$$ax_1(t) \xleftrightarrow{FT} aX_1(f)$$

$$ax_1(t) + bx_2(t) \xleftrightarrow{FT} aX_1(f) + bX_2(f)$$

Proof :

$$FT\left[ax_1(t) + bx_2(t)\right] = \int_{-\infty}^{\infty}\left[ax_1(t) + bx_2(t)\right] e^{-j2\pi ft}\, dt$$

$$= a\int_{-\infty}^{\infty} x_1(t)\, e^{-j2\pi ft}\, dt + b\int_{-\infty}^{\infty} x_2(t)\, e^{-j2\pi ft}\, dt$$

$$\boxed{FT\left[ax_1(t) + bx_2(t)\right] = aX_1(f) + bX_2(f)}$$

(ii) Time Shift Property :

if

$$x(t) \xleftrightarrow{FT} X(f)$$

then

$$x(t-k) \xleftrightarrow{FT} e^{-j2\pi fk} X(f)$$

Proof :

$$FT\left[x(t-k)\right] = \int_{-\infty}^{\infty} x(t-k)\; e^{-j2\pi ft}\, dt$$

Put $t - k = m$

$\therefore t = m + k$

$$\therefore\ FT\left[x(t-k)\right] = \int_{-\infty}^{\infty} x(m)\; e^{-j2\pi f(m+k)}\, dt$$

$$= e^{-j2\pi fk}\int_{-\infty}^{\infty} x(m)\; e^{-j2\pi fm}\, dt$$

$$\boxed{FT\left[x(t-k)\right] = e^{-j2\pi fk} X(f)}$$

(iii) Convolution Property :

If

$$x(t) \xleftrightarrow{FT} X(f)$$

$$h(t) \xleftrightarrow{FT} H(f)$$

then

$$x(t) * h(t) \xleftrightarrow{FT} X(f) \cdot H(f)$$

Proof :

$$FT\left[x(t) * h(t)\right] = FT\left(\int_{-\infty}^{\infty} x(m)\, h(t-m)\, dm\right)$$

$$= \int_{-\infty}^{\infty}\left(\int_{-\infty}^{\infty} x(m)\, h(t-m)\, dm\right) e^{-j2\pi ft}\, dt$$

Put $t - m = n$

$\therefore t = m + n$

$$FT\left[x(t) * h(t)\right] = \int_{-\infty}^{\infty}\int_{-\infty}^{\infty} x(m)\, h(n)\, e^{-j2\pi f(m+n)}\, dm\, dn$$

$$= \int_{-\infty}^{\infty} x(m)\, e^{-j2\pi fm}\, dm \int_{-\infty}^{\infty} h(n)\, e^{-j2\pi fn}\, dn$$

$$\boxed{FT\left(x(t) * h(t)\right) = X(f) \cdot H(f)}$$

Note : *Definition of convolution is :*

$$x(t) * h(t) = \int_{-\infty}^{\infty} x(m)\, h(t-m)\, dm$$

Q.4. ***What is Null to Null bandwidth ?***

Ans. Null to Null bandwidth is the interval between the nulls on either side of the main lobe.

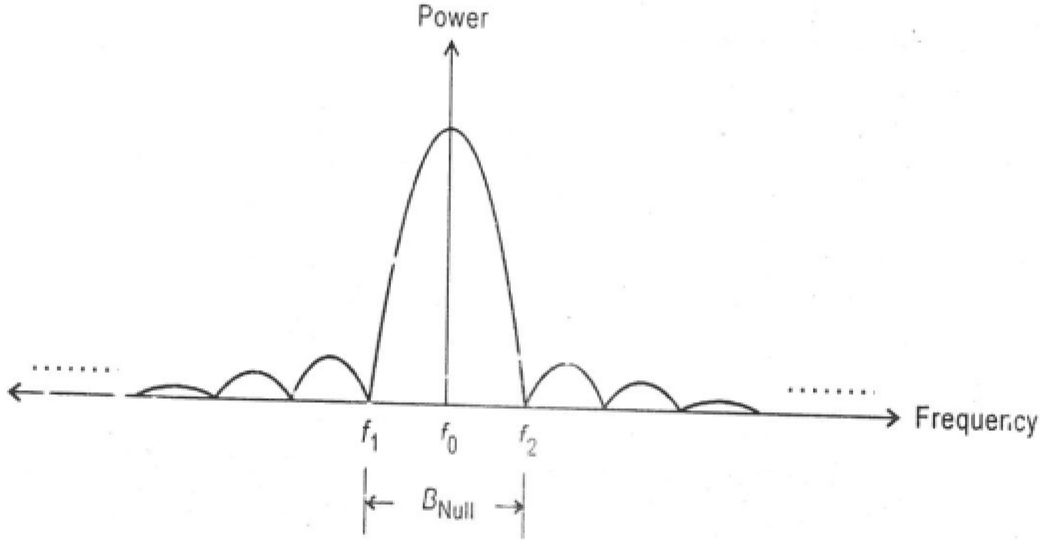


Fig. 11.20

Thus, Null to Null bandwidth is :

$$\boxed{B_{Null} = f_2 - f_1}$$

11.6 Solved Problems

Problem 1 : For binary data 1001101, sketch (i) Bipolar NRZ (ii) Bipolar RZ and (iii) Manchester Code Signaling Formats.

Solution :

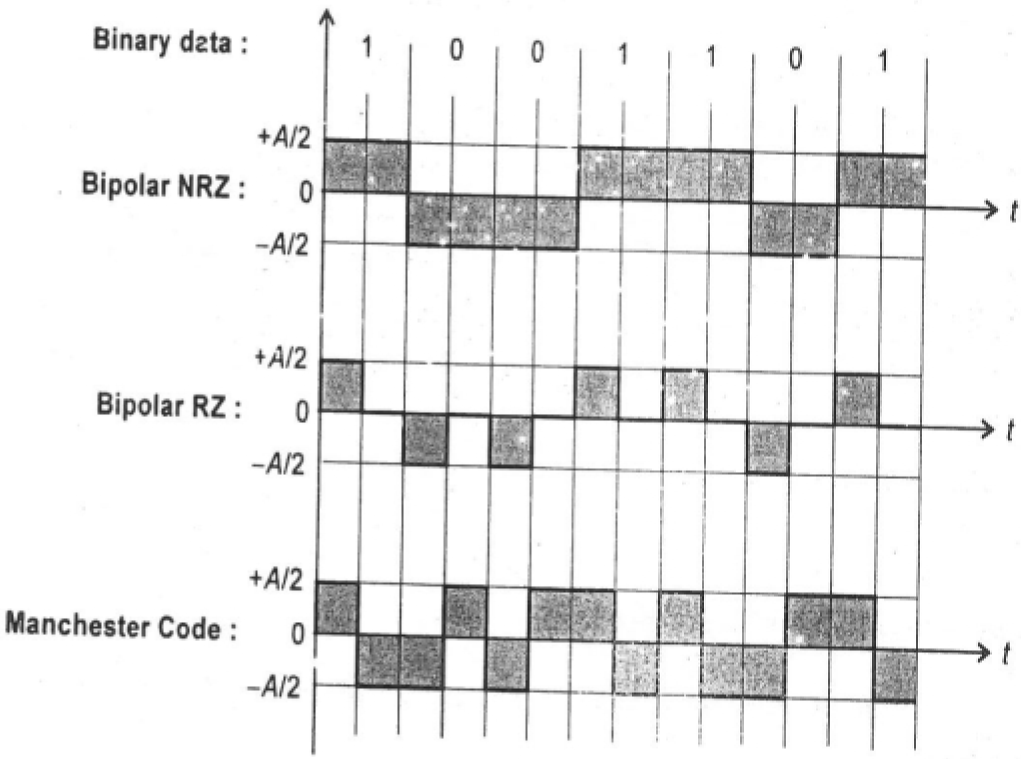


Fig. 11.21

Problem 2 : *For the binary data 10101101, sketch :*

(i) *Bipolar RZ*

(ii) *Manchester Code*

(iii) *Bipolar NRZ .*

Solution :

