

**GOVERNMENT OF TAMILNADU
DIRECTORATE OF TECHNICAL EDUCATION
CHENNAI – 600 025**

STATE PROJECT COORDINATION UNIT

Diploma in Electronics And Communication Engineering

**Course Code: 1040
M – Scheme**

**e-TEXTBOOK
on
COMMUNICATION ENGINEERING**

**for
IV Semester DECE**

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34042 - COMMUNICATION ENGINEERING DETAILED SYLLABUS

UNIT- 1

Networks, Antenna and Propagation

Networks: Symmetrical and asymmetrical networks, characteristic impedance and propagation constant.

Equalizer: Definition, types and applications.

Attenuator: Definition, types - symmetrical T and Pi attenuators- simple problems – applications.

Filters: Definition, types – circuit elements and cut off frequencies of LPF, HPF and BPF - simple problems applications.

Antennas: Definition-types of antenna: Mono pole and dipole antenna, directional and Omni directional Antenna, Dipole arrays, Yagi antenna, parabolic antenna- Antenna parameters: directive gain, directivity, radiation pattern and polarization-applications.

Propagation: Ground wave propagation, sky wave propagation and space wave propagation.

UNIT - 2

Introduction to Modulation and Amplitude Modulation

Introduction to Modulation: Definition- Need for modulation- types of modulation - Frequency spectrum - relationship between wavelength and frequency.

Amplitude modulation: Definition - Simple signal diagram for amplitude modulation, Expression for amplitude modulation, expression for modulation index –sidebands: DSB, SSB and VSB.

AM Transmitter: Types of transmitters: high level AM transmitter, low level AM transmitter and SSB transmitter.

AM Receiver: Types of receiver: TRF receiver, super heterodyne receiver and SSB receiver. - Selection of IF-AGC types: simple and delayed AGC.

UNIT - 3

Frequency and Pulse Modulation

Frequency modulation: Definition-Simple signal diagram for frequency modulation, Expression for frequency modulation, expression for modulation index.

FM Transmitter: Types of transmitters: Direct FM transmitter, Indirect FM transmitter and stereophonic FM transmitter.

FM Receiver: stereophonic FM receiver-AFC. Comparison of FM and AM.

Pulse modulation: Definition- Types: Generation and detection of PAM, PWM, PPM, PCM & DPCM.

UNIT - 4

Audio systems

Microphones: Definition-Construction and performance of the following microphones: carbon, condenser, piezoelectric, moving coil and velocity ribbon.

Loud speakers: Definition-Constructional details of dynamic cone type, Horn type and electro-static loud speakers, woofer, midrange and tweeter, cross-over network. Surround-sound systems.

Audio recording and reproduction: Compact disc system- MP3 system - DVD system - stereophonic system - Hi-Fi system principles-DTS.

UNIT - 5

Video systems

Monochrome Television: Scanning principles - synchronization - aspect ratio - composite video signal - TV broadcasting standards. TV transmitter - TV receiver.

Color TV: Principles of color transmission and reception color CCD camera, LCD, LED display unit – plasma display - Principles of Handy cam, CCTV and cable TV.

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UNIT - I

NETWORKS, ANTENNA AND PROPAGATION

LEARNING OBJECTIVES

On completion of this unit, the students must be able to....

- Understand the various network concepts
- Understand the principles of working of antennas
- Understand the theory of Radio wave Propagation

1.0 NETWORKS

An electric network is any interconnection of electric circuit elements or branches. In a complicated network the circuit elements may be impedances and sources, where each branch may include R, L, C or other types of elements. Such a network having two distinct pairs of terminals is called a four-terminal network.

Fig 1.1 shows a four-terminal network

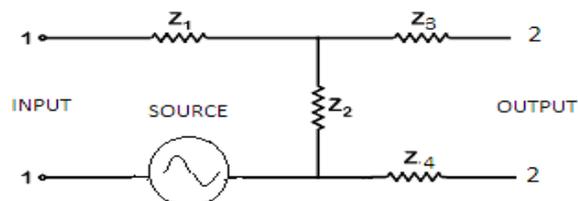


Fig.1.1 Four terminal Network

1.1 SYMMETRICAL NETWORK

Symmetrical network is a network in which the electrical properties are unaffected by interchanging the input and output terminals.

Fig 1.2 shows one type of symmetrical network.

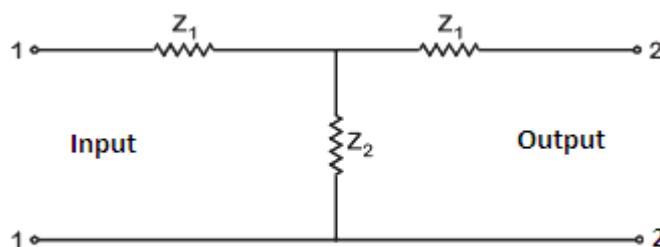
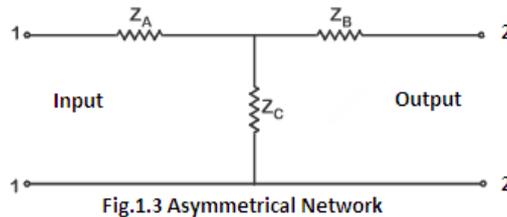


Fig.1.2 Symmetrical Network

1.2 ASYMMETRICAL NETWORK

An Asymmetrical network is a network in which the electrical properties are affected by interchanging the input and output terminals.

Fig 1.3 shows one type of Asymmetrical network



Comparison between Symmetrical and Asymmetrical Networks

Sl. No	Symmetrical Network	Asymmetrical Network
1.	Input and output terminal can be interchanged	Input and output terminal cannot be interchanged
2.	While interchanging the terminals, electrical characteristics will not be changed.	While interchanging the terminals, electrical characteristics will be changed.
3.	Terminal impedance on both sides is equal.	Terminal impedance on both sides is not equal.
4.	It has two characteristics, 1.characteristic impedance, Z_0 2. Propagation constant, P	It has four characteristics, 1.Image impedances, Z_{i1}, Z_{i2} 2.Iterative impedance, Z_{t1}, Z_{t2} 3.Image transfer constant 4.Iterativetransfer constant
5.	It may be balanced or unbalanced network	It is always an unbalanced network.

1.3 CHARACTERISTICS OF SYMMETRICAL NETWORKS

Symmetrical networks have two electrical characteristics namely,

- (i) Characteristic impedance (Z_0)
- (ii) Propagation constant (P)

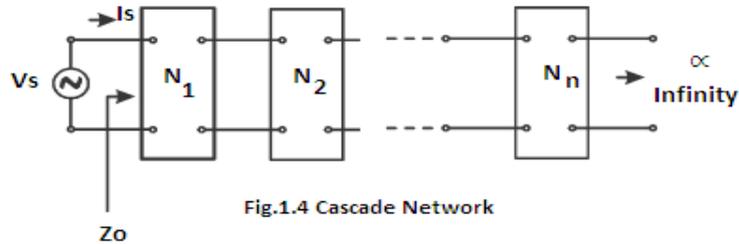
1.3.1 Characteristic impedance, Z_0

When an infinite number of symmetrical networks are interconnected in cascade manner as shown in Fig 1.4, the input impedance measured at the sending end will have a finite value and it depends on the structure of the network. This impedance is called characteristic impedance of the network. It is denoted by ' Z_0 '

i.e.)
$$Z_o = \frac{V_s}{I_s}$$

Where, V_s - sending end voltage

I_s - sending end current



Definition: Characteristic Impedance

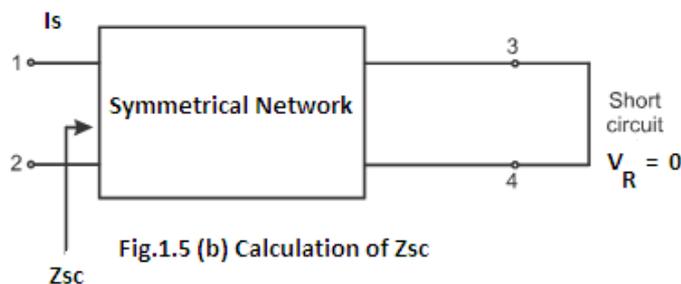
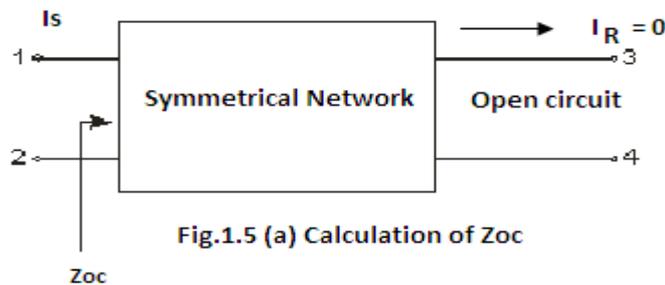
It is the impedance which when connected to one pair of terminals produces the same impedance at the other pair of terminals'

1.3.1.1 Methods of Calculating Characteristic Impedance, Z_o .

1. The characteristic impedance can be calculated from the components of the network.
2. It can be computed from the values of open circuit impedance Z_{oc} and short circuit impedance Z_{sc} .

i.e.)
$$Z_o = \sqrt{Z_{oc} \cdot Z_{sc}}$$

Fig 1.5 (a) and (b) illustrate the measurement of Z_{oc} and Z_{sc} respectively.
 Z_{oc} is the impedance of a network when its output terminals are open circuited
 Z_{sc} is the impedance of a network when its output terminals are short-circuited.



1.3.2 Propagation Constant

Propagation constant of a symmetrical network is defined as the natural logarithm of the ratio of the input to output current and is denoted by 'P'



Fig.1.6 Calculation of Propagation constant

Referring to Fig 1.6

$$P = \log_e \frac{I_S}{I_R}$$

$$e^P = \frac{I_S}{I_R}$$

In general, propagation constant P is a vector quantity and can be represented by,

$$P = \alpha + j \beta$$

Where, α is called attenuation constant

β is called phase constant

1.3.2.1 Computation of Propagation constant

1. Propagation constant can be computed by finding out the ratio of the input to the output current.
2. Alternatively P can also be calculated by finding the open circuit impedance Z_{OC} and short circuit impedance Z_{SC} .

i.e.)

$$\tanh P = \sqrt{\frac{Z_{OC}}{Z_{SC}}}$$

(or)

$$P = \tanh^{-1} \sqrt{\frac{Z_{OC}}{Z_{SC}}}$$

1.4 CHARACTERISTICS OF ASYMMETRICAL NETWORKS

Asymmetrical networks have the following four characteristics.

- (i) Image impedances Z_{i1} and Z_{i2}
- (ii) Iterative impedances Z_{t1} and Z_{t2}
- (iii) Image transfer constant
- (iv) Iterative transfer constant

1.4.1 Image Impedance (Z_{i1} and Z_{i2})

Image impedances of a network are defined as the impedances which simultaneously terminate each pair of terminals of a network in such a way that at each pair of terminals the impedance in both directions are equal and is normally denoted by Z_{i1} and Z_{i2} .

- ❖ Thus, the impedance at terminals 1 and 2 of the network is $200\ \Omega$ when terminals 3 and 4 are terminated in $500\ \Omega$.
- ❖ Input impedance of the terminals 3 and 4 is $500\ \Omega$ when 1 and 2 are terminated in $200\ \Omega$ as shown in Fig 1.7.
- ❖ Here $200\ \Omega$ and $500\ \Omega$ are called image impedances of the asymmetrical network.

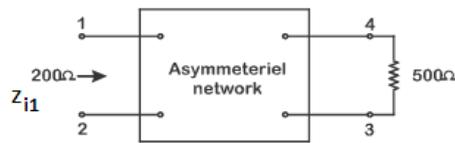


Fig.1.7 (a) Calculation of Z_{i1}

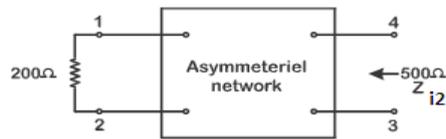


Fig.1.7 (b) Calculation of Z_{i2}

1.4.2 Iterative Impedance (Z_{t1} and Z_{t2})

Iterative impedances are defined as the impedances which will terminate the other pair of terminals in such a way that the impedance measured at the first pair of terminals is equal to the terminating impedance.

Thus, iterative impedance is the value of the impedance measured at one pair of terminals of the network when the other pair of terminals is terminated with the impedance of the same value and is normally denoted by Z_{t1} and Z_{t2} .

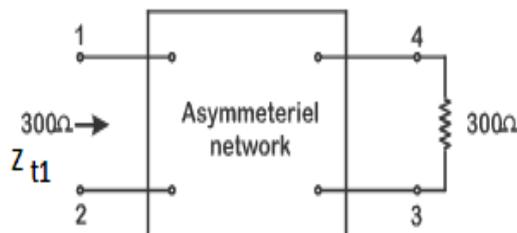


Fig.1.8 (a) Calculation of Z_{t1}

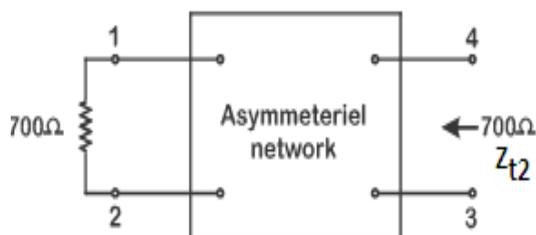


Fig.1.8 (b) Calculation of Z_{t2}

- ❖ For example, the impedance looking into terminals 1, 2 of Fig.1.8 (a) is 300Ω , when terminals 3 and 4 are terminated in 300Ω
- ❖ The impedance looking into terminals of 3 and 4 Fig.1.8 (b) is 700Ω , when the terminals 1 and 2 are terminated in 700Ω .

Image transfer constant

The image transfer constant is defined as one half the natural logarithm of the vector ratio of the input volt-ampere to the output volt-ampere, provided the network being terminated by its image impedances as shown in the fig. 1.9

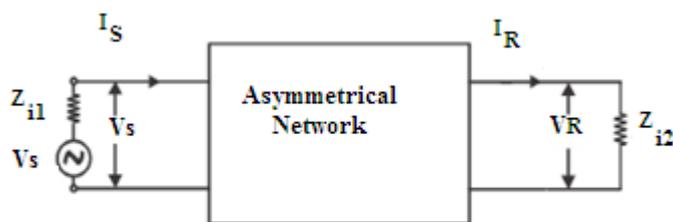


FIG.1.9

$$\text{From the definition, } \theta_i = \frac{1}{2} \log_e \left[\frac{V_s I_s}{V_R I_R} \right]$$

Iterative transfer constant, θ_t

The iterative transfer constant θ_t may be defined as the natural logarithm of the ratio of sending end voltage or current to the receiving end voltage or current, provided the network is terminated by its iterative impedance.

$$\begin{aligned} \therefore \theta_t &= \log_e \left[\frac{V_s}{V_R} \right] = \log_e \left[\frac{I_s}{I_R} \right] \\ \Rightarrow e_{\theta_t} &= \frac{V_s}{V_R} = \frac{I_s}{I_R} \end{aligned}$$

Since iterative transfer constant is also a vector quantity, it can be expressed as $\theta_t = A_t + jB_t$

Where, $A_t \rightarrow$ Iterative attenuation constant

$B_t \rightarrow$ Iterative phase constant.

1.5 EQUALIZERS

Equalizers are correcting networks designed to counteract the attenuation or phase distortion occurring in any part of the circuit.

1.5.1 Classification or types of Equalizers

Based on Function

1. Attenuation equalizer or amplitude equalizer
2. Phase or delay equalizer

Based on Connection

1. Series equalizer
2. Shunt equalizer

Based on Terminals

1. Two terminal equalizers
2. Four terminal equalizers

Based on change in impedance

1. Constant resistance equalizer

1. Attenuation Equalizer

The networks that counteract the attenuation are called attenuation or amplitude equalizers.

2. Phase or delay equalizer

The networks that counteract the phase or delay distortion are called phase or delay equalizers.