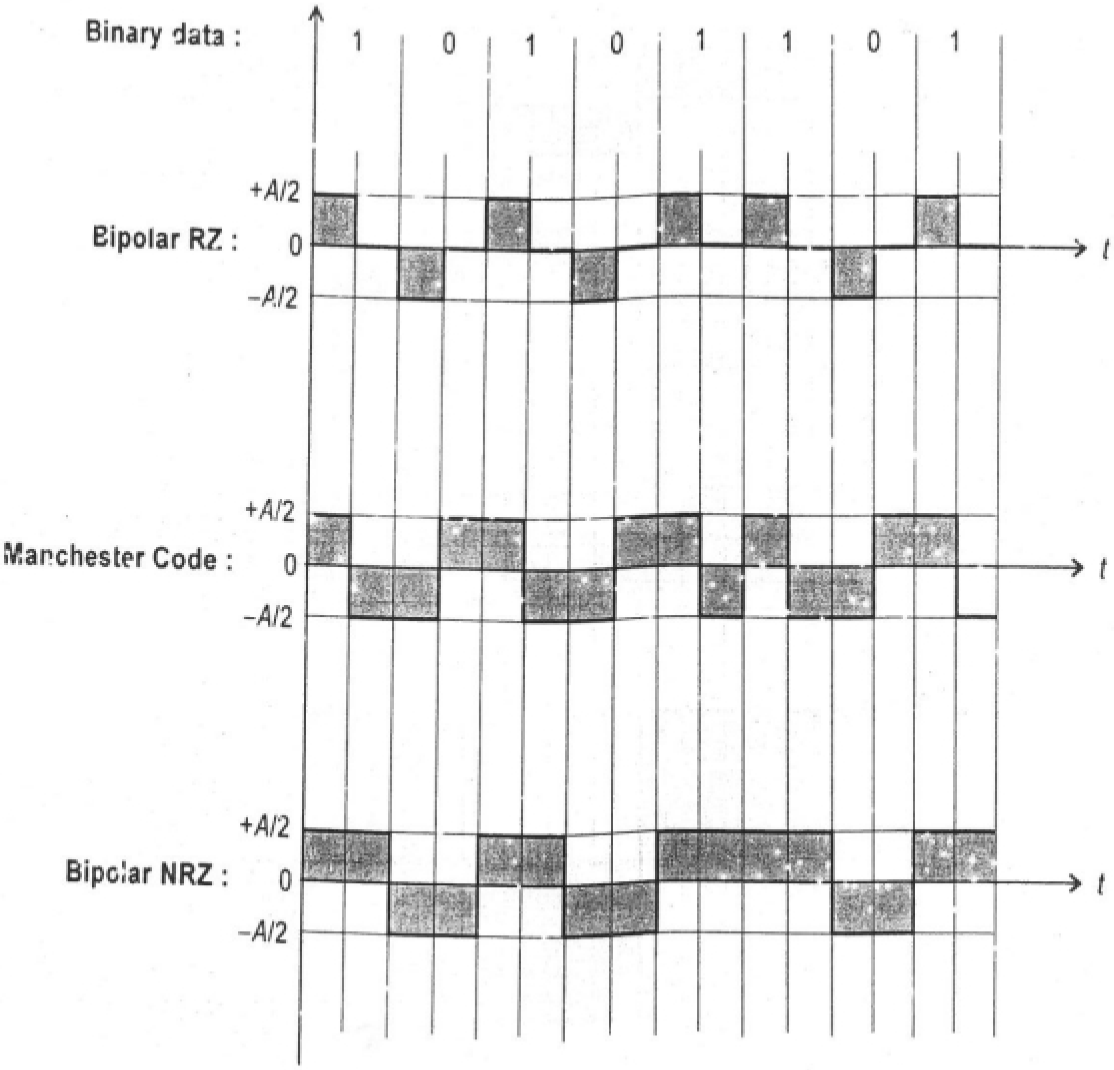


Fig. 11.22

Problem 3 : *Draw the following data forms for the bit stream 1100110 :*

(i) *Polar NRZ* *(ii)* *Unipolar RZ* *(iii)* *Manchester*

(iv) *Polar Quarternary NRZ* *(v)* *AMI.*

Solution :

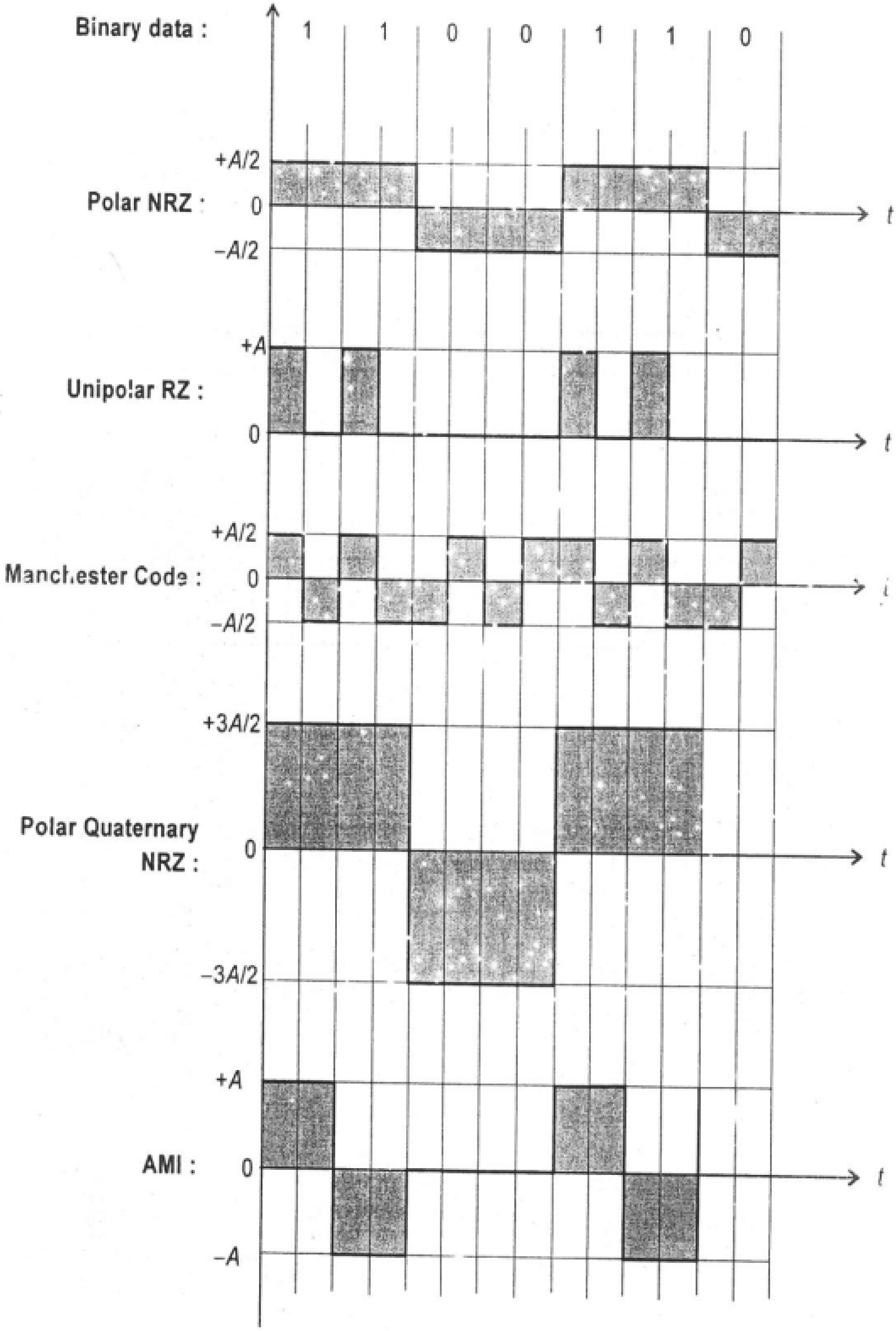


Fig. 11.23

Problem 4 : *Draw the following data waveforms for the bit stream 10111001 :*

(i) AMI *(ii) Manchester* *(iii) Bipolar NRZ*

(iv) Polar Quarternary NRZ *(v) Unipolar RZ.*

Solution :

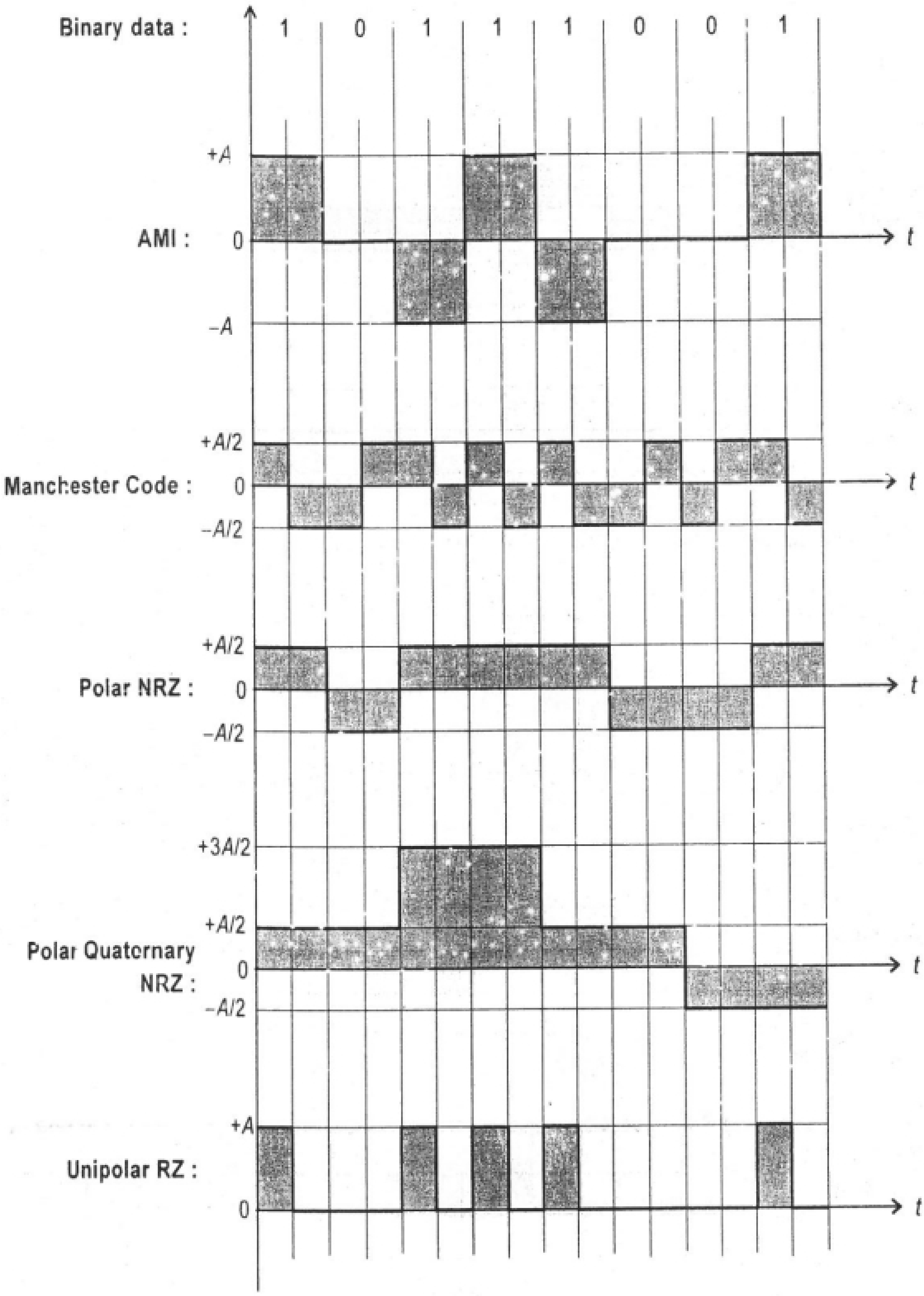


Fig. 11.24

Problem 5 : Draw the following data waveforms for the bit stream 1011000110 :

(i) Polar NRZ *(ii) Unipolar RZ* *(iii) Manchester*

(iv) Polar Quarternary NRZ *(v) AMI.*

Solution :