3. Series equalizer and Shunt Equalizer

When the equalizer is connected in series with the sending or receiving end of the network, it is called series equalizer. It is shown in Fig 1.10 (a).

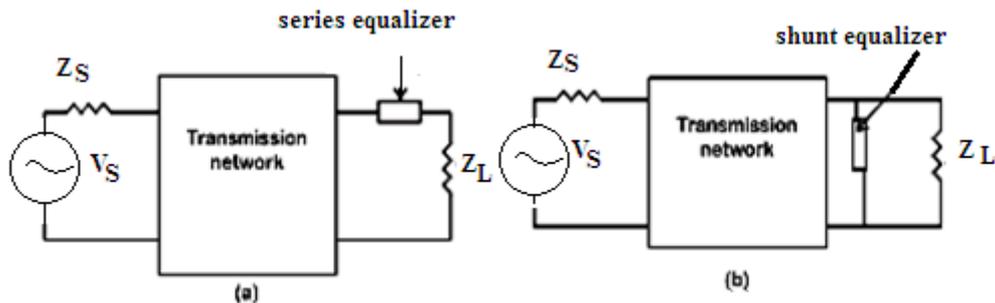


Fig.1.10

When the equalizer is connected in parallel with the sending or receiving end of the network, it is called Shunt equalizer. It is shown in Fig.1.10 (b).

4. Two terminal Equalizers

When the equalizer has only two terminals as shown in Fig.1.11, it is called two terminal equalizer.

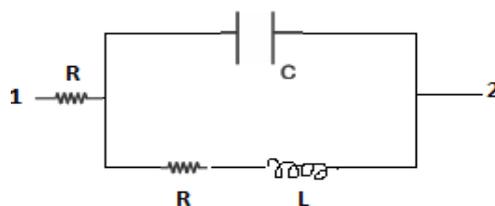


Fig.1.11 Two terminal equalizer

5. Four terminal equalizers:

When the equalizers have two pairs of terminals then they are called four terminal equalizers.

1.5.2 Applications of equalizers

Attenuation equalizers are largely used in,

1. Transmission of speech
2. Recording of speech
3. Reproduction of speech.

Phase or delay equalizers are largely used in,

1. Facsimile telegraphy.
2. Transmission of Television signals over lines.

1.5.3 Constant resistance equalizers

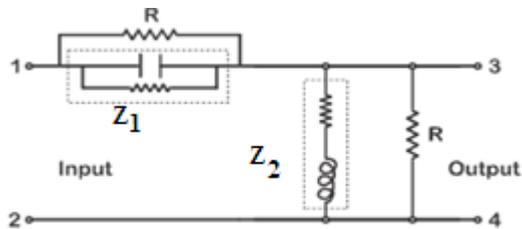


Fig.1.12 Four terminal equalizer

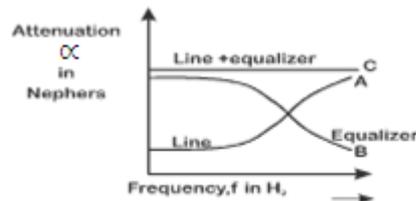


Fig.1.13 Frequency response of equalizer

Definition: Constant resistance equalizer

The four terminal equalizer whose series and shunt arm impedances are inverse of each other with respect to constant resistance is called constant resistance equalizer.

- ❖ It always presents a constant input and output resistance whatever be the frequency.
- ❖ A simple constant resistance equalizer is shown in figure 1.12.
- ❖ In this the series impedance Z_1 is the inverse of the shunt impedance Z_2 that is,

$$Z_1 Z_2 = R^2$$

- ❖ If R is made equal to the characteristic impedance (Z_0) of the line then there will be no reflection loss in the line.
- ❖ The equalizer will then have characteristics as shown in fig 1.13.
- ❖ The graph is drawn between attenuation (α) and frequency (f).
- ❖ At low frequencies, the line attenuation is very low when compared to high frequencies (curve A)
- ❖ The Equalizer increases, the attenuation at low frequencies and decreases attenuation at high frequencies (curve B).
- ❖ The curve(C) shows that attenuation is almost constant at all frequencies.

1.6 ATTENUATORS

Resistive networks designed to introduce a known amount of loss are called Attenuators.

1.6.1 Types of Attenuators

Basically attenuators are classified into two types,

1. **Symmetrical attenuators**
2. **Asymmetrical attenuators**

Attenuators may be,

1. **Fixed**
2. **Variable**

Attenuators may be,

1. **Resistive**
2. **Capacitive**

1.6.1.1 Symmetrical Attenuators

- ❖ Symmetrical attenuator is a four terminal network.
- ❖ Its electrical properties are unaffected by interchanging the input and output terminals.
- ❖ Symmetrical attenuator is used between equal values of load and source resistances.
- ❖ Since these are having resistive components, these will introduce no phase shift, hence $\beta = 0$.
- ❖ Hence, the propagation constant, $P = \alpha + j\beta = \alpha$ only.

Fig 1.14 shows symmetrical T and π attenuators which have been introduced between two equal impedance (Resistances), R_0 , where R_0 is called characteristic resistance of symmetrical attenuator.

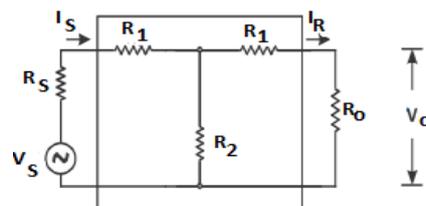


Fig.1.14 Symmetrical T Attenuator

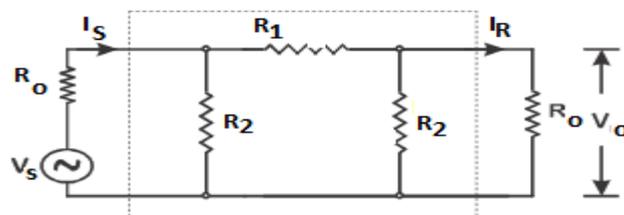


Fig.1.15 Symmetrical π attenuator

In Fig 1.15, R_1 and R_2 are the elements of symmetrical π attenuator I_S and I_R are the input and output currents respectively,

$$\text{Input power, } P_1 = I_S^2 R_0$$

$$\text{Output power, } P_2 = I_R^2 R_0$$

\therefore Attenuator

$$= \frac{P_1}{P_2} = \frac{I_S^2 R_0}{I_R^2 R_0}$$

$$= \frac{I_S^2}{I_R^2}$$

$$\frac{P_1}{P_2} = \left[\frac{I_S}{I_R} \right]^2$$

Attenuation = N^2 where,

Attenuation is usually given in decibels (dB)

$$N = \frac{I_S}{I_R} \text{ or } N = \sqrt{\frac{P_1}{P_2}}$$

Decibel (dB): Decibel is defined as the ten times common logarithm of the ratio of input power to the output power

Therefore,

$$\text{Attenuation in dB, } D = 10 \log_{10} \frac{P_1}{P_2}$$

$$= 10 \log_{10} N^2$$

$$= 20 \log_{10} N$$

$$D = 20 \log_{10} N$$

(or)

$$\text{Attenuation in Nepers, } N = \text{Antilog}_{10} \left[\frac{D}{20} \right]$$

Neper (N): Neper is defined as the natural logarithm of the ratio of input voltage or current to the output voltage or current provided the network is properly terminated with Z_0 .

$$\text{Attenuation in dB} = 8.686 \times \text{Attenuation in Neper.}$$

1.6.2 Elements of Symmetrical T attenuator

A symmetrical T attenuator is shown in Fig 1.16.

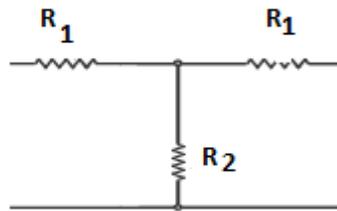


Fig.1.16 Element of Symmetrical T attenuator

Series arm of symmetrical attenuator,

$$R_1 = R_0 \left[\frac{N-1}{N+1} \right] \dots\dots\dots (1)$$

Shunt arm element of symmetrical T attenuator is

$$R_2 = R_0 \left[\frac{2N}{N^2-1} \right] \dots\dots\dots (2)$$

Equations (1) and (2) are the design equations of symmetrical T attenuator. Given the values of N and R_0 , series and shunt arm elements can be calculated from these equations. N can be calculated if the loss in dB (D) which is to be introduced by the attenuator.

1.6.3 Elements of symmetrical π attenuator

The configuration of symmetrical π attenuator is shown in Fig 1.17.

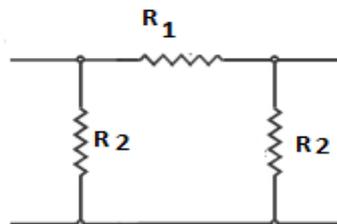


Fig.1.17 Elements of Symmetrical π attenuator

The value of series arm element of symmetrical π attenuator is

$$R_1 = R_0 \left[\frac{N^2-1}{2N} \right] \dots\dots\dots 1$$

Shunt arm element of Symmetrical π attenuator is

$$R_1 = R_0 \left[\frac{N+1}{N-1} \right] \quad \text{--- 2}$$

Equations (1) and (2) are the design equations of symmetrical π attenuator. Given the value of N and R_0 , series and shunt arm elements can be calculated using the above equations (1) and (2) N can be calculated from the attenuation in dB (D) which is to be introduced by the attenuator.

1.6.4 Applications of attenuator

1. Resistance attenuators are used to match impedances between two networks.
2. Resistance attenuators are used as volume control in Broadcasting stations.
3. Attenuators are also used in laboratory when it is necessary to obtain small value of voltage or current for test purposes.
4. Capacitive attenuators are used in high frequency applications.

1.6.5 Problems in Symmetrical Attenuators.

Problem 1.1

Design a π -type attenuator to give 20dB attenuation and to have a characteristic impedance of 600 ohms.

Given data

Attenuation, D = 20dB

Characteristic impedance, $R_0 = 600\Omega$

Solution:

Attenuation in Nepers, $N = \text{Antilog}_{10} \left[\frac{D}{20} \right]$

$$N = \text{Anti log}_{10} \left[\frac{20}{20} \right]$$

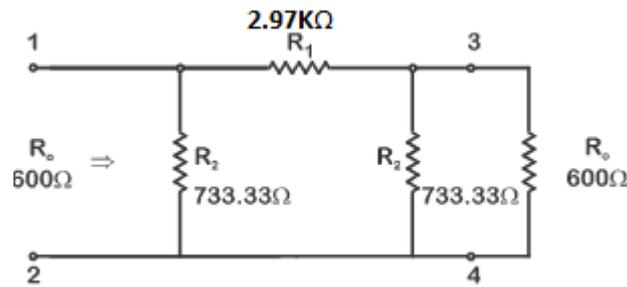
$$= \text{Anti log}_{10} \left[\frac{20}{20} \right]$$

$$N = 10$$

$$\begin{aligned} \text{Series arm impedance, } R_1 &= R_0 \left[\frac{N^2 - 1}{2N} \right] \\ &= 600 \left[\frac{10^2 - 1}{2 \times 10} \right] = 2970 \text{ ohms} \\ &= 2.97 \text{ k}\Omega \end{aligned}$$

$$\begin{aligned} \text{Shunt arm impedance, } R_2 &= R_0 \left[\frac{N + 1}{N - 1} \right] \\ &= 600 \left[\frac{10 + 1}{10 - 1} \right] \\ &= 733.33 \Omega \end{aligned}$$

The designed π attenuator is



1.7 FILTERS

- ❖ A Filter is a reactive network which passes the desired frequencies only.
- ❖ Filters attenuate all frequencies in the attenuation band called stop band.
- ❖ Filters allow all frequencies in the pass band.
- ❖ Frequency which separates the pass band and attenuation or stop band is called cut-off frequency.

1.7.1 Types of Filters

Filters can be classified as

1. Low Pass Filter (LPF)
2. High Pass Filter (HPF)
3. Band Pass Filter (BPF)
4. Band Elimination Filter (BEF)

1. Low Pass Filter (LPF)

LPF is one which passes all frequencies up to its designed cut-off frequency and attenuates all other frequencies greater than the cut-off frequency.

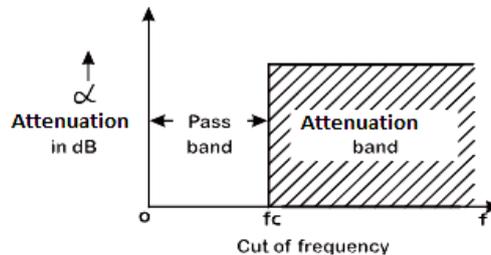


Fig.1.19 Lowpas filter

In Fig 1.19, LPF passes all frequencies below f_c . This band of frequencies is called pass band. Here attenuation α is zero.

While it attenuates all frequencies above f_c this band is called attenuation band. Here attenuation α is maximum.

2. High Pass Filter (HPF)

HPF is one which passes all frequencies above its designed cut-off frequency and attenuates all other frequencies below the cut-off frequency.

Fig 1.20 shows that HPF passes all frequencies above f_c and attenuates all frequencies below f_c .

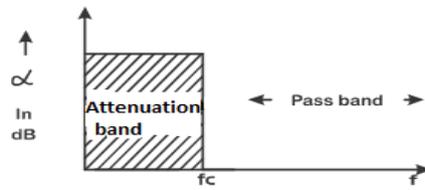


Fig.1.20 High pass filter

3. Band Pass Filter (BPF)

BPF is one which passes all frequencies between two designated cut-off frequencies f_1 (lower cut-off frequency) and f_2 (upper cut-off frequency).

Fig 1.21 shows that BPF has two attenuation bands one below f_1 and one above f_2 . Here pass band lies in between f_1 and f_2 .

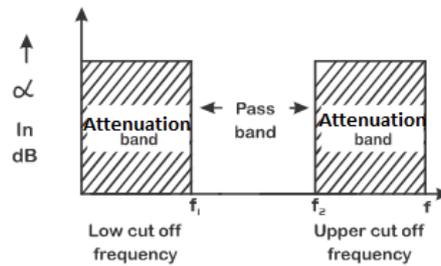


Fig.1.21 Band pass filter

4. Band Elimination Filter (BEF)

As the name implies it eliminates i.e) attenuates all frequencies between the two designated cut-off frequencies f_1 and f_2 , while transmits all frequencies below f_1 and above f_2 .

Fig 1.22 shows the frequency response of BEF.

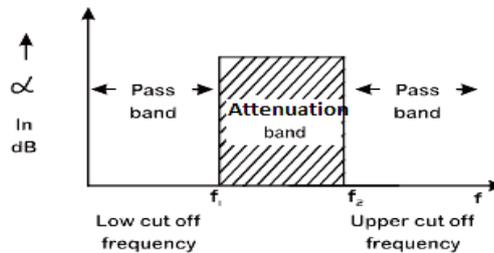


Fig.1.22 Band elimination filter

1.7.2 Elements and cut-off frequency of Low Pass Filter (LPF)

Filter (LPF)

The prototype Low Pass Filter is shown in FIG.1.23 (a) and Fig.1.23 (b). Fig 1.23(a) shows Symmetrical T-LPF and (b) shows a symmetrical π LPF.

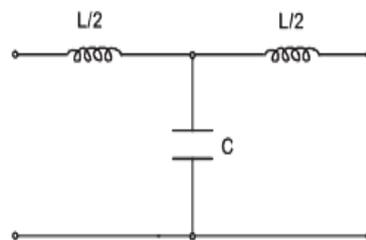


Fig.1.23(a) Symmetrical T-LPF

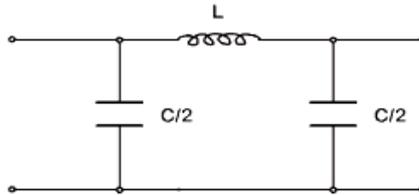


Fig.1.23(b) SYMMETRICAL π LPF

- ❖ Both T and π filter are constant K LPFs in which the series and shunt arm impedances Z_1 and Z_2 are connected by the relationship,

$$Z_1 Z_2 = R_K^2$$

Where, R_K is a real constant that is resistance independent of frequency. R_K is termed as nominal impedance of the constant $-K$ filter.

- ❖ Both T and π filters of fig 1.23 have the same total series and shunt impedances, they will have the same design impedance and cut-off frequency.

Cut-off frequency of LPF is

$$f_c = \frac{1}{\pi\sqrt{LC}}$$

Capacitance of LPF is

$$C = \frac{1}{\pi R_K f_c} \quad \dots\dots(1)$$

Inductance of LPF is

$$L = \frac{R_K}{\pi f_c} \quad \dots\dots(2)$$

Equations (1) and (2) are called design equations because they help in the design of filters if f_c and R_K are given.

1.7.3 Elements of High Pass Filter

The prototype high pass filter is shown in Fig 1-24

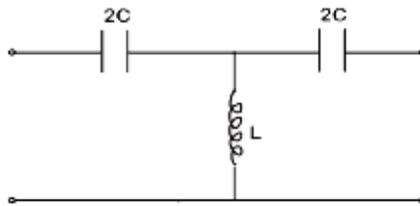


FIG.1.24 Symmetrical T-HPF

Since the high pass filter shown in 1.26 is a constant K filter, we can have,

$$Z_1 Z_2 = R_K^2$$

Cut-off frequency of HPF is

$$f_c = \frac{1}{\pi\sqrt{LC}}$$

Capacitance of HPF is

$$C = \frac{1}{4\pi R_K f_c} \quad \dots\dots(1)$$

Inductance of HPF is

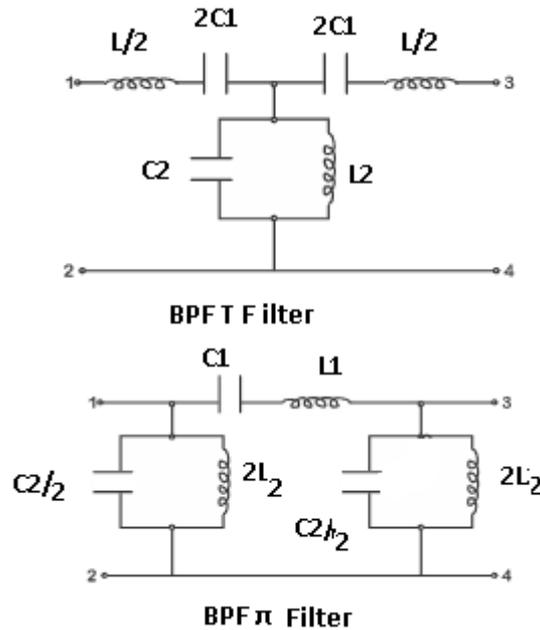
$$L = \frac{R_K}{4\pi f_c} \quad \dots\dots(2)$$

Equations (1) and (2) are called design equations because they help in the design of filters if f_c and R_K are given.

1.7.4 Elements of Band Pass filter (BPF)

A band pass filter may be obtained by using a low pass filter followed by a high pass filter.

The T-type and π -type band pass filters are shown in Fig. below



Its elements are,

$$L_1 = \frac{R_K}{\pi(f_2 - f_1)}$$

$$L_2 = \frac{R_K(f_2 - f_1)}{4\pi f_1 f_2}$$

$$C_1 = \frac{f_2 - f_1}{4\pi R_K f_1 f_2}$$

$$C_2 = \frac{1}{\pi R_K(f_2 - f_1)}$$

$$f_0^2 = f_1 f_2$$

Resonant frequency, $f_0 = \sqrt{f_1 f_2}$

1.7.5 APPLICATION OF FILTERS

1. Used in DC power supplies to remove ac ripples
2. Used in Audio players to separate voice channels.
3. Used in communication receivers to split various frequency bands.
4. Used in instrumentation and telemetry to attenuate or allow the wanted frequency bands.

1.7.6 Problems in Filters

Problem 1.2

1. Design a constant K low pass filter having $f_c = 2000$ c/s and load 5000 ohms.

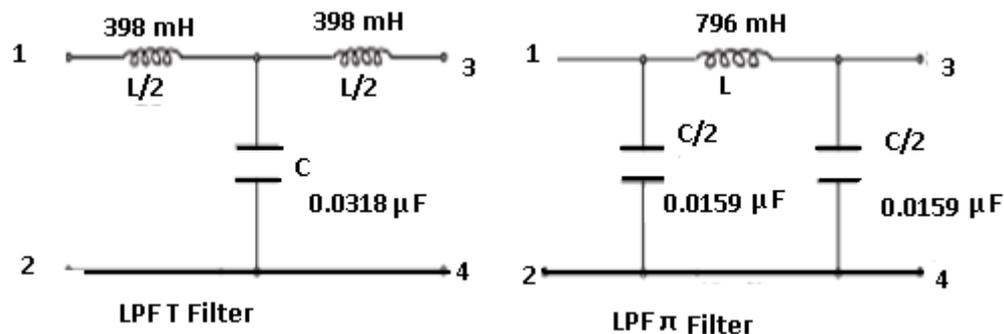
Given data:

Cut-off frequency, $f_c = 2000$ c/s = 2×10^3 Hz
 $R_k = 5000 \Omega = 5 \times 10^3 \Omega$

Solution:

$$\begin{aligned} \text{Capacitance, } C &= \frac{1}{\pi R_k f_c} \\ &= \frac{1}{\pi \times 5000 \times 2000} = \frac{1}{31.4} \times 10^{-6} \\ &= 0.03184 \mu\text{F} \\ \text{Inductance, } L &= \frac{R_k}{\pi f_c} \\ &= \frac{5 \times 10^3}{3.14 \times 2 \times 10^3} \\ &= 0.796 \times 10^3 \times 10^{-3} \\ &= 796 \text{ mH} \end{aligned}$$

The designed constant K low pass filter is shown below



Problem 1.3

A prototype high pass filter has a cut-off frequency 6 KHZ and nominal impedance 300 ohms. Calculate the values of elements L and C .

Given data:

Cut-off frequency, $f_c = 6 \text{ KHZ}$
 $= 6 \times 10^3 \text{ HZ}$
 $R_k = 300 \Omega$

Solution:

The value of inductance, $L = \frac{R_k}{4\pi f_c}$

$$= \frac{300}{4 \times 3.14 \times 6 \times 10^3}$$

$$= 3.98 \times 10^{-3} \text{ H}$$

$$= 3.98 \text{ mH}$$

The value of Capacitor, $C = \frac{1}{4\pi R_k f_c}$

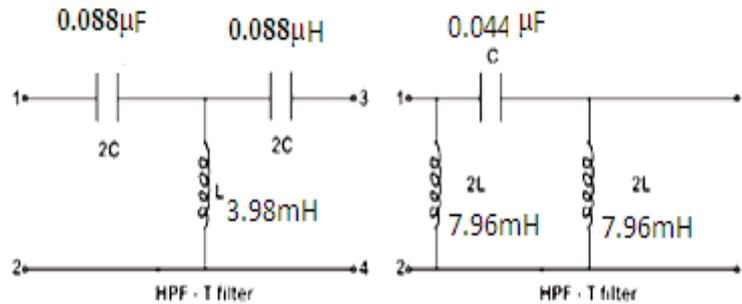
$$= \frac{1}{4 \times 3.14 \times 300 \times 6 \times 10^3}$$

$$= 4.423 \times 10^{-5} \times 10^{-3}$$

$$= 0.044 \times 10^{-6} \text{ F}$$

$$= 0.044 \mu\text{F}$$

The designed HPF shown in Figure below.



Problem 1.4

Design a band pass filter to work an input and output resistance of 100 ohms and have a pass band from 4800 to 5200 Hz.

Given data:

$$R_k = 100 \Omega$$

$$f_1 = 4800 \text{ Hz}$$

$$f_2 = 5200 \text{ Hz}$$

Solution

$$C_1 = \frac{f_2 - f_1}{4\pi R_k f_1 f_2}$$

$$= \frac{5200 - 4800}{4 \times \pi \times 100 \times 5200 \times 4800}$$

$$= 0.0128 \mu\text{F}$$

$$L_1 = \frac{R_k}{\pi(f_2 - f_1)}$$

$$= \frac{100}{\pi(5200 - 4800)}$$

$$= 79.58 \text{ mH}$$

$$L_2 = \frac{R_k(f_2 - f_1)}{4\pi f_1 f_2}$$

$$= \frac{100(5200 - 4800)}{4 \times \pi \times 4800 \times 5200}$$

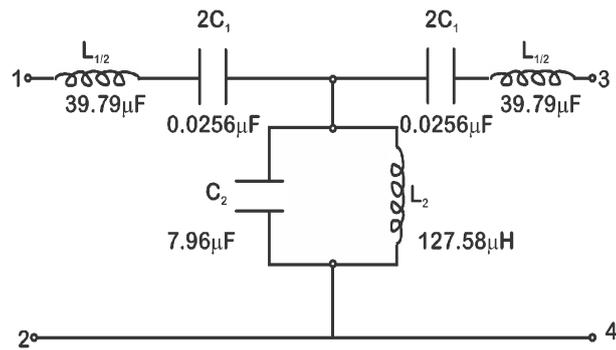
$$= 127.53 \mu\text{H}$$

$$C_2 = \frac{1}{\pi R_k (f_2 - f_1)}$$

$$= \frac{1}{\pi \times 100 (5200 - 4800)}$$

$$= 7.96 \mu\text{F}$$

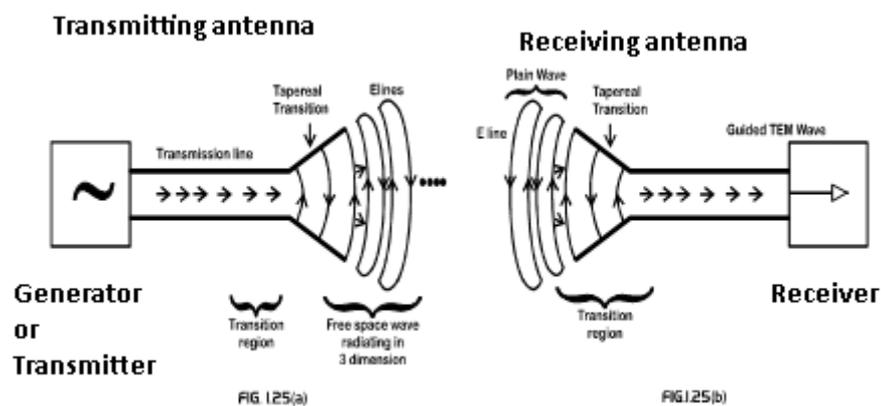
The designed band pass filter is



1.14 ANTENNAS

Basic Antenna Principle

- ❖ Antenna may be defined as the structure associated with the region of transition between a guided wave and a free-space wave or vice-versa.
- ❖ Regardless of antenna type, all involve the same principle that radiation is produced by accelerated (or decelerated) charge.
- ❖ Antennas convert electrons into photons or vice-versa as shown in Fig.1.25.



- ❖ Transmitting antenna in Fig.1.25 (a) is a region of transition from a guided wave on a transmission line to a free space wave.
- ❖ The receiving antenna in Fig.1.25 (b) is a region of transition from a space wave to a guided wave on a transmission line.

Thus,

'An antenna is a transition device, a transducer, between a guided wave and a free-space wave, or vice-versa.'

Functions of Antenna

The antenna has dual function.

1. It accepts energy from transmission line and radiating it into space in desired directions.
2. It receives electromagnetic energy from space and converts it into electrical energy so as to pass the transmission line.

In general

'Antenna is defined as a transducer which converts electrical energy into electromagnetic energy or vice – versa'.

1.14.1 Types of Antennas

Antennas can be classified in several ways.

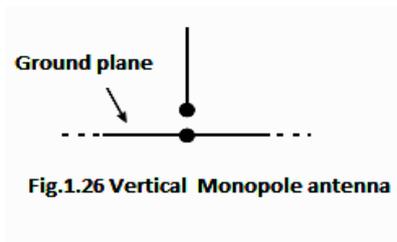
- ❖ According to the frequency band of operation.
- ❖ According to the radiating direction
- ❖ According to the physical structure
- ❖ According to the electrical/electromagnetic design.

Examples of widely used Antennas types are,

- ❖ Dipole antenna
- ❖ Monopole antenna
- ❖ Helical antenna
- ❖ Yagi-Uda antenna
- ❖ Parabolic reflector antenna
- ❖ Horn antenna

1.14.2 Monopole antenna

A monopole antenna is a class of radio antenna consisting of a straight rod-shaped conductor, often mounted perpendicularly over some type of conductive surface, called a ground plane.



- ❖ Fig.1.26 shows one type of monopole antenna. In this the rod functions as an open resonator for radio waves, oscillating with standing waves of voltage and current along its length.
- ❖ Therefore, the length of the antenna is determined by the wavelength of the radio waves it is used with.
- ❖ The most common or fundamental form is **the quarter-wave monopole**, in which the antenna is approximately $\frac{1}{4}$ of the wavelength of the radio waves.
- ❖ The monopole antenna was invented in 1895 by radio pioneer Guglielmo Marconi; for this reason it is sometimes called the *Marconi antenna*. Common types of monopole antenna are the helical, random wire.
- ❖ The driving signal from the transmitter is applied, or for receiving antennas the output signal to the receiver is taken, between the lower end of the monopole and the ground plane.
- ❖ One side of the antenna feed line is attached to the lower end of the monopole, and the other side is attached to the ground plane, which is often the Earth.

- ❖ This contrasts with a dipole antenna which consists of two identical rod conductors, with the signal from the transmitter applied between the two halves of the antenna.

1.14.3 Dipole antenna

Dipole antenna consists of two identical rod conductors, with the signal from the transmitter applied between the two halves of the antenna.

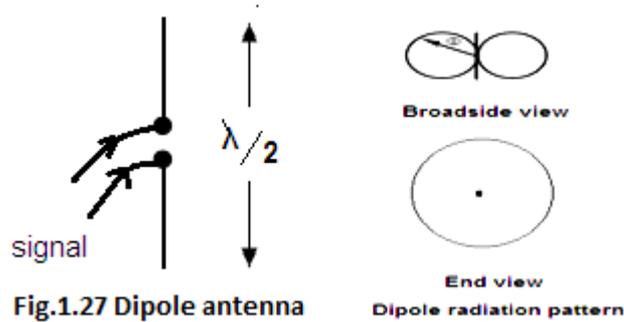


Fig.1.27 Dipole antenna

Dipole radiation pattern

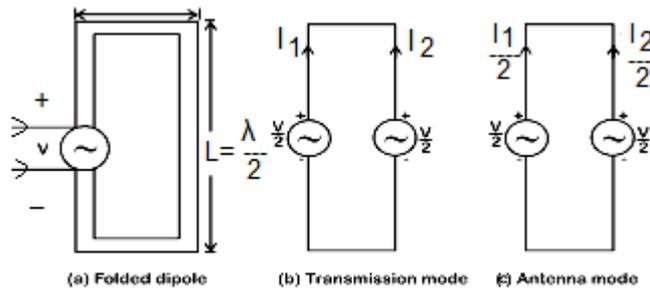


Fig.1.28 Folded Dipole

Folded dipole

- ❖ To achieve good directional pattern characteristics and at the same time provide good matching to practical coaxial lines with 50 or 75 ohms characteristic impedances, the length of a single wire element is usually chosen to be $L = \lambda/4$.
- ❖ The most widely used dipole is that whose overall wave length is $\lambda/2$.
- ❖ In order to provide good matching characteristics, variations of the single dipole element must be used.
- ❖ One simple geometry that can achieve this is a folded wire which forms a very thin (Spacing $S \ll \lambda$) rectangular loop as shown in Fig.1.28.
- ❖ This antenna, when the spacing between the two larger sides is very small usually $S < 0.05\lambda$, is known as a folded dipole and it serves as a step-up impedance transformer of the single element impedance.
- ❖ Thus when $L = \lambda/2$ the antenna is resonant, impedance on the order of 300Ω can be achieved.