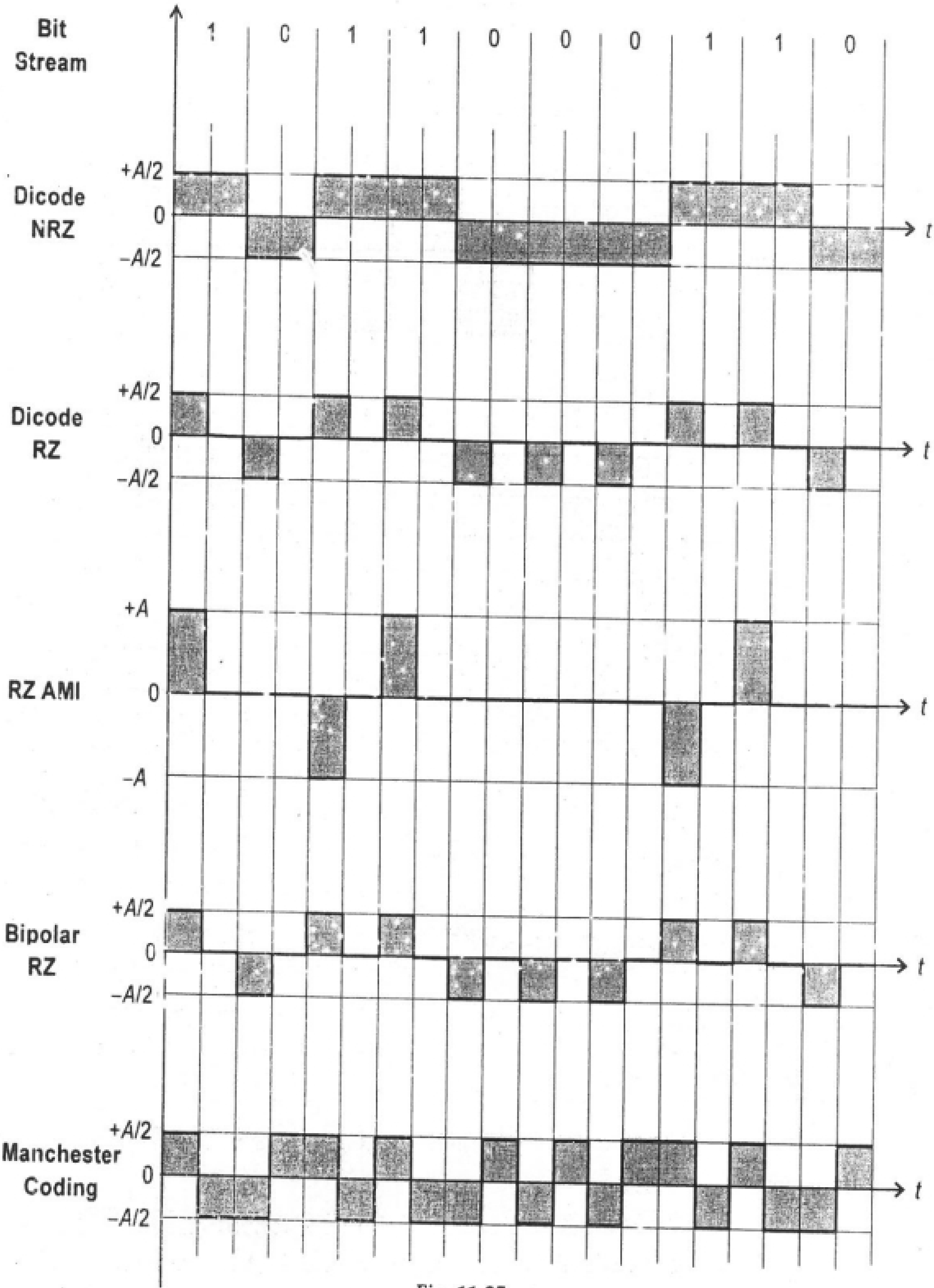


Fig. 11.25

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12 Radio Wave Propagation

Topic	Theory imp	Oral imp
Introduction		
Electromagnetic Waves	☆☆	☆☆
Properties of Radio Waves & Propagation of Waves	☆☆	☆☆☆
Ground Wave Propagation	☆☆☆	☆☆
Direct Wave Propagation	☆☆	☆☆
Sky Wave Propagation	☆☆	
Terms and Definitions	☆	☆☆☆
FAQ's		☆☆☆

12.0 Introduction

- Very first chapter we studied about channel.
- It is a medium between transmitter and receiver.
- In Radio communication the channel is physical space between the transmitting and receiving antenna.
- To understand characteristics of physical space is the objective of this chapter.

12.1 Electromagnetic Waves

Q. *Write short note on EM waves.*

Note : *If the question is asked for 8 marks expl. in the properties also.*

- According to Maxwell's equations, a time-varying electric field generates a magnetic field and vice versa.
- Therefore, as an oscillating (varying with time period) electric field generates an oscillating magnetic field, the magnetic field in turn generates an oscillating electric field, and so on. These oscillating fields together form an electromagnetic wave.

> **Note :** *Oscillating electric field means that electric field vector oscillates, same theory is applicable for Magnetic field.*

- The direction of oscillations of the electric and magnetic fields are perpendicular and the direction of propagation is again perpendicular to these fields. Hence EM waves are transverse in nature.

> **Note :** *If direction of propagation is same as the direction of oscillation then they are called longitudinal waves e.g. wave generated by throwing a stone in pond, else transverse waves.*

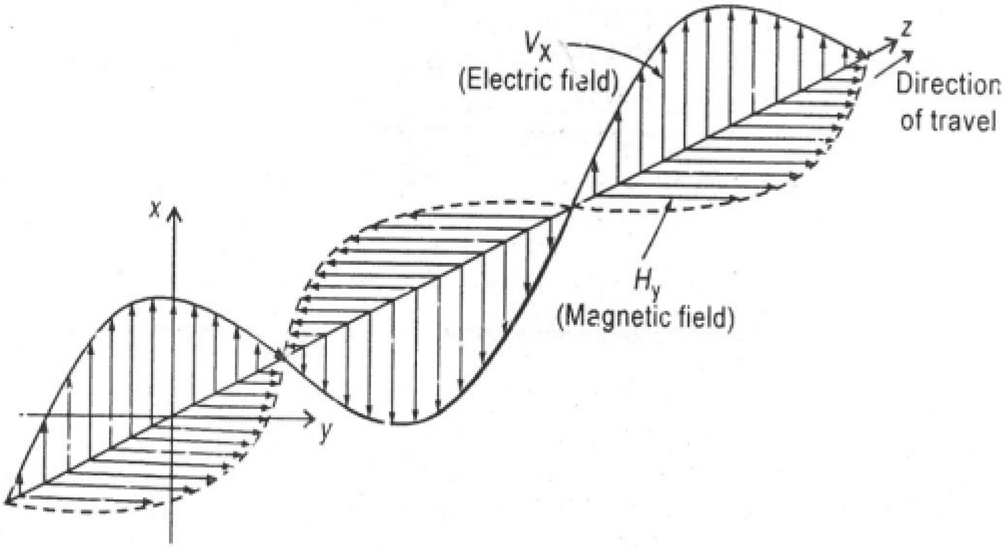


Fig. 12.1 : Transverse EM wave diagram.

- Electromagnetic waves represent the flow of energy in free space in the direction of propagation.
- The speed of prorogation in free space is equal to speed of light i.e. 3×10^8 m/s.
- In other mediums it is given by

$$v = \frac{c}{\sqrt{\epsilon_r}} \quad \text{where } \epsilon_r = \text{Relative permitivity of the medium}$$
$$c = \text{Speed of light}$$

12.1.1 EM Waves Properties

A. Reflection - **(i)** EM waves obey laws of reflection

(ii) Usually gets reflected from a conductor

B. Refraction - **(i)** It is defined as change in direction of ray when it goes from one medium to other medium.

(ii) EM waves obey Snell's law

$$n_1 \sin \theta_1 = n_2 \sin \theta_2$$

(iii) Refractive index

$$n = \frac{c}{v}$$ where c = Speed of light in free space

v = Speed of light in given material

C. Diffraction - (i) It is defined as redistribution of energy within a wave front when it passes through a medium.

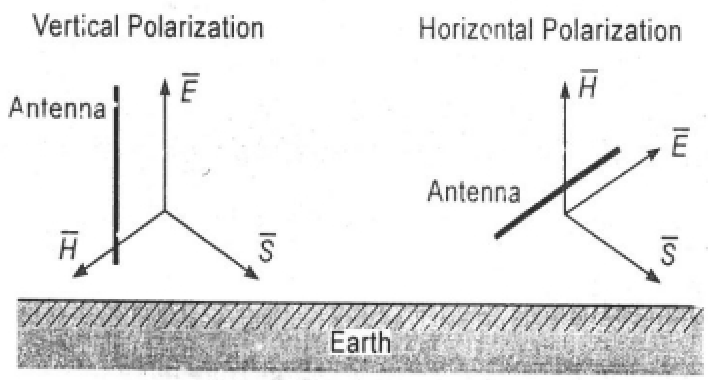
(ii) It causes scatter of energy.

D. Interference -(i) **Interference** is the addition (superposition) of two or more waves those results in a new wave pattern.

(ii) As most commonly used, the term **interference** usually refers to the interaction of waves which are correlated or coherent with each other, either because they come from the same source or because they have the same or nearly the same frequency.

12.1.2 Electromagnetic Polarization

- Polarization is the property of electromagnetic waves, that describes the direction of the electric field vector.
- More generally, the polarization of a transverse wave describes the direction of oscillation in the plane perpendicular to the direction of travel.
- Longitudinal waves such as sound waves do not exhibit polarization, because for these waves the direction of oscillation is along the direction of propagation.



$\overline{E}$ - Electric field vector

$\overline{H}$ - Magnetic field vector

$\overline{S}$ - Pointing vector (indicates direction of energy flow)

Fig. 12.2

12.1.3 Rays and Wave Front

Q. Explain the term - Electromagnetic ray and Wave front.

- A ray is line drawn along the direction of propagation.
- A wavefront shows constant phase of wave.
- A wavefront is formed when points of equal phase on the ray propagated from the same source are joined together.

12.1.4 Power Density

Q. Explain the term - Power Density.

- The rate at which energy passes through a surface area is called *power density*.
- Power density = Energy per unit area for particular instance of time.
- Unit of power density is watts/m^2.
- Power density

$$P = E \times H \text{ w/m}^2$$

E = RMS electric field intensity in volts/m

H = RMS magnetic field intensity in amp-turns/m

12.1.5 Inverse Square and Power Density

Q. Explain the term - Inverse Square Law.

- The power density is reduced by factor of R^2 where R is distance from source, when the ray travels through the medium

$$P = \frac{P_T}{4\pi R^2} \qquad \text{where } P_T = \text{Transmitted power}$$

Above equation is governed by *Inverse Square Law*.

12.2 Properties of Radio Waves and Propagation of Waves

Q. Explain the term - Wave Attenuation and Wave Absorptions.

12.2.1 Wave Attenuation

- The reduction in power density of wave is called *wave attenuation*.
- The wave attenuation is also called *space attenuation* as space attenuation occurs due to spherical spreading of wave.
- The wave attenuation is governed by inverse square law stated above.
- Wave attenuation is expressed in dB.

- It is denoted by γ_a

$$\gamma_a = 10 \log\left(\frac{P_1}{P_2}\right)$$

12.2.2 Wave Absorption

- Earth atmosphere is made up of atoms, molecules and variable substance.
- As EM travels through the space they get absorbed by these substance.
- Practically absent bellow 10 GHz.
- It is frequency dependent.

12.2.3 Propagation

- The process of communication involves the transmission of information from one location to another.
- An electromagnetic wave is created by a local disturbance in the electric and magnetic fields. From its origin, the wave will propagate outwards in all directions.
- If the medium in which it is propagating (air for example) is the same everywhere, the wave will spread out uniformly in all directions.
- it is only the characteristics of the wave and the channel, which determine how the signal will propagate over any significant distance.
- Channel is usually a dielectric medium, it allows passage of EM Waves easily.

> ***Dielectric :*** *A dielectric material is a substance that is a poor conductor of electricity, but an efficient supporter of* ***electrostatic fields****. Examples include porcelain (ceramic), mica, glass, plastics, and the oxides of various metals. Some liquids and gases can serve as good dielectric materials. Dry air is an excellent dielectric. Distilled water is a fair dielectric. A vacuum is an exceptionally efficient dielectric.*

- Characteristics of wave on which how far a ray will go depends on its frequency.

High frequency → High energy → Will go far

- Different frequency wave follow different modes of propagation.
- Signal will go more far if we increase the power but there is limit to which we can increase the power, so we use different modes of propagation for different frequencies.

- If we have a high frequency and small distance we can transmit it directly.
- The three basic modes are
  (a) Ground wave or surface wave
  (b) Sky wave
  (c) Space or line of sight or direct.

Band		Frequency	Wavelength	Propagation via.
VLF	Very Low Frequency	3 - 30 kHz	100 - 10 km	Guided between the earth and the ionosphere.
LF	Low Frequency	3 - 300 kHz	10 - 1 km	Guided between the earth and the D layer of the ionosphere. Surface waves.
MF	Medium Frequency	300 - 3000 kHz	1000 - 100 m	Surface waves. E, F layer ionospheric refraction at night, when D layer absorption weakness.
HF	High Frequency (Short Wave)	3 - 30 MHz	100 - 10 m	E layer ionospheric refraction. F1, F2 layer ionospheric refraction sky wave.
VHF	Very High Frequency	3 - 300 MHz	10 - 1 m	Direct wave.
UHF	Ultra High Frequency	300 - 3000 MHz	100 - 10 cm	Direct wave.
SHF	Super High Frequency	3 - 30 GHz	10 - 1 cm	Direct wave.
EHF	Extremely High Frequency	30 - 300 GHz	10 - 1 mm	Direct wave limited by absorption.

Table 12.1 : Radio frequencies and their primary mode of propagation.

- Relationship between the signal and kind of propagation is given in the above table.
- Do not confuse ground wave as direct wave though in direct wave propagation, waves travel very close to earth.

12.3 Ground Wave Propagation

Q.1. Describe ground wave propagation.

Q.2. Ground wave propagation and its advantages and limitations.

Q.3. Short note on ground wave propagation.

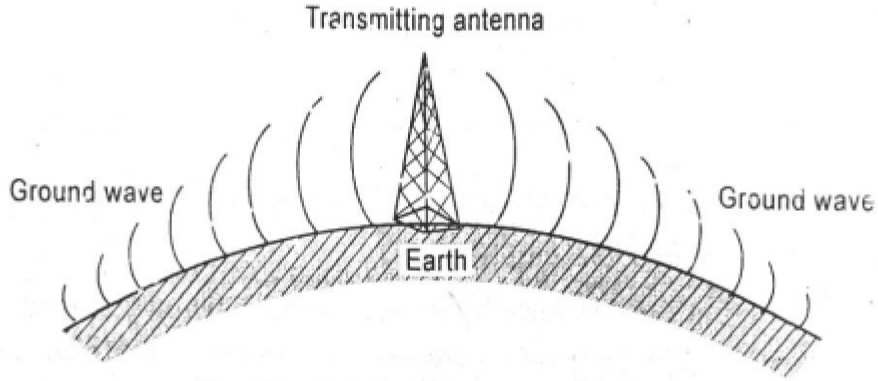


Fig. 12.3 : Ground wave propagation.

- Radio wave that travels along the earth's surface are called *surface wave.*
- The ground wave will actually follow the curvature of the earth and therefore can travel a distance beyond horizon.
- It is vertically polarized to prevent short circuiting of component.
- Changes in terrain have strong effect.

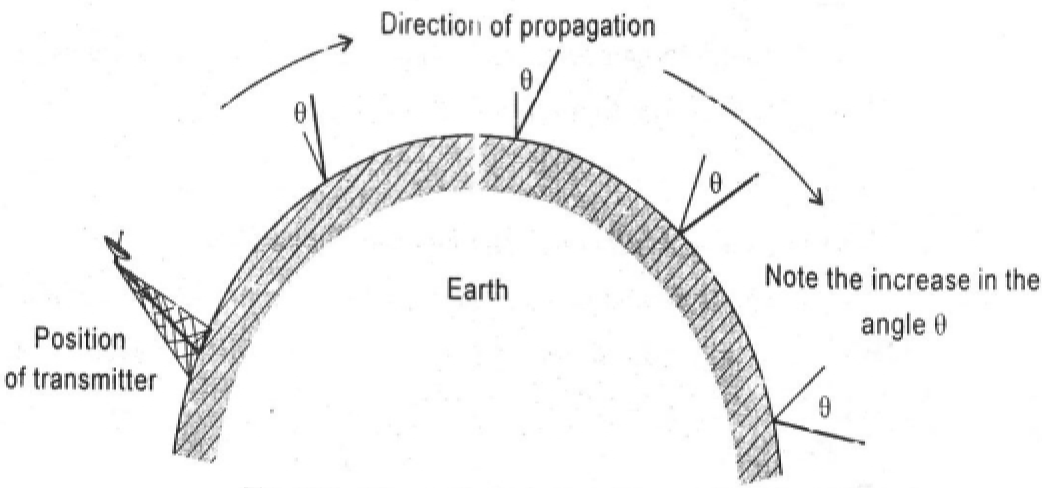


Fig. 12.4 : Attenuation of ground waves.