A folded dipole operates basically as an imbalanced transmission line, and it can be analyzed by assuming that its current is decomposed into two distinct modes;

- (1) Transmission line mode
- (2) Antenna mode as shown in Fig.1.28

Application

- ❖ A two element folded dipole is widely used as feed element of TV antennas such as Yagi-Uda elements.

Advantages

- ❖ Although the impedance of an isolated folded dipole may be around 300 ohms, its value will be somewhat different when it is used as an element in an array or with a reflector.
- ❖ The folded dipole has better bandwidth characteristics than a single dipole of the same size.

1.14.4 Directional antenna

Antennas that offer enhanced performance in some directions, at the expense of other directions, are called [directional antennas](#) or [directional radiators](#).

- ❖ Directional antennas consist of arrays of elements, such as dipoles, or use one active and several passive elements, as in the Yagi-Uda antenna.



Fig.1.28 Directional antenna

- ❖ The most common example is the [dish antenna](#) used with [satellite Internet](#), satellite television, and space-communications installations.
- ❖ Other examples include quad antenna, billboard antenna, and [helical antenna](#).

1.14.5 Omni directional antenna

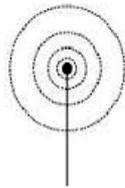


Fig.1.29 Omnidirectional antenna

An Omni directional antenna is a [wireless](#) transmitting or receiving [antenna](#) that radiates or intercepts radio-frequency (RF) [electromagnetic fields](#) equally well in all horizontal directions in a flat, two-dimensional (2D) geometric plane.

- ❖ Omni directional antennas are used in most consumer RF wireless devices, including [cellular telephone](#) sets and [wireless routers](#).

1.14.6 Antenna Arrays

An antenna array is a system of similar antennas, similarly oriented to achieve great directivity.

(or)

'System of Similar antennas kept in some distance in the same axis is called an antenna array'.

- ❖ Antenna Arrays are used to improve directivity.
- ❖ Since more, similar antennas are placed in the same axis they interact with one another to produce a resulting radiation pattern that is the vector sum of the individual ones.

- ❖ The reinforcement or cancellation takes place in any given direction is determined by the spacing between elements and the phase difference between various feed points.

Types of Antenna Arrays

Basically there are two types of antenna arrays.

- (i) Driven arrays (ii) Parasitic arrays.

Driven Arrays

There are three types of Driven Arrays.

1. Broad side array
2. End fire array
3. Turnstile antenna

All are linear array of antennas consists of a number of basic antenna elements, usually $\lambda/2$ dipole, equi - spaced out along a line referred to as the array axis.

By suitably phasing the radiation from each element, the directivity can be altered.

1.14.6.1 The Broadside array

The broad side array is the simplest array consists of a number of dipoles of equal size, equally spaced along a straight line, with all dipoles fed in the same phase from the same source.

- ❖ It is strongly directional at right angles to the plane of the array. The arrangement and radiation pattern are shown in Fig. 1.30

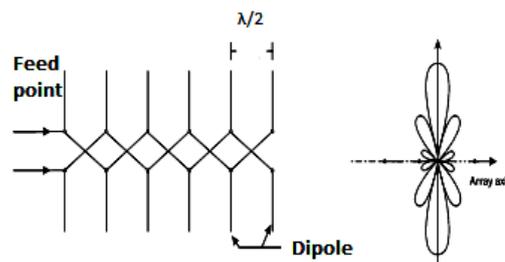


Fig.1.30 Broadside array Radiation pattern of Broadside array

- ❖ This array radiates equally well in all directions within the plane normal to the axis, but produces very little radiation off this plane and none along the array axis.
- ❖ Typical antenna lengths in the broadside array are from 2 to 10 wave lengths, typical spacing are $\lambda/2$ or λ , and dozens of elements may be used in one array.

1.14.4.2 The End-Fire Array

The end-fire array, as the name suggests has the main beam directed along the axis of the array (as shown in fig 1.31).

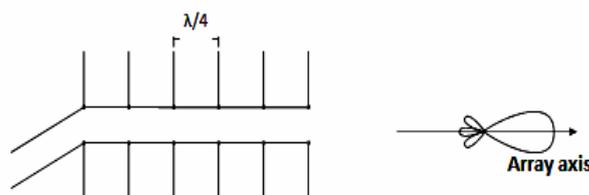


Fig.1.31 Endfire array and its radiation pattern

A common arrangement used in practice is to space the elements by $\lambda/4$ and directly feed them in parallel.

- ❖ Depending on the number of elements some minor back lobes may exist, but these are much smaller than the forward lobe.
- ❖ There is no radiation at right angles to the plane of the array because of cancellation.

1.14.4.3 Applications of Array of Antennas

Both the board-side and end-fire arrays may be used for **point to point communication**.

1.14.4.3 Parasitic Arrays

A parasitic array antenna consists of minimum three elements.

(i) Simple folded dipole antenna

(ii) Parasitic reflectors

(iii) Parasitic directors

- ❖ Parasitic elements are secondary antennas which are placed in close proximity to the main or driven antenna. (Dipole).
- ❖ They are not directly fed, but have currents induced in them from the main element.
- ❖ The reflector element is placed behind the driven dipole and is made about 5% longer than the driven dipole ($\lambda/2$).
- ❖ Optimum spacing is 0.15λ
- ❖ Parasitic directors are placed in front of the driven dipole.
- ❖ It is made about 5% shorter than the dipole so that it is capacitive.
- ❖ It is spaced 0.15λ to provide maximum radiation in the forward direction.

Yagi-Uda Array

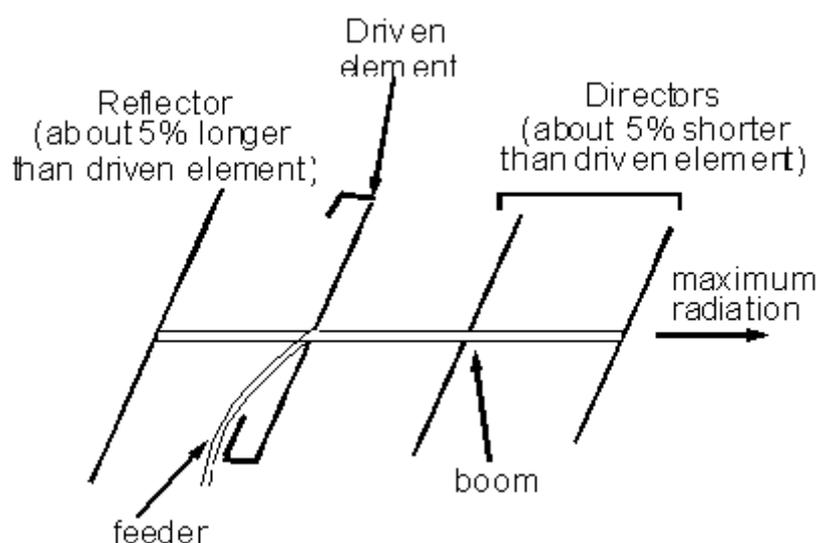


Fig 1.32 shows a Yagi-Uda antenna.

It is a parasitic array comprising a driven half-wave dipole antenna which is usually a folded dipole, a single parasitic reflector, and one or more director elements.

- ❖ Director 1 acts as driven element for director 2 and so on.
- ❖ Only one reflector need be used.
- ❖ The directive gain is improved considerably by the addition of more directors.
- ❖ For a five elements Yagi-Uda antenna the directive gain is about 15db.
- ❖ A Yagi-Uda antenna with a folded dipole would have a terminating impedance of 200 to 300Ω

Applications of Yagi-Uda Antenna

Yagi-Uda antenna is used in the VHF range, particularly as a VHF television receiving antenna.

Advantages of Yagi-Uda Antenna

1. It is very compact.
2. It has good unidirectional radiation pattern.
3. It is employed at higher frequencies,

1.14.4.4 Parabolic reflector antenna

The most widely used antenna for microwaves is the paraboloidal reflector antenna.

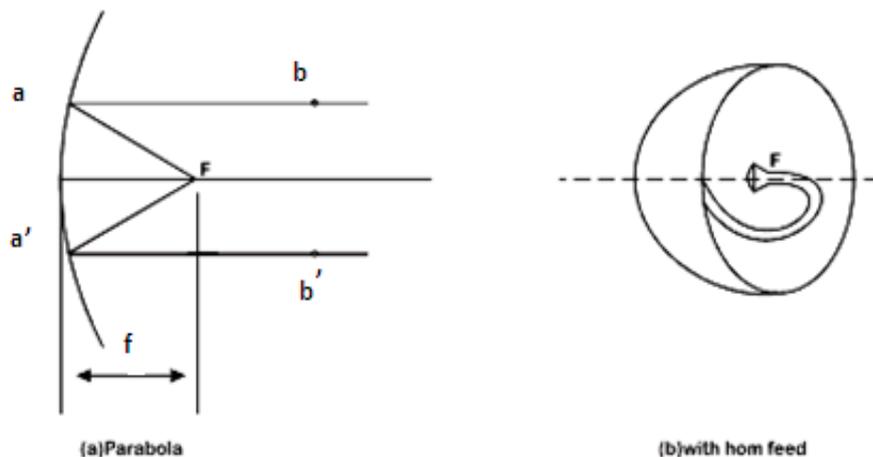


Fig.1.33 Parabolic reflector antenna

It consists of a primary antenna such a horn, situated at the focal point of a paraboloidal reflector as shown in fig 1.33.

- ❖ The mouth of the reflector is circular.
- ❖ The parabolic reflector employs properties of the parabola.
- ❖ The path length $Fa b = Fa'b'$ where bb' is perpendicular to the reflector axis.
- ❖ Thus the reflector can focus parallel rays onto the focal point 'F' and conversely, it can produce a parallel beam from radiation emanating from the focal point.
- ❖ The directivity of the paraboloidal reflector is a function of the primary antenna directivity and the ratio of focal length 'f' to reflector diameter 'd' (f/d). This ratio is known as aperture number.

1.14.5 ANTENNA PARAMETERS

1.14.5.1 Directive Gain

'The directive gain in a particular direction is defined as the ratio of the radiation intensity (or power density) in that direction to the average radiated power'.

It is denoted by G_d .

$$G_d = \frac{P_r}{P_i}$$

Where, P_r is radiated power

P_i is incident (input) power.

1.14.5.2 Directivity

The directivity of an antenna is defined as the maximum directive gain being specified for a particular direction.

1.14.5.3 Radiation Pattern

- ❖ *The radiation pattern of an antenna is a graphical representation of the radiation of the antenna as a function of direction.*
- ❖ When the radiation is expressed as field strength, E volts per meter, the radiation pattern is a field strength pattern.
- ❖ If the radiation in a given direction is expressed in terms of power per unit solid angle, the resulting pattern is a power pattern.
- ❖ Unless specified, the radiation patterns normally referred to, will be field-strength patterns.

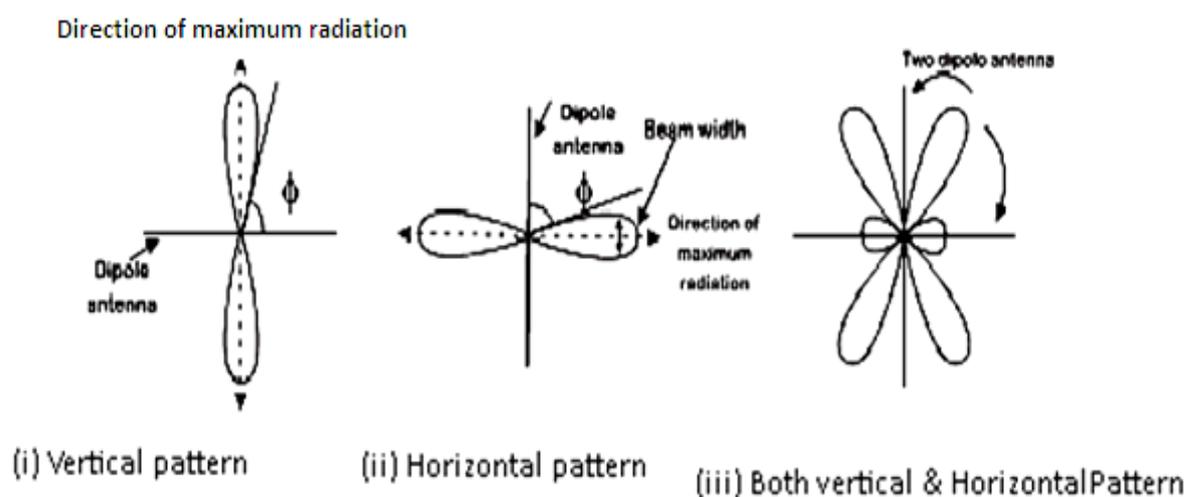


Fig.1.34 Radiation pattern

- ❖ A complete radiation pattern gives the radiation for angles θ and ϕ and requires three dimensional presentations.
- ❖ Cross sections in which the radiation patterns are most frequently given are the horizontal and vertical.

- ❖ These respectively called the horizontal patterns and vertical patterns which are shown in Fig 1.34.

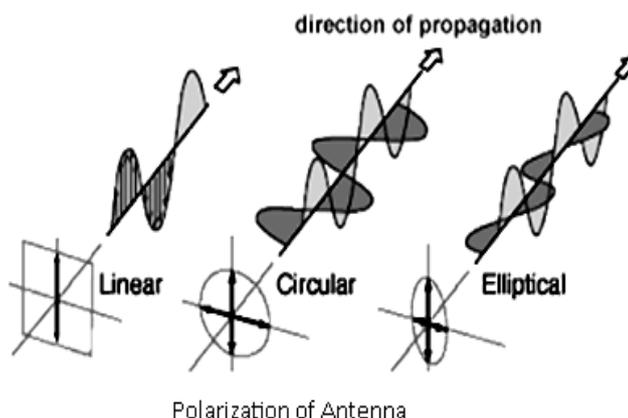
The radiation pattern in free space mainly depends on its length (length of the antenna)

From the radiation pattern of an elementary dipole shown in Fig 1.34, we see that very little directivity is achieved. Maximum radiation takes place at right angles to the axis.

1.14.5.4 Antenna Polarization

Polarization

- ❖ Polarization of an antenna in a given direction is defined as ‘The polarization of the wave transmitted (radiated) by the antenna’.
- ❖ At a given position, the polarization describes the orientation of the electric field.
- ❖ Polarization of radiated wave is defined as that property of an electromagnetic wave describing the time varying direction and relative magnitude of the electric-field vector.’
- ❖ Polarization may be classified as linear, circular or elliptical.
- ❖ If the vector that describes the electric field at a point in space as a function of time is always directed along a line, the field is said to be linearly polarized which is shown in figure below.



- ❖ Vertical polarization: electric force lines lie in a vertical direction
- ❖ Horizontal polarization: electric force lines lie in a horizontal direction.
- ❖ Circular polarization: electric force lines rotate 360° every cycle

1.14.6 Propagation

When electric power (high frequency) is applied to an antenna system it emits electromagnetic waves.

Electromagnetic waves are oscillations that travel through free space with the velocity of light (3×10^8 Meters /sec.)

This is called propagation of radio waves.

- ❖ The direction of the electric field, the magnetic field and propagation are mutually perpendicular in electromagnetic wave.
- ❖ In the earth environment the electromagnetic waves spread in all the directions equally irrespective of their own properties if it is a free space.
- ❖ The actual environment may have obstacles, and propagation medium variation.
- ❖ The radiation and propagation of radio waves can not be seen by our naked eye.

1.14.6.1 Electromagnetic waves in free space

Free space is space that does not interfere with the normal radiation and propagation of radio waves. Thus it has no magnetic or gravitational fields, no solid bodies and no ionized particles. It does not exist near the earth.