- Attenuation is directly related to surface impedances.
- More conductive the surface, more attenuated the wave is.
- It propagates better over water.
- Attenuation is related to frequency.
- Loss in power increases with increase in frequency.
- Reception is not affected by daily or seasonal weather changes.

- Extremely Low Frequency (ELF) (30 to 300 Hz) propagation is utilized.
- Field strength

$$E = \frac{120\pi\, h_t\, I}{\lambda d}$$

where h_t = Height of transmitting antenna

λ = Wavelength $= \frac{1}{t}$

d = Distance from transmitter

- Ground waves get attenuated due to following reason
  - **(a)** Energy is lost due to absorption.
  - **(b)** Due to diffraction the wave fronts will gradually tilt over. The tilt angle goes on increasing as the wave progresses and eventually lies down .
  - **(c)** Tilt angle increases with greater rate for higher frequency.
- **Applications**
  - **(a)** Best suitable for Submarines.
  - **(b)** In AM broadcasting.
  - **(c)** Ship communication.
- **Advantages**
  - **(a)** If transmitted with large power any two points on earth can communicate.
  - **(b)** Atmospheric condition do not affect it.
- **Disadvantage**
  - **(a)** Limited range to low frequency can be used above 2 MHz.
  - **(b)** Very tall antennas should be used because antenna height should be $\lambda/4$.
  - **(c)** High transmission power for long distance is required, power in range of MW.

12.4 Space Wave Propagation

Q. Describe space wave or line of sight propagation.

- The simplest and most easily understood way in which a signal travels from one antenna to another is by '*line-of-sight*' propagation.
- Line-of-sight propagation requires a path where both antennas are visible to one another and there are no obstructions. VHF and UHF communication typically uses this path.

- Unless you are VERY close to your destination, you need to keep the antenna as high as possible. Because radio waves follow a straight-line in this mode, they simply go off into space as the curvature of the earth causes the ground to drop away beneath the radio waves.
- As we elevate the antenna, the distance to the horizon gets further and further away. With enough power to reach the other antenna and a high enough antenna to see it, we can talk without problems.
- VHF repeaters are usually mounted on high buildings or mountain tops for this very reason.
- When you are operating with a small VHF hand held, your signal must be able to travel in a straight-line to the repeater or your signal will be lost to someone beyond line-of-sight.
- This is probably the most common of the radio propagation modes at VHF and higher frequencies.
- *Ground plane reflection* effects are an important factor in VHF line of sight propagation. The interference between the direct beam line-of-sight and the ground reflected beam often leads to an effective inverse fourth-power law for ground-plane limited radiation.

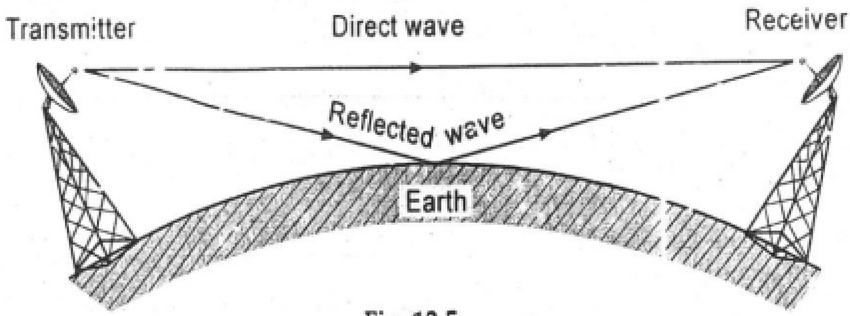


Fig. 12.5

- Examples would include propagation between a satellite and a ground antenna or reception of television signals from a local TV transmitter. Because radio signals can travel through many non-metallic objects, waves can be picked up through walls.
- Space waves are horizontally polarized because noise created by electrical systems are vertically polarized and hence noise interference is reduced
- **Applications**

  (a) TV broadcasting

  (b) FM radio broadcasting

  (c) Microwave links

  (d) Satellite communication.

12.5 Sky Wave Propagation

Q.1. Write a short note on ionospheric propagation.

Q.2. Write a short note on sky wave propagation.

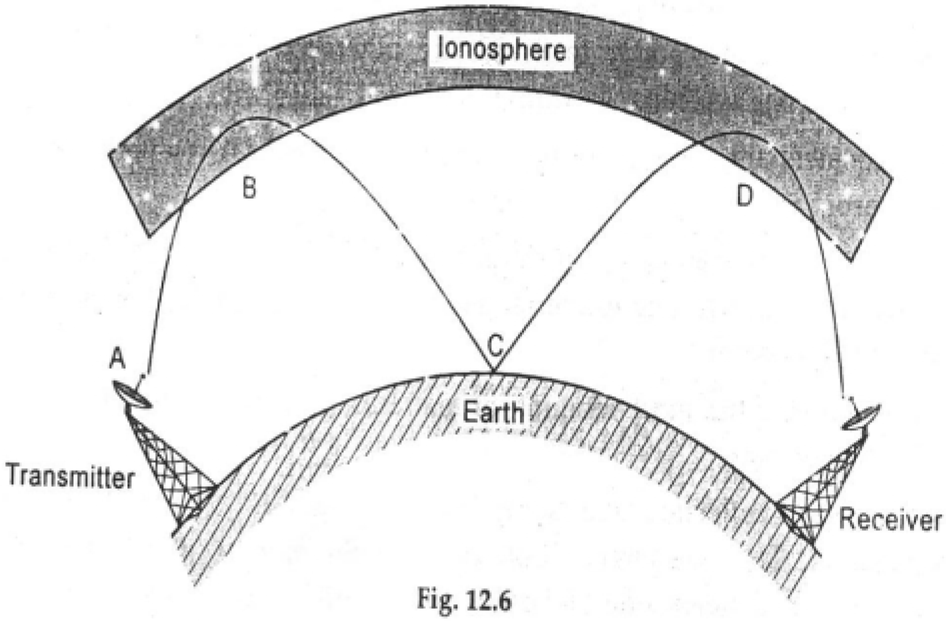


Fig. 12.6

- Radio waves are radiated from the transmitting antenna in a direction toward the ionosphere. This wave is also called *sky wave.*
- Sky wave strike the ionosphere, it is reflected back to ground, it strikes the ground, and reflected back toward the ionosphere, and so on until it reaches the receiving antenna.

> **How Does Signal Bend ?**
>
> *You can picture the ionosphere like a bunch of glass panes stacked on top of one another. As you probably remember from high school physics, light passing through a different medium (like water or glass) is bent. Imagine that as the signal, arriving at 45 degrees, passes through a layer, it bends over just a little.*
>
> *If it bends 10 degrees every time it enters a pane of glass, then after 9 panes, it will have turned by 90 degrees and will be heading down towards the ground at a 45 degree with respect to the glass panes.*
>
> *The ionosphere works somewhat similarly. The bending is gradual but increasing as the signal keeps going into the ionosphere until it leaves either at the top going towards outer space (because it wasn't bent enough to come back) or it leaves at the bottom going back to the earth.*

- Heavily depends on atmospheric conditions and time of day

12.5.1 Layers of Ionosphere

Q. Describe various layers of ionosphere.

- Among all the layers, the layer important for our understanding is ionosphere.
- The **ionosphere** is the uppermost part of the *atmosphere*, distinguished because it is *ionized* by solar radiation.
- It plays an important part in *atmospheric electricity* and forms the inner edge of the *magnetosphere*.
- Solar radiation, acting on the different compositions of the atmosphere with height, generates layers of ionization. The different layers are :

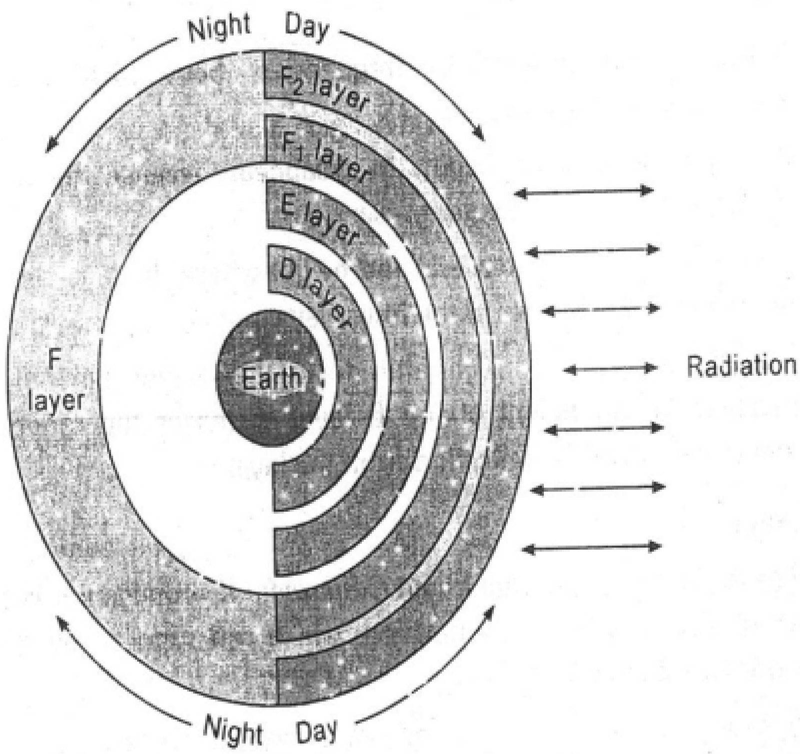


Fig. 12.7

- **D Layer**
  - **(a)** The *D layer* is the innermost layer, 50 km to 90 km above the surface of the Earth.
  - **(b)** Ionization here is due to *Lyman series*-alpha hydrogen radiation
  - **(c)** Recombination is high in this layer, thus the net ionization effect is very low and as a result high-frequency (HF) *radio waves* aren't reflected by the D layer.

(d) The D layer is mainly responsible for *absorption of HF radio waves*, particularly at 10 MHz and below, with progressively smaller absorption as the frequency gets higher.

(e) The absorption is small at night and greatest about mid-day. The layer reduces greatly after sunset, but remains due to galactic cosmic rays.

(f) A common example of the D layer in action is the disappearance of distant AM broadcast band stations in the daytime.

- **E Layer**

(a) The *E layer* is the middle layer, 90 km to 120 km above the surface of the Earth.

(b) This layer can only reflect radio waves having frequencies less than about 10 MHz.

(c) It has a negative effect on frequencies above 10 MHz due to its partial absorption of these waves.

(d) At night the E layer begins to disappear because the primary source of ionization(sun) is no longer present.

(e) This results in an increase in the height where the layer maximizes because recombination is faster in the lower layers.

(f) The changes in the high altitude neutral winds also plays a role. The increase in the height of the E layer increases the range to which radio waves can travel by reflection from the layer.

- **E_s Layer**

(a) The E_s layer or sporadic E-layer. *Sporadic E propagation* is characterized by small clouds of intense ionization, which can support radio wave reflections from 25 - 225 MHz.

(b) Sporadic-E events may last for just a few minutes to several hours and make *radio amateurs* very excited, as propagation paths which are generally unreachable, can open up.

(c) There are multiple causes of sporadic-E that are still being pursued by researchers. This propagation occurs most frequently during the summer months with major occurrences during the summer, and minor occurrences during the winter.

Why Sporadic ?

Sporadic refers to something that occurs at random without any known reasons. Just as the formation of E_S layer and hence called sporadic E- layer.

- **F Layer**

  (a) The *F layer* or region, also known as the Appleton layer, is 120 km to 400 km above the surface of the Earth. It is the top most layer of the ionosphere.

  (b) The F region is the most important part of the ionosphere in terms of HF communications.

  (c) The F layer combines into one layer at night, and in the presence of sunlight (during daytime), it divides into two layers, the F_1 and F_2.

  (d) The F layers are responsible for most sky wave propagation of radio waves, and are thickest and most reflective on the side of the Earth facing the Sun.

12.5.2 Virtual Height, Critical Frequency, MUF and Skip Distance

Q.1. Define the following terms :

(i) Skip distance

(ii) MUF

(iii) Virtual height.

Q.2. Define the following terms in relation with sky wave propagation :

(i) Virtual Height

(ii) Critical Frequency

(iii) MUF

(iv) Skip Distance.

(a) Virtual Height

- It is a height of surface from which the refracted wave which returns back appears to be reflected.
- When we send a wave towards the atmosphere it gradually bends down due to refraction.
- We have to find height of the surface such that reflected wave and refracted wave have the same end path.
- Virtual height is calculated because calculations become simple for finding the angle at which a wave has to be transmitted to reflect it back to a particular point.
- Virtual height is different for different layers.

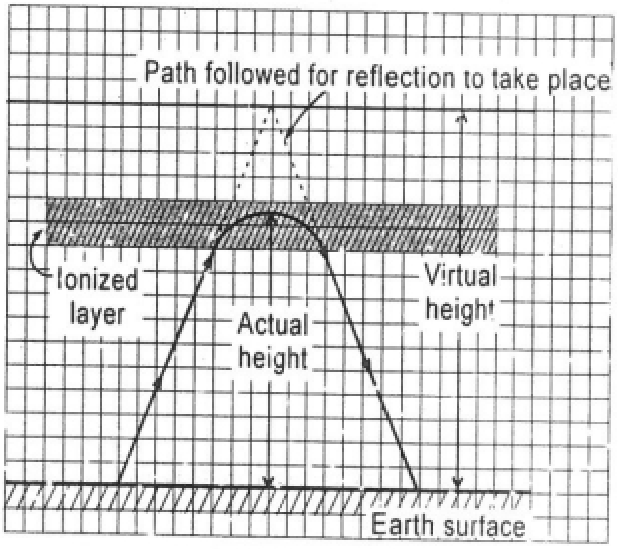


Fig. 12.8 : Virtual height of an ionized layer.