Since no interference or obstacles are present in free space, electromagnetic waves will spread uniformly in all directions from a point source. The wave-front is thus spherical as shown in cross-section in fig 1.35.

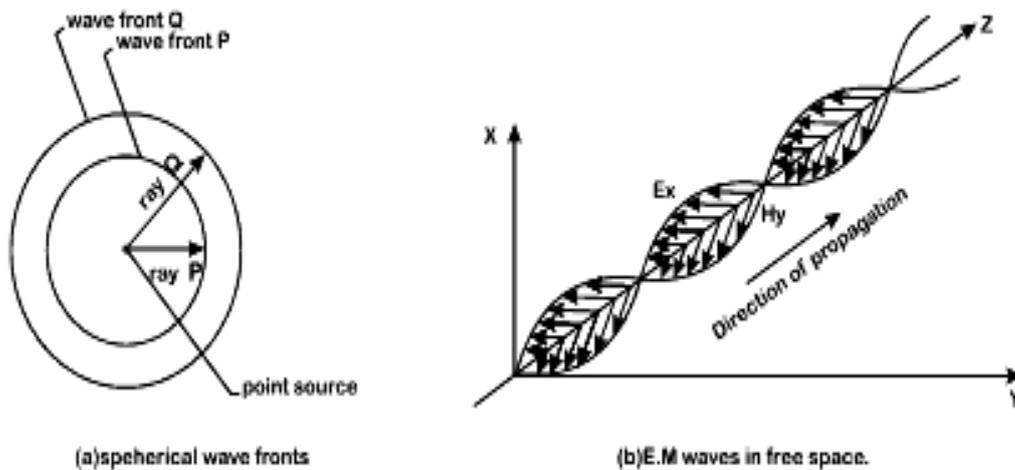


Fig.1.35

1.14.5.2 Various modes of propagation

Fig 1.33 illustrates various modes of radio wave propagation.

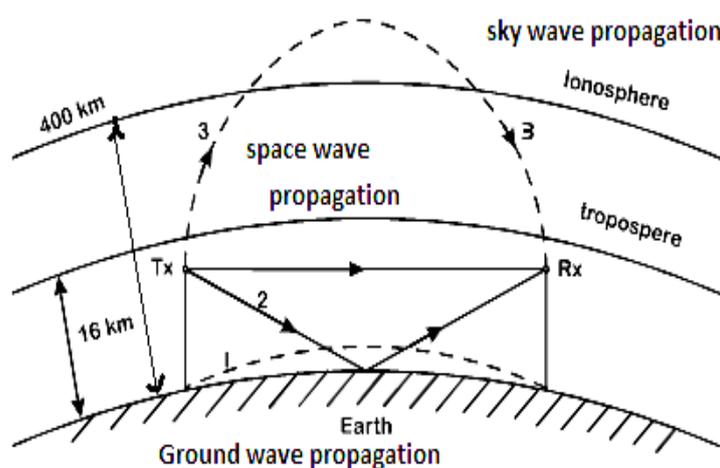


Fig.1.36 Various modes of radiowave propagation

Tx - Transmitting antenna.

Rx - Receiving antenna.

Path 1. Ground wave propagation or surface wave propagation.

Path 2. Space wave propagation or Tropospheric propagation.

Path 3. Sky wave propagation or Ionospheric propagation

1.14.5.3 Ground Wave Propagation

A ground wave is an electromagnetic wave that is guided along the surface of the earth. It will follow the curvature of the earth due to diffraction. This mode of propagation is also called as surface wave propagation.

- ❖ **The wave suffers varying amounts of attenuation according to its nature surface irregularities, permittivity and conductivity.**
- ❖ The energy lost in the surface comes from the radio wave which is therefore attenuated as it passes over the surface.
- ❖ **The attenuation increases with increase in frequency. Therefore this mode of propagation is suitable for medium waves, i.e. up to 2MHz only.**
- ❖ The ground wave propagation is often referred as medium wave propagation and can be used for local broadcasting only.

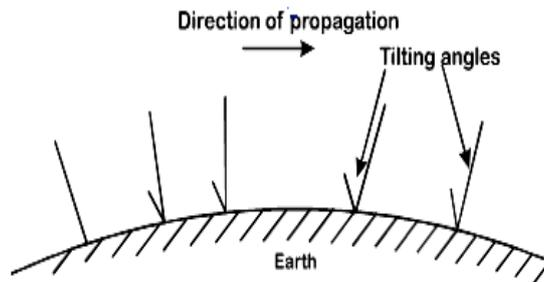


Fig. 1.37 Ground wave propagation

- ❖ The surface wave is always vertically polarized, as the conductivity of the ground would effectively short-circuit any horizontal electric field component.
- ❖ The surface wave induces currents in the ground over which it passes and thus loses some energy by absorption. Because of diffraction the wave front gradually tilts over as shown in fig 1.37.
- ❖ The tilting angle goes on increasing and it causes greater short circuiting of electric field components of the wave and hence field strength reduction happens. At some distance from the antenna the wave lies down and dies.

Application of ground wave Propagation

Used in Medium wave (MW) radio broadcast.

Limitations

1. The frequency usage is only 2MHz.
2. It cannot be used while frequency is above 2MHz.

1.14.5.4 Space Wave Propagation

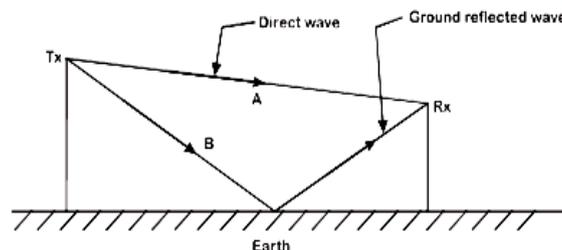


Fig.1.38 Space wave propagation

- ❖ In space wave propagation, the radio waves travel either straight through space or being reflected at ground shown by paths (A) and (B) respectively in fig 1.38.
- ❖ In this simplified picture the earth is assumed to be flat and the space wave reaching the receiver has two components, the direct wave (path A) and a ground reflected wave (path B).

- ❖ The troposphere is the region of the earth's atmosphere immediately adjacent to the earth's surface and extending upward for a height of about 15KM.

Since space wave propagates in troposphere, this mode of propagation is popularly known as Tropospheric Propagation.

Space wave propagation is also known as *Line of sight propagation*.

Applications of Space wave or Tropospheric Propagation.

This mode of propagation is suitable for radio waves with *frequency from 30 MHz onwards (30MHz to 300MHz)*.

Limitation:

1. It cannot be used below 30MHz.
2. It is suitable only for line of sight propagation.

1.14.5.5 Sky Wave Propagation (or Ionospheric Propagation)

In sky wave propagation radio wave is reflected back to earth from ionized region in the upper atmosphere called ionosphere as shown in fig. 1.36 (path (3)).

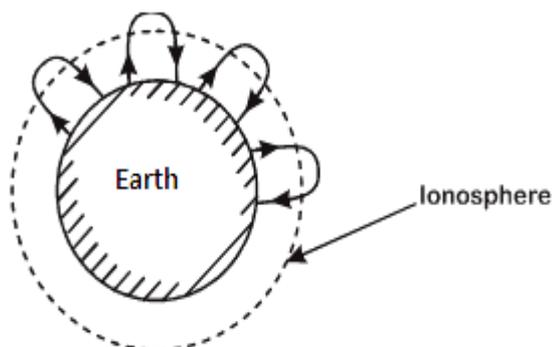


Fig.1.39 Multiple reflections from Ionosphere

However, electromagnetic waves of frequencies beyond 30MHz are not reflected back from the ionosphere instead they penetrate it. **Therefore sky wave propagation is suitable for short waves only (2MHz to 30MHz)**. Therefore this mode of propagation is also known as short wave propagation.

Sky wave propagation is useful for long distance communication round the globe through multiple reflections as shown in Fig 1.39 After a single reflection from the ionosphere, radio waves can cover distance not exceeding 4000 KM.

Since sky wave propagation is taken place by reflection from ionosphere it is often referred as Ionospheric propagation.

Applications of Sky wave (or) Ionospheric propagation.

Short wave point to point communication is achieved with sky wave propagation.

Limitation:

1. It is suitable for short wave only 2MHz to 30MHz.
2. Radio waves can cover distance not exceeding 4000 KM. (area of coverage is limited)
3. It is subjected to fading because of number of waves coming by different paths to the receiving point.

Structure of Ionosphere

- ❖ The upper part of atmosphere absorbs large quantities of radiant energy from the sun which produces some ionization in the form of free electrons and positive ions.
- ❖ This happens above 60KM from the earth's surface. The part of the upper atmosphere where the ionization is appreciable is called ionosphere.
- ❖ The ionization density varies with height as shown in fig 1.41.

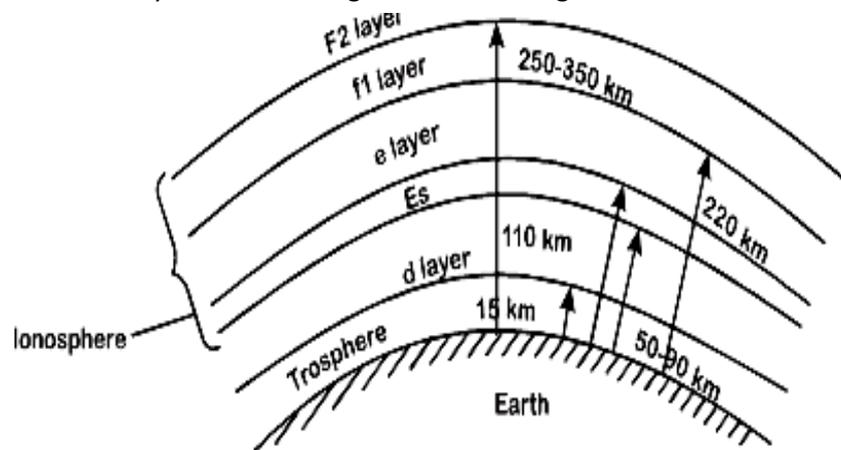


Fig.1.40 Various layers of Ionosphere

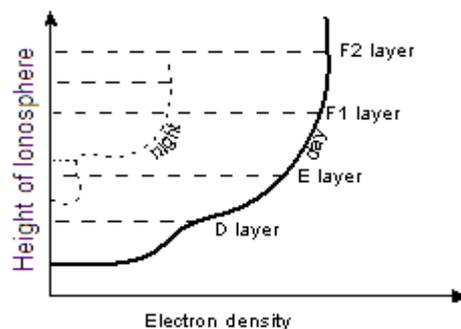


Fig.1.41 Variation of electron density in various layers of ionosphere

As shown in fig 1.40 atmosphere has the following layers, defined each by its varying electron density.

D-Layer

- ❖ This is the lowermost region of the ionosphere and is situated in the height range of 50 to 90KM.
- ❖ This layer is present only during day light hours and disappears at night. Its electron density is 10^4 to 10^6 per cubic centimetre, and reaches its maximum value at noon.
- ❖ Its critical frequency is about 100 KHz.

E-Layer

- ❖ E-layer is next in height existing at about 100KM, with a thickness of perhaps of 25KM.
- ❖ This layer is formed by ionization of all gases, by soft X-rays.
- ❖ Its electron density is maximum at noon and in summer increases with increased solar activity.
- ❖ Critical frequency of E-layer lies in the region of 3 to 5MHz at noon.

Es-Layer

- ❖ The sporadic E-layer is a thin layer sometimes making an appearance with the E-layer.
- ❖ Its electron density may be up to 10 times that of the E-layer.
- ❖ Its height above the earth is about 100KM.
- ❖ Its height differs from that of E-layer by not more than 5 to 10KM.
- ❖ Es-layer consists of relatively dense patch of electrons.

F₁-Layer

- ❖ It exists at a height of 180KM in day time and combines with the F₂-layer at night.
- ❖ Its day time thickness is about 25KM.
- ❖ Its critical frequency lies in the region of 5 to 7MHz.
- ❖ Most of the high frequency waves pass through it, some of them are reflected by it.
- ❖ Thus the main effect of the F₁ layer is to provide more absorption for HF waves.

F₂-Layer

- ❖ The F₂-layer is situated at a height of about 250 to 350KM having highest electron density of all the layers of ionosphere and varies from 3×10^5 to 2×10^6 .
- ❖ It is formed by ionization of UV, X-rays and other radiations.
- ❖ Its maximum critical frequency is 10MHz.
- ❖ F₂ layer is present throughout the day and night.
- ❖ The ionization in the F₂-layer is largely affected by earth's magnetic field.
- ❖ Being situated highest, F₂-layer is affected most by the atmosphere (tides and winds).
- ❖ It is also affected by the ionospheric storms.

Reflection mechanism of Sky waves

- ❖ Electromagnetic waves returned to earth by one of the layers of the ionosphere appear to have been reflected.
- ❖ In actual fact the mechanism involved is refraction.
- ❖ As the ionization density increases for a wave approaching the given layer at an angle.
- ❖ So the refractive index of the layer is reduced.
- ❖ Hence the incident wave is gradually bent farther and farther away from the normal.
- ❖ If the rate of change of refractive index per unit height is sufficient, the refracted ray will eventually become parallel to the layer.
- ❖ It will then be bent downward, finally emerging from the ionized layer at an angle equal to the angle of incidence.
- ❖ Some absorption has taken place, but the wave has been returned by the ionosphere.

Review Questions

Part – A

(Two Marks)

1. Define Symmetrical and Asymmetrical Network.

Asymmetrical Network.

A Symmetrical network is a four terminal network in which the electrical properties are unaffected by interchanging the input and output terminals.

Asymmetrical Network.

An asymmetrical network is a four terminal network in which the electrical properties are affected by interchanging the input and output terminals.

2. State the characteristics of Asymmetrical Network.

An Asymmetrical four terminal network has the following four electrical characteristics.

- 1) Image impedances Z_{i1} and Z_{i2}
- 2) Iterative impedances Z_{t1} and Z_{t2}
- 3) Image transfer constant
- 4) Iterative transfer constant

3. Define the characteristics of Symmetrical Network.

Characteristic Impedance

It is the impedance which when connected to one pair of terminals produces the same impedance at the other pair of terminals'

4. Give the expression for the characteristic impedance and Propagation constant of symmetrical network.

Characteristic impedance

$$Z_0 = \sqrt{Z_{oc} \cdot Z_{sc}}$$

Propagation constant

$$P = \alpha + j \beta$$

5. What is equalizer? What are the different types of equalizer?

Equalizers are electrical networks designed to counteract the attenuation (or) phase distortion occurring in any part of the circuit.

Equalizers are classify in to two types

1. Amplitude (or) Attenuation equalizer
2. Phase (or) Delay equalizer

6. What is attenuator?

An attenuator is a network which is used to reduce a known amount of the voltage, current or power between it's properly terminated input and output terminals.

7. What are the different types of attenuators?

Attenuators are of two types,

1. Symmetrical Attenuators
2. Asymmetrical Attenuators
3. Fixed (or) Variable attenuators
4. Resistive (or) Capacitive attenuators

8. What are the applications of attenuator?

1. It is used in waveguides

2. It is used for minimizing radiation
3. It is used to prevent the losses in co-axial cable
4. It is used in UHF and microwave applications

9. Define filter.

A filter can be defined as a network having at least one range of frequency in which the attenuation is zero and at least one other range of frequency in which the attenuation is infinite.

10. Mention the types of filter.

1. Low pass filter
2. High pass filter
3. Band pass filter
4. Band Eliminator filter

11. What are constant – k Filters?

Filters which satisfy the relation $Z_1 Z_2 = R^2$ are called as constant-K filters.