(b) Critical Frequency f_c

- The maximum frequency that can be returned by the ionosphere when the radio waves are vertically incident on the ionosphere (transmitted straight up or angle of transmission is 90) is called the *critical frequency.*

(c) MUF

- MUF is defined maximum frequency that will return back to earth for certain value of the angle of incidence θ.
- MUF is defined for particular θ rather than defining it at normal as in case of critical frequency.
- MUF tells that we can not operate on higher frequency if we increase the angle θ.
- Relation between

$$\text{MUF} = \frac{f_c}{\cos\theta}$$

(d) Skip Distance

- The skip distance is defined as *the shortest distance from transmitter, measured along the surface of the earth at which sky wave of fixed frequency returns back to earth.* The relation between skip distance and θ is illustrated in *figure 12.9.*

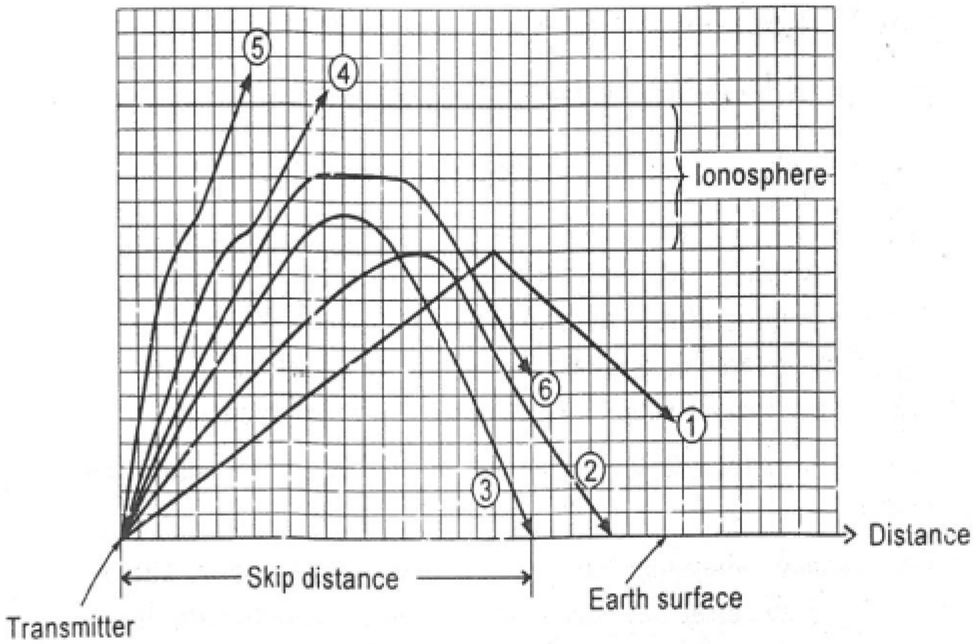


Fig. 12.9 : Effect of variation in θ at constant frequency.

- As the angle θ increases distance reaches a minimum value know as SKIP DISTANCE.
- We further increase the angle, distance goes on increasing → Ray 6.
- And after a particular θ, the ray does not return back to earth → Ray 4,5.

(e) Formula for MUF, Critical Frequency f_c and Virtual Height

$$\text{MUF} = \frac{f_c}{\cos\theta}$$

$$\cos\theta = \frac{H'}{\left[H' + \left(\frac{d}{2}\right)^2\right]^{1/2}}$$

H' = Virtual height

d = Distance between transmitter and receiver

> **Note :** *the following terms are given only for viva point of view they have very less importance from theory point of view except a few which have been underlined and mentioned. Though if paper comes difficult then these terms could come handy in theory also.*

12.6 Terms and Definition

(a) Blind Zone : The blind zone is the area around a radio station which cannot normally be worked by either ground waves or normal ionospheric sky waves. Usually stations in the blind zone can only be worked via intermittent backscatter propagation. This zone is also called the *skip zone* by the US Military.

(b) Troposphere Scatter :

Q. Short note on Tropospheric Scatter Propagation.

(1) It is the only form of propagation that is directly influenced by the surface weather of the earth.

(2) Our troposphere (0-10km altitude) is composed of layers of air having different temperatures and moisture contents. When a sharp transition, called an inversion, appears between a cold dry layer and a warm moist layer of air, this transition causes refraction of radio waves.

(3) This is analogous to the refraction caused by the transition between water and air.

(4) For instance, when you put a stick into the water, it looks like it is bent. This same type of refraction occurs when a radio wave travels through a climate inversion; if the inversion is strong enough, radio waves can be refracted back to the surface of the earth after travelling significant distances (up to several hundred kilometers on the 6 m band).

(5) Finally, this propagation effect is seen most often in the VHF and UHF bands, especially the 6 m band.

(c) Ducting :

(1) On rare occasions, two or more inversions may appear at different altitudes. Sometimes certain radio waves can be transported between these two inversions. Therefore, this type of propagation is called *ducting* (or *tunneling*).

(2) Records of over 2500 km have been set due to such ducting on VHF and UHF. Unfortunately, the effect is usually confined to 2 m, but it can occur as high as 1.2 GHz (usually along frontal systems), and it almost never occurs below frequencies of 50 MHz.

(3) When ducting does occur on these frequencies, communication distances are typically in the range of 400 km.

(4) Inversions usually develop under the influence of high pressure weather systems when there is very little air movement.

(5) Also, low pressure systems may produce an inversion when a cold air mass collides with a warmer air mass (called a *frontal system in meteorology*).

(6) Inversions that occur along frontal systems support propagation along a line parallel to the weather front, and radio amateurs using frontal inversion often point their antennas parallel to the frontal system to take advantage of this form of propagation.

12.7 Frequently Asked Questions

Q.1. Why better reception during night ?

Ans. There is no loss due to absorption as there are no E and D layer in the night.

Q.2. What path do radio waves usually follow from a transmitting antenna to a receiving antenna at VHF and higher frequencies ?

Ans. HF and UHF frequencies usually use line-of-sight propagation because of the frequency.

Q.3. What type of propagation involves radio signals that travel along the surface of the Earth ?

Ans. Ground-wave propagation

Q.4. What is the meaning of the term ground-wave signal ?

Ans. Signals that travel along the surface of the earth

Q.5. Two Amateur Radio stations a few miles apart and separated by a low hill blocking their line-of-sight path are communicating on 3.725 MHz. What type of propagation is probably being used ?

Ans. Ground-wave

Q.6. When compared to sky-wave propagation, what is the usual effective range of ground-wave propagation ?

Ans. Much smaller

Q.7. What type of propagation uses radio signals reflected back to earth by the ionosphere ?

Ans. Sky-wave

Q.8. What type of radio wave propagation makes it possible for amateur stations to communicate long distances ?

A. Direct-inductive propagation

B. Knife-edge diffraction

C. Ground-wave propagation

D. Sky-wave propagation

Ans. How far do you want to get ? Very close, you'll use straight-line. Over the horizon, you'll use ground-wave. But if you want your DXCC or WAS, you'll HAVE to use sky-wave propagation.

Q.9. Compare Ground wave and Space wave propagation ?

Ans.

Sr.	Ground Wave	Space Wave
(1)	Waves are propagated close to earth's surface.	Waves travel in straight line.
(2)	Attenuation is more.	Less attenuation.
(3)	Range is upto 2 MHz only.	Range is from (30 MHz to 300 MHz).
(4)	Area covered is more.	Area covered is less.
(5)	Used for local broadcasting.	Used for satellite communication.

Table 12.2 : Comparison between ground wave and space wave propagation.

Q.10. Write short note : Spherical Wave Front.

Ans. The point source produces a spherical wave front of radius r as shown in *figure 12.10*.

All the points at distance r from the source lie on the surface of the sphere and have equal power densities.

The electromagnetic waves spread uniformly in all the directions in space.

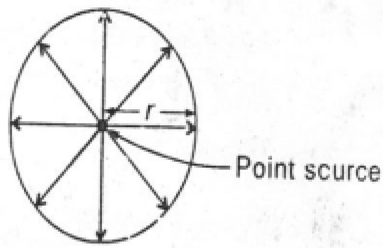


Fig. 12.10

We can now find out power density at a point at distance r from the source as :

$$P = \frac{P_T}{4\pi r^2}$$

P_T = Total transmitted power.

$4\pi r^2$ = Area of the sphere.

Q.11. Describe the following : The atomospheric conditions that cause electromagnetic refractions.

Ans. The main cause of electromagnetic refraction in atmosphere is the change of density of the atmosphere.

The causes of change in density of the atmosphere can be

(1) **Change in Temperature** : Density is inversely proportional to