12. What is the value of Cut off frequency of constant-K LPF and constant – k HPF?

Cut off frequency of constant-K LPF is,

$$f_c = \frac{1}{\pi\sqrt{LC}}$$

Cut off frequency of constant-K HPF

$$f_c = \frac{1}{4\pi\sqrt{LC}}$$

13. What is the unit of Attenuation?

Unit of Attenuation is Nepers (or) Decibels

14. What is Neper?

Neper is another unit of Attenuation. (i.e)

$$N = \log_e \left| \frac{I_S}{I_R} \right| \quad (\text{or}) \quad N = \log_e \left| \frac{V_S}{V_R} \right|$$

15. Define Neper.

It is defined as the natural logarithm of the ratio of input voltage or current to the output voltage or current.

16. What is decibel?

Unit of power is called decibel. It is denoted by 'dB'

17. Define decibel.

It is defined as the natural logarithm of the ratio of the input power to the output power in attenuator and

It is defined as the natural logarithm of the ratio of the output power to the input power in amplifier.

18. State any two applications of Filters.

- Filters are commonly used in,
- (1) Power supplies

(2) Radio receivers

19. Define Band pass filter.

The band pass filter circuit passes signals of frequencies between two designated cut off frequencies F_{c1} and f_{c2} and attenuates all other frequencies.

20. Define Band elimination filter.

The band elimination filter circuit attenuates signals of frequencies between two designated cut off frequencies F_{c1} and f_{c2} and passes all other frequencies.

21. Define transmission line. What is the use of Transmission line?

Transmission lines are a means to convey electrical signals (or) power between two points separated appreciably in distance.

Transmission line is used to connect transmitter with Antenna

22. Define Antenna.

The antenna may be defined as a transducer which converts electrical energy in to an electromagnetic energy and vice versa.

23. Define Directivity.

It is defined as the gain in the direction of one of the major lobes of the radiation pattern. It is nothing but a max directive gain.

24. Define radiation pattern

The radiation pattern of an antenna is a graphical representation of the radiation of the antenna as a function of direction.

25. Define directive gain.

It is defined as the ratio of power density in a particular direction at a given point by the antenna to the power density that would be radiated by an isotropic antenna.

26. Define antenna array.

An array of antenna is a system of similar antenna orient similarly to get greater directivity in a desired direction. Individual antenna of an array system is called the elements of the array.

27. Mention the types of antenna array.

1. Broad side array
2. End fire array
3. Collinear array
4. Parasitic array

28. Mention the types of propagation.

1. Ground wave
2. Sky wave
3. Space wave.

29. Name the elements of Yagi- Uda.

1. Folded dipole antenna
2. Parasitic reflector
3. One or more director elements

30. Mention the frequencies used in different propagation.

Ground wave propagation = 2 MHz

Sky wave propagation = 2 to 30 MHz

Space wave propagation = above 30 MHz

Part – B

(Three Marks)

1. Compare the characteristics of Symmetrical and Asymmetrical Networks.
2. List out the various types of equalizer and their applications
3. List out the various types of Attenuators and their applications
4. List out the various types of Filters and their applications.
5. What do you mean by Array of antenna? List the various types of antenna arrays.
6. Give the expression for cut-off frequency of LPF, HPF and BPF.
7. Define the following terms related to Antennas.
(i) Directive Gain (ii) Directivity (iii) Radiation Pattern
8. Briefly explain about Polarization of Antenna.
9. State the limitations of various radio wave propagations.
10. State the applications of various radio wave propagations.

Part – C

(Ten Marks)

1. Design a band pass filter to work an input and output resistance of 100 ohms and have a pass band from 4800 to 5200Hz.
2. Design a band pass filter to work an input and output resistance of 100 ohms and have a pass band from 4800 to 5200Hz.
3. Design a constant K low pass filter having $f_c = 2000$ c/s and load 5000 ohms.
4. Explain about Antenna Arrays. State their applications.
5. Explain about (i) Monopole antenna (ii) Directional antenna (iii) Omni directional antenna
6. Explain about (i) Antenna Polarization (ii) Radiation pattern
7. Explain about various radio wave propagations.
8. Explain about the various layers of Ionosphere.
9. Explain about Dipole and Folded Dipole antenna. What is the advantage of Folded Dipole over Folded Dipole antenna



UNIT - II

INTRODUCTION TO MODULATION AND AMPLITUDE MODULATION

LEARNING OBJECTIVES

On completion of this unit the students must be able to....

- Know the Electromagnetic frequency spectrum
- Under the concepts of modulation
- Study the process of AM Modulation
- Understand the sidebands of AM and its frequency spectrum in time domain and frequency domain
- Learn the methods of transmission of AM wave

2.0. ELECTROMAGNETIC FREQUENCY SPECTRUM

- ❖ In an earth environment electromagnetic waves propagate in ways that depends not only on their own properties but also on those of the environment itself.
- ❖ The total electromagnetic frequency spectrum shown in fig.2.1 shows the approximate locations of various services.

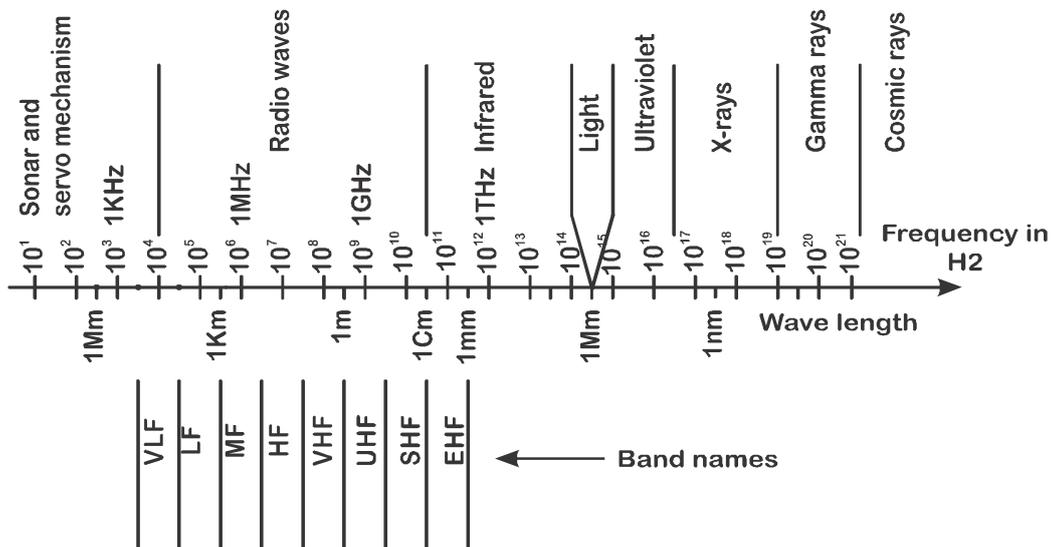


Fig. 2.1 Electro magnetic frequency system

- ❖ The electromagnetic frequency spectrum is divided into subsections, or bands, with each band having a different name and boundary.
- ❖ The International Telecommunications Union (ITU) is an international agency in control of allocating frequencies and services within the overall frequency spectrum.
- ❖ The total usable radio-frequency (RF) spectrum is divided into narrower frequency bands, which are given descriptive names and band numbers.

2.1 Relationship between Wavelength and Frequency

Frequency: Number of cycles completed in one second is called frequency. It is denoted by 'f'.

Units of frequency

1KHZ	=	1x10 ³ HZ
1MHZ	=	1x10 ⁶ HZ
1GHZ	=	1x10 ⁹ HZ
1THZ	=	1x10 ¹² HZ

Wavelength: The distance travelled by a wave in one cycle is called wavelength. It is denoted by 'λ'. It is expressed in mm or cm or meters.

$$1 \text{ mm} = 1 \times 10^{-3} \text{ meters}$$
$$1 \text{ cm} = 1 \times 10^{-2} \text{ meters}$$

Relationship between Frequency and Wavelength

Since electromagnetic waves travel at the speed of light in the free space, their wavelength is given by,

$$\text{Wave length, } \lambda = \frac{\text{Velocity of light}}{\text{frequency}}$$
$$= \frac{c}{f}$$

where 'c' is velocity of light in free space $c = 3 \times 10^8$ meters/sec.

$$\therefore \lambda = \frac{3 \times 10^8}{f} \text{ m/s}$$

Hence, wavelength decreases with increases in frequency.

2.2 MODULATION

Two necessary functions in successful transmission and reception of intelligence by the use of Radio waves are

(i) Modulation (ii) Demodulation.

Modulation is performed at the transmitting end and Demodulation is performed at the receiving end of a communication system.

Modulation

It is defined as the process by which some characteristics (amplitude, frequency or phase) of a high frequency carrier (short wave) is varied in accordance with the instantaneous voltage of a low frequency modulating signal (long wave).

2.2.1 Need for modulation

Transmission of audio frequencies to very long distance is impractical through transmission lines. If we choose space as the medium, the design of antenna is impractical for low frequency signals (Because antenna size will be very large at low frequencies). Hence if long distance transmission is required the only practical solution is to modulate a radio frequency wave with audio or video signals (By this antenna size can be reduced.)

Apart from these primary requirements of modulation of signals, there are additional objectives which are met by modulation. These are,

- (i) Ease of radiation

- (ii) Adjustment of Bandwidth
- (iii) S/N ratio can be improved by increasing bandwidth
- (iv) Shifting signal frequency to an assigned value
- (v) Number of transmission channels can be increased.
- (vi) Reduction in height of antenna

2.2.2 Types of modulation

Modulation may be grouped as shown in Fig 2.2

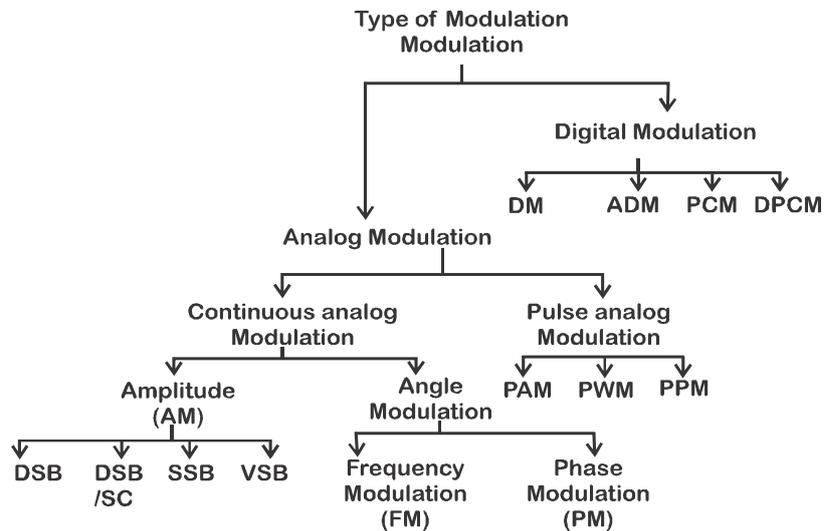


Fig. 2.2 Types of Modulation

2.3 AMPLITUDE MODULATION

Definition

Amplitude modulation is the process in which the amplitude of the carrier is varied in accordance with the instantaneous amplitude of the modulating voltage.

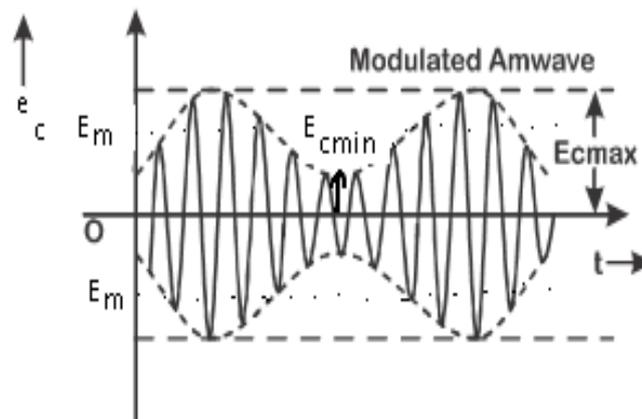
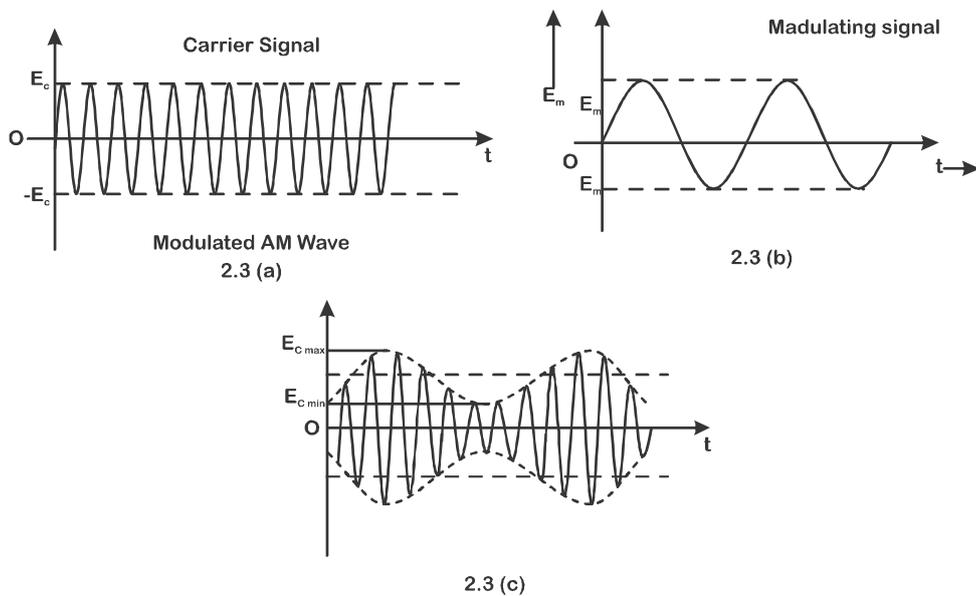


Fig.2.3 AM Wave in Time Domain

2.3.1 Wave form Representation of AM wave

Fig 2.3(a) shows the waveform of un modulated carrier voltage, Fig 2.3(b) that of sinusoidal modulation voltage and Fig 2.3(c) gives the waveform of Amplitude modulated carrier voltage.



It is seen from the Fig 2.3(c) that the carrier remains unchanged but its amplitude varies in accordance with the variation of the modulating voltage, e_m .

2.3.2 Expression for Modulation Index, m_a

It is seen further that,

$$m_a = \frac{k E_m}{E_c}$$

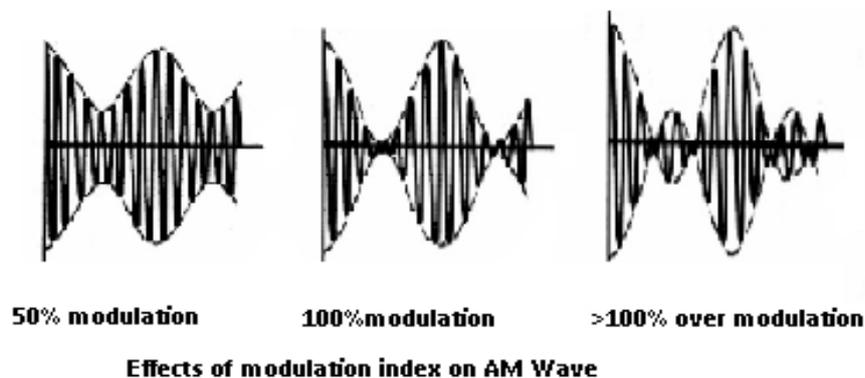
$$m_a = \frac{E_{c \max} - E_c}{E_c} \text{-----(1)}$$

$$m_a = \frac{E_c - E_{c \min}}{E_c} \text{-----(2)}$$

Solving 1 and 2, we can get

$$m_a = \frac{E_{c \max} - E_{c \min}}{E_{c \max} + E_{c \min}}$$

Where m_a is called the depth of modulation or modulation index Its value lies in between 0 and 1. The maximum value of m_a is unity. If $m_a > 1$ then over modulation will occur over modulation will distort the AM wave. It will lead to failure in reproduction of the original message.



2.3.3 AM sidebands

We know that the expression of AM Wave is,

$$e = E_c (1 + m_a \cos \omega_m t) \cos \omega_c t$$

From this expression we can find out the sidebands of AM wave.

$$e = E_c$$

$$\begin{aligned} & (1+m_a \cos \omega_m t) \cos \omega_c t \\ & = E_c \cos \omega_c t + m_a E_c \cos \omega_m t \cos \omega_c t \\ & = E_c \cos \omega_c t + \frac{m_a E_c}{2} 2 \cos \omega_m t \cos \omega_c t \\ & = E_c \cos \omega_c t + \frac{m_a E_c}{2} \cos(\omega_c + \omega_m)t + \frac{m_a E_c}{2} \cos(\omega_c - \omega_m)t \end{aligned}$$

In the above equation the frequency terms are,

(i) $E_c \cos \omega_c t$ is the original carrier voltage of angular frequency ω_c

(ii) $\frac{m_a E_c}{2} \cos(\omega_c + \omega_m)t$ is the upper side band (USB) term of angular frequency, $(\omega_c + \omega_m)$

(iii) $\frac{m_a E_c}{2} \cos(\omega_c - \omega_m)t$ is the lower side band (LSB) term of angular frequency $(\omega_c - \omega_m)$

2.3.4 Frequency spectrum of AM

Fig 2.4 shows the frequency spectrum of an AM DSBFC wave with reference to frequency f_m .

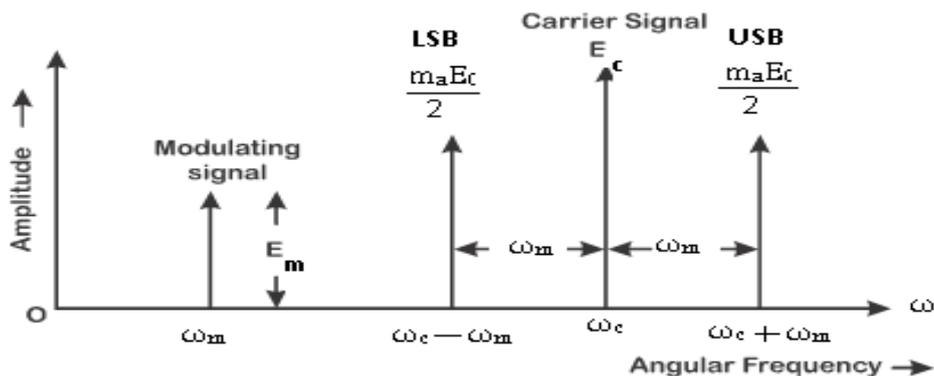


Fig. 2.4 Frequency Spectrum of AM

Fig 2.4 shows the frequency spectrum of AM with reference to frequency.

- ❖ Lower sideband and Upper side band terms are located in the frequency spectrum on either side of the carrier at a frequency interval of f_m .
- ❖ It is further noted that the magnitude of both sidebands is $m_a E_c / 2$ times the carrier amplitude E_c .
- ❖ If $m_a = 1$, then each sideband is half the carrier voltage in amplitude.
- ❖ It is seen from the above Fig 2.4 that AM signal shifts the frequency of the intelligence from audio level f_m to carrier level f_c .
- ❖ Intelligence appears also in both sidebands. The total bandwidth of this AM is $2f_m$.

2.4 Types of AM

Amplitude modulation may be classified as follows.

1. AM/DSB/FC
2. AM/VSB/FC
3. AM/DSB/SC

4. AM/SSB/SC

5. AM/VSB/SC

Where, DSB/FC is Double Side Band Full Carrier

VSB/FC is Vestigial Side Band Full Carrier

DSB/SC is Double Side Band Suppressed Carrier

SSB/SC is Single Side Band Suppressed Carrier

VSB/SC is Vestigial Side Band Full Suppressed Carrier

AM/DSB (or) AM/DSB/FC

In this system the modulated wave contains the carrier and both sidebands USB & LSB. The intelligence lies only in the sidebands not in the carrier. So the total transmitting power will be reduced. It is seldom used. Widely used in Radio broadcasting.

AM/VSB (or) AM/VSB/FC

In this system, the modulated wave contains the carrier, Upper Side Band and only trace of the Lower Side Band. It is only used in TV broadcasting.

AM/DSB/SC

In this system, the carrier power is suppressed only sidebands present. This will lead to a considerable power saving. But receiving end design is complicated. It is widely used for mobile communication and Radio telephone links.

AM/SSB/SC

In this system, only one sideband either, LSB or USB is present. Carrier and the other side band are suppressed. Power saving is

better than DSB/SC. It is widely used in wireless Mobile links & FM broadcast.

2.5 GENERATION OF SSB (AM/SSB/SC)

Single sideband modulation has become a very popular method of modulation particularly applied in mobile communication, because of its advantages shown below.

Advantages of SSB

1. Having ability to transmit good quality communication signals.
2. Having narrow bandwidth
3. Having low power for distances involved
4. Having power saving of 83.3% over DSB/FC AM
5. May be used as second independent channel.

However in practice it has some disadvantages

Disadvantages of SSB

1. It is difficult to detect or demodulate the original modulating signal from SSB signals.
2. It needs special synchronous detection process
3. It is difficult to design a filter with sharp cut-off on either side
4. Filter method of detection will introduce attenuation in the wanted side band also.
5. Increasing the bandwidth results in passing some of the un wanted side bands to the output.

There are two methods, commonly used for the generation of

SSB. i.e.,

1. Filter method or frequency discrimination method using frequency domain
2. Phase shift or phase discrimination method using time domain.

2.6 AM VSB SYSTEM (VESTIGIAL SIDEBAND MODULATION)

- ❖ Practical SSB system has poor low frequency response.
- ❖ On the other hand, DSB works quite well for low message frequencies but transmission bandwidth is twice that of SSB.
- ❖ Thus a compromise modulation scheme is desired which Vestigial Side Band modulation is called VSB system.
- ❖ VSB is derived by filtering DSB-AM in such a fashion that one sideband (USB) is passed almost completely while just a trace or vestige of the other side band (LSB) is included.
- ❖ The following fig 2.5(a) illustrates that how the VSB is generated.

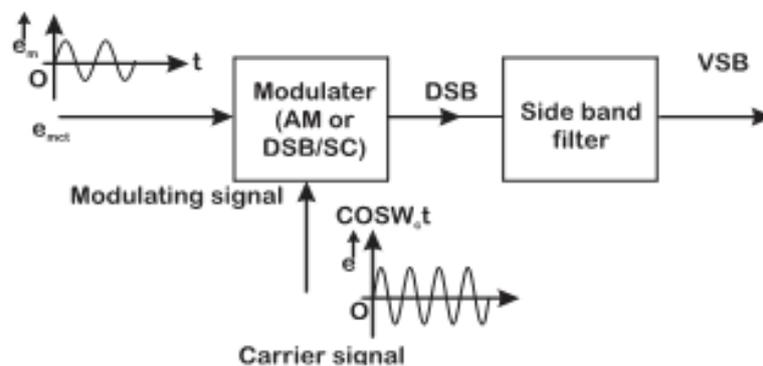


Fig.2.5 VSB Modulator

- ❖ In the above Fig 2.5(a) if the carrier suppression is wanted then the AM modulator is replaced by balanced modulator.
- ❖ The sideband filter passes one side band completely and the other sideband partially.
- ❖ This type of VSB having the advantage of bandwidth saving and power saving. Its bandwidth lies in between B and 2B of SSB and DSB and power saving is 75%. It is adopted mainly in TV broadcasting.

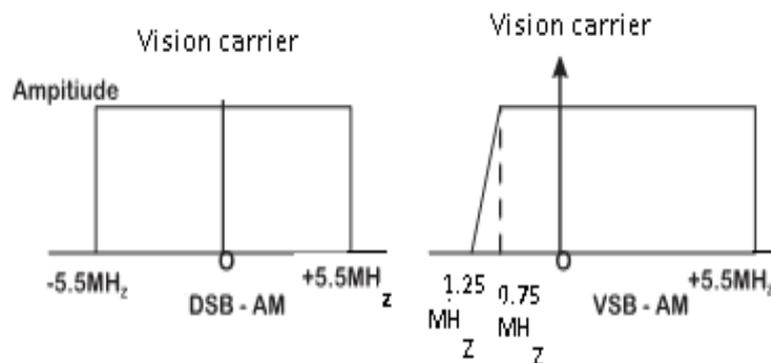


Fig.2.5 (b) VSB in Frequency Domain

- ❖ Fig 2.5(b) shows the VSB in frequency domain when it is used in TV transmission.
- ❖ When the image signal having frequency components up to 5 MHz is used to amplitude modulate the carrier, the resulting wave contains sideband frequency components extending 5MHz above and below the carrier frequency.

- ❖ It means bandwidth is 10MHz. In commercial practice, each transmitting station is assigned the use of a 7 MHz channel.
- ❖ Within this, channel provision must also be made for the conveyance of sound intelligence which accompanies the usual image transmission.

Advantages of VSB

1. Bandwidth is saved
2. 75% of the total power is saved
3. Generation of VSB is easy when carrier is present.
4. Detection is also easy

Applications

It is adopted mainly for TV transmission.

2.7 RADIO TRANSMITTERS

Radio Transmitters generates RF energy and convey this energy into transmitting antenna for radiation. The necessary information is then superimposed on Radio waves.

Classification of transmitters

Radio transmitters are classified on the following basis.

1. Types of modulation and
2. Types of service involved
3. Range of frequency involved

(a) Based on modulation used

- (i) Amplitude Modulation Transmitters (AM)
- (ii) Frequency Modulation Transmitters (FM)
- (iii) Pulse Modulation Transmitters(PM)

(b) Based on types of Service involved

- (i) Radio broadcast transmitters
- (ii) Radio telephone transmitters
- (iii) Radio telegraph transmitters
- (iv) TV transmitters
- (v) Radar transmitters

(c) Based on Frequency involved

- (i) Medium wave transmitters (MW) [550 KHz to 1650KHz]
- (ii) Short wave transmitters (SW) [3 to 30MHz]
- (iii) VHF and UHF transmitters VHF [30 to 300 MHz]
UHF [300 to 3000 MHz]
- (iv) Microwave transmitters [1GHz and above]

2.7.1 AM TRANSMITTERS

TYPES

AM transmitters can be classified as follows.

1. DSB/FC Transmitter : (i) Low level Modulation transmitter

(ii) High level Modulation transmitter

2. DSB/SC Transmitter
3. SSB/SC Transmitter
4. VSB Transmitter
5. Interrupted continuous wave (ICW) transmitter
6. Modulated continuous wave (MCW) transmitter

2.7.1.1 HIGH LEVEL MODULATION AM TRANSMITTER

Modulation takes place at the high power level of the carrier is called High level modulation.

AM transmitter which uses the process of High level modulation is called High level modulation AM transmitter.

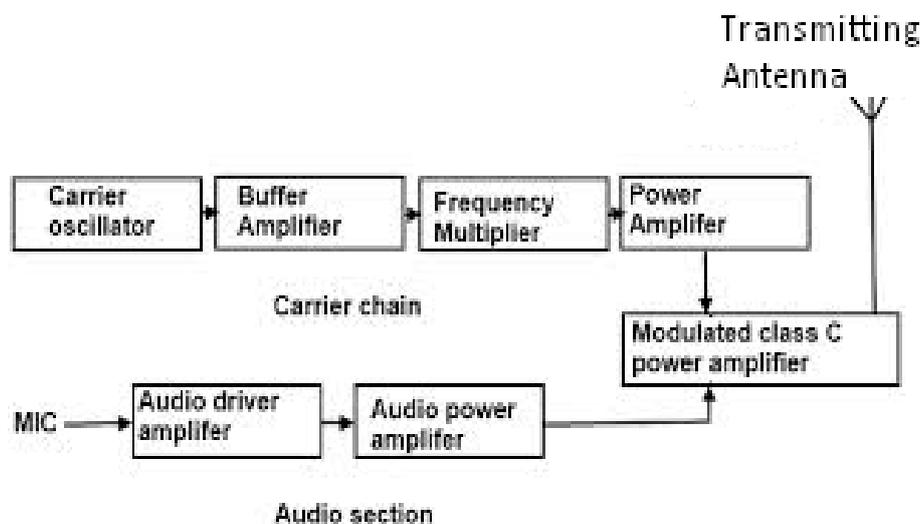


Fig.2. 6 High level modulated AM Transmitter

The above Fig 2.6 gives the block diagram of AM high level modulation transmitter using modulation at high power level of the carrier. The various elements and functions of each constituent stage are briefly given below.

Master oscillator (carrier oscillator)

- ❖ It is the carrier wave generator. It generates oscillation of desired frequency with high degree of stability.
- ❖ For better frequency stability crystal controlled oscillators are used.

Buffer Amplifier

- ❖ If Master oscillator directly drives the harmonic generator or class c power amplifier these may load the master oscillator causes frequency variation of it.
- ❖ To eliminate the interaction between master oscillator and transmitter load, a buffer amplifier is used between the master oscillator and harmonic generator.
- ❖ It isolates the master oscillator from succeeding stages.

Harmonic Generators (Frequency multiplier)

- ❖ It generates voltage at a frequency which is sub multiple of the carrier frequency.
- ❖ By this channel bandwidth can be increased.

Intermediate power amplifier

- ❖ RF voltage generated by the master oscillator has very small power, of the order of a few watts.
- ❖ Hence to increase the power level, class C amplifiers having high collector efficiency of the order of 70% are used to deliver an appreciable amount of power at succeeding stages.

Final Power Amplifier (Modulated class c amplifier)

- ❖ The last stage in RF channel is operated at the highest power level of the system and determines the amount of RF power that is delivered to the antenna.
- ❖ It is a class C tuned amplifier and is modulated by audio modulating voltage.

Audio Frequency Channel

- ❖ The audio frequency section of the transmitter generates the modulating voltage.
- ❖ It is designed to convert intelligence conveying sound signals into corresponding electrical signal of sufficient magnitude to impart a desired degree of modulation to the RF carrier wave.
- ❖ The energy conversion is performed by microphone.
- ❖ The audio power amplifier provides the required audio power to modulate the carrier.

Antenna matching networks

The high level modulated AM wave is transmitted to space through this network.

2.7.1.2 LOW LEVEL MODULATION

Modulation takes place at low carrier power is called low level modulation. Here modulation takes place before the final power amplification.

Low level modulation AM Transmitter

AM transmitter which uses the process of low level modulation is called low level modulation AM Transmitter.

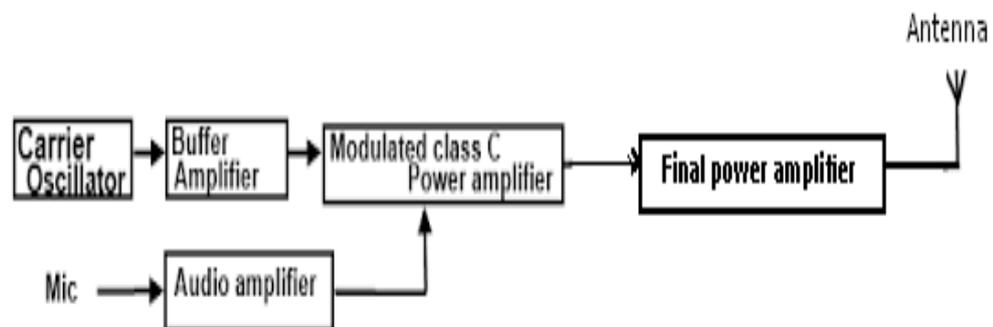


Fig.2.7 Low level AM Modulation transmitter

The above Fig 2.7 gives the block diagram of AM low level modulation transmitter. The various elements and functions of each constituent stage are briefly given below.

Master oscillator (carrier oscillator)

- ❖ It is the carrier wave generator.
- ❖ It generates oscillation of desired frequency with high degree of stability.
- ❖ For better frequency stability crystal controlled oscillators are used.

Buffer Amplifier

- ❖ If Master oscillator directly drives the harmonic generator or class c power amplifier these may load the master oscillator causes frequency variation of it.

- ❖ To eliminate the interaction between master oscillator and transmitter load, a buffer amplifier is used between the master oscillator and harmonic generator.
- ❖ It isolates the master oscillator from succeeding stages.

Modulated class C Amplifier

- ❖ Here the modulation takes place at an early stage of the transmitter where the carrier level is low.
- ❖ Such a system would require a modulating signal of low magnitude.
- ❖ For this reason; this type of modulation circuit is termed as low level modulation transmitter.

Audio Frequency channel

- ❖ The audio frequency section of the transmitter generates the modulating voltage.
- ❖ It is designed to convert intelligence conveying sound signals into corresponding electrical signal of sufficient magnitude to impart a desired degree of modulation to the RF carrier wave.
- ❖ The energy conversion is performed by microphone.

Audio power amplifier

- ❖ The audio power amplifier provides the required audio power to modulate the carrier.

Final Power Amplifier

- ❖ Since the power contained by modulated waves produce by a low level modulation circuit is quite low, it is usual practice to produce one or more stages of power amplification to increase the power level of the modulated wave to the required level.
- ❖ Since these power amplifier stages are required to amplify the carrier as well as the side bands equally they must possess sufficient bandwidth to accommodate these frequencies otherwise sideband cutting will occur at the output.
- ❖ Class B linear RF power amplifiers are usually employed for this purpose.

2.7.1.3 COMPARISON OF LOW LEVEL MODULATION AND HIGH LEVEL MODULATION

<i>Low Level Modulation</i>	<i>High level modulation</i>
1. Modulation takes at low carrier power level	1. Modulation takes place at high carrier power level.
2. Modulation takes place before final power amplification.	2. Modulation takes place at the final RF power amplification.
3. It needs more amplifier stages to get transmitted power after modulation.	3. It does not need amplifier stages after modulation.
4. It creates more noise.	4. Noise is reduced.
5. It needs wideband for amplifying carriers as well as all sidebands.	5. Narrow band can be obtained.
6. Efficiency is poor.	6. Efficiency is high.
7. Final stage should be class B or class A amplifier.	7. Final stage should be a class C amplifier.
8. Low power gain.	8. High power gain.

2.8 SSB TRANSMITTER

Single sideband transmission is a special form of AM radio telephony in which only one sideband is transmitted while the second side band and the carrier are suppressed.

Almost all long distance radio telephone networks use SSB signal transmission

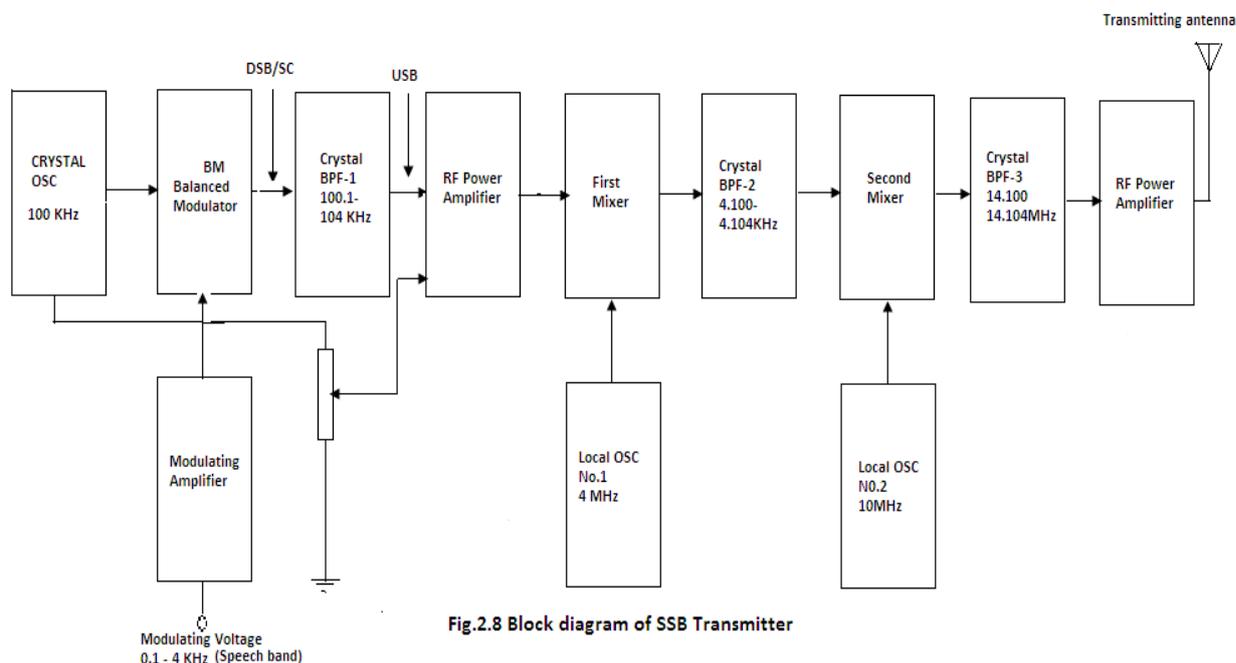


Fig 2.8 gives the block diagram of SSB transmitter used in Radio telephone networks. The various elements and functions of each constituent stage are briefly given below.

Crystal Oscillator

The crystal oscillator produces a stable carrier frequency of say 100 KHz.

Balanced modulator

The carrier is amplitude modulated by the speech voltage lying in the range of say 0.1 to 4 KHz in a suppressed carrier balanced modulator.

Band pass filter (BPF) 1

A Crystal band pass filter separates out one side band say the upper sideband (USB) of 100.1 – 104 KHz from the lower sideband of 96–99.9 KHz.

RF amplifier and first mixer

The USB is amplified by the RF amplifier and heterodyned in the first mixer with the output of 4 MHz oscillator No.1 produce sum and difference frequency sideband.

Band pass filter (BPF) 2

The sum frequency 4.100–4.104 MHz is selected by another crystal BPF 2.

Second Mixer

The output of the BPF2 is applied to the second mixer, to which is added the output of oscillator No.2 generating a frequency of say 10 MHz.

Band pass filter (BPF) 3

The sum frequency band of 14.100–14.104 MHz is passed by the crystal BPF 3.

Final RF Power Amplifier

The output from the BPF 3 is amplified by the RF power amplifier to raise the power level to the desired value and is then radiated through the antenna.

Advantages

1. Since SSB system occupies only half the frequency band as compared to AM–DSB system, narrow band transmission involved leads in bandwidth saving.
2. Since carrier is suppressed in SSB transmitter low power is necessary for transmission
3. Ordinary broadcast receiver cannot detect SSB signals. Thus certain amount of privacy is automatically included.

Disadvantages

1. Detection process is complicated
2. System is costlier than AM–DSB system

Applications

Almost all long distance radio telephone networks use SSB signal transmission

2.9 RADIO RECEIVERS

A Radio Receiver is a device that picks up the desired signal from the numerous signals propagating at the same time through the atmosphere, amplifies the desired signal to the requisite level, recovers from it the modulating signal and eventually displays it in the desired manner.

Radio receivers are classified in various ways depending on the nature of modulation, the method by which the modulation is superimposed, method of processing the modulating signal.

Classification of Radio Receivers

Depending on the method of superimposing the modulating signal, the corresponding receiver can be classified as

- (1) AM receiver
- (2) FM receiver
- (3) SSB receiver

AM Receiver

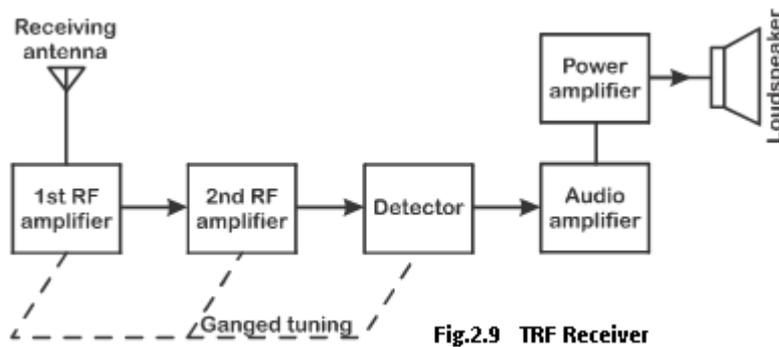
Receiver which is employed for the reception of AM waves received from AM transmission is called AM Receiver.

AM receiver may be one of the types below:

- (1) TRF receiver (Tuned Radio Frequency Receiver)
- (2) Super heterodyne Receiver
- (3) SSB receiver

2.9.1 TUNED RADIO FREQUENCY RECEIVER (TRF RECEIVER)

The TRF receiver is a simple receiver employing straight forward circuit arrangements. Fig 2.9 shows the block schematic of such a receiver.



- ❖ It uses two or three stages of RF amplification all tuned simultaneously to the desired frequency.
- ❖ The signal intercepted by the receiving antenna is selected by tuned LC circuits at the input of each stage and amplified by several stages of tuned RF amplification.
- ❖ The amplified RF signal is demodulated in the detector stage so that only audio frequency variations of the carrier remain.
- ❖ The AF output from the detector is amplified by one or more stages of audio amplifier till it is strong enough to drive the loudspeaker.
- ❖ The loud speaker converts the audio variation into sound waves.

Drawbacks of TRF receivers

1. Instability in gain
2. Poor selectivity
3. Bandwidth varies with frequency range
4. Creates problems of adjacent channel interference
5. Some of the sidebands are attenuated when selectivity is too sharp.

2.9.2 SUPER-HETERODYNE RECEIVER

The word super-het stands for SUPER-sonic HET-aerodyne which means that heterodyning takes place with resulting output frequency higher than audio.

Super heterodyne is a form of heterodyne reception in which one or more changes take place before detection.

- ❖ A Super heterodyne receiver may thus be defined as one in which one or more changes of frequency takes place before AF signal is extracted from the modulated wave.
- ❖ The problems of TRF receiver such as instability, insufficient adjacent frequency rejection and bandwidth variation are all solved by the use of a super heterodyned receiver.

The block diagram of super heterodyne receiver is shown in Fig 2.10.

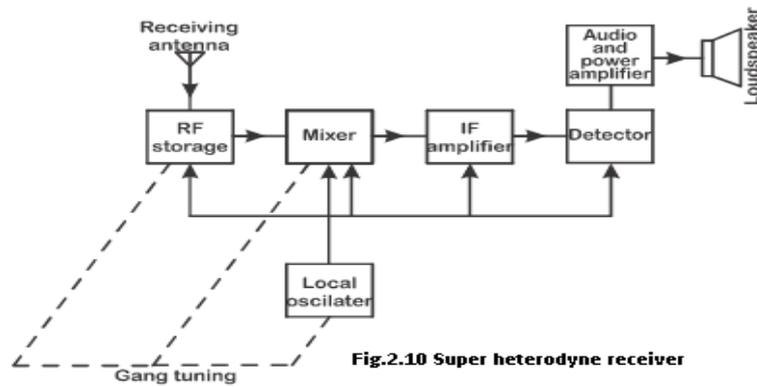


Fig.2.10 Super heterodyne receiver

The working of super heterodyne receiver with the help of circuit blocks is discussed below.

Antenna

- ❖ The antenna intercepts the electromagnetic waves.
- ❖ Voltages corresponding to all transmitting frequencies are induced in the antenna and are carried to the receiver input with the help of feeder wires.

RF amplifier

- ❖ The RF stage is a small signal tuned amplifier.
- ❖ The RF amplifier selects the wanted frequency and amplifies the signal to a suitable level before feeding to the mixer.

Frequency converter

- ❖ The frequency converter stage consists of a local oscillator and a frequency mixer.
- ❖ The local oscillator generates a frequency f_0 which is higher by f_{IF} than the incoming frequency f_s .
- ❖ The tuned signal frequency f_s and f_0 are fed to the mixer.
- ❖ The output of the mixer consists of various inter modulation frequencies.
- ❖ The difference frequency, f_{IF} is picked up by the tuned circuit in the output of the mixer

$$\text{ie, } f_{if} = f_0 - f_s$$

(Intermediate frequency)

IF amplifier

- ❖ There may be two or more stages of IF amplification.
- ❖ The IF amplifiers are fixed frequency tuned amplifiers and are tuned to the intermediate frequency. The IF amplifiers provide most of the receiver amplification and selectivity.

Detector

- ❖ The output of the last IF amplifier stage is fed to a detector stage.
- ❖ The diode in its linear region is generally used as a detecting device.
- ❖ The output of the detector contains original intelligence broadcast by the transmitter.

AF amplifier

- ❖ The detected intelligence which is an audio frequency signal is first fed to the AF amplifier and then to the AF power amplifier stage.
- ❖ This gives sufficient power to drive the loudspeaker.

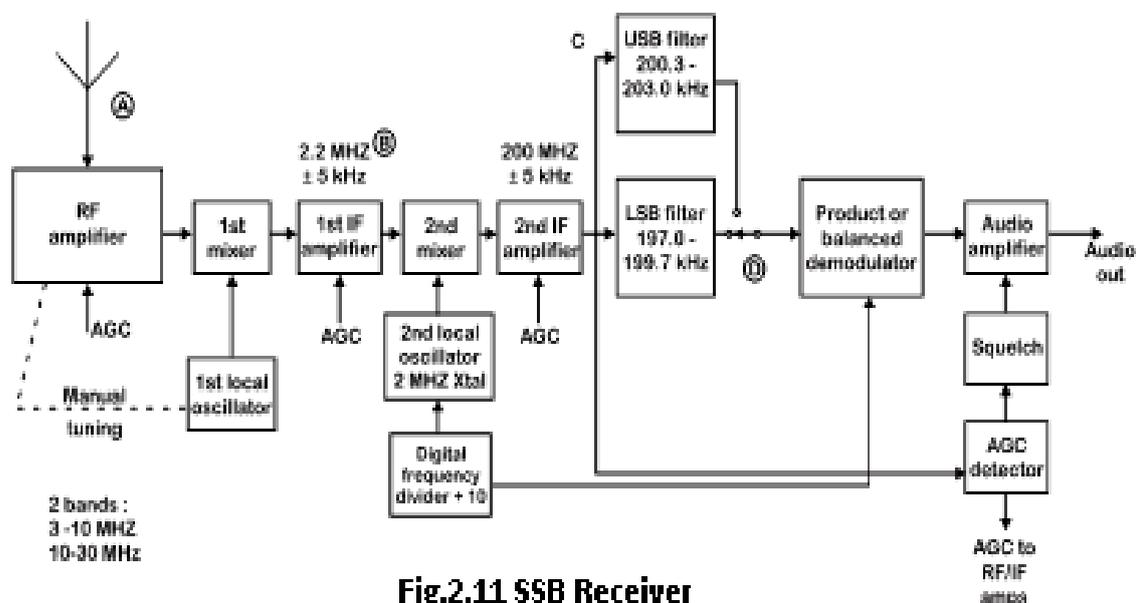
Loud speaker

The loudspeaker converts the electrical signal to the sound signal and reproduces the original intelligence.

Advantages

1. Greater gain
2. Better Sensitivity
3. Improved image frequency rejection
4. Better selectivity

2.9.3 SSB RECEIVER (SINGLE SIDEBAND RECEIVER)



The figure.2.11 shows the block diagram of a communications receiver designed for SSB reception in the HF range (3 to 30 MHz).

- ❖ The SSB receiver is similar to that of a standard double conversion AM receiver.
- ❖ The first IF has a bandwidth of 10 kHz centered at 2.2 MHz.
- ❖ The second IF also has a bandwidth of 10 kHz, but centered at 200 kHz, down to the SSB filter inputs.
- ❖ This band pass is wide enough to pass a normal AM signal.

1st Mixer

- ❖ Here the first local oscillator and RF amplifier are manually tuned in two switched bands (3 to 10 MHz, 10 to 30 MHz).

Second Mixer

- ❖ In this the second local oscillator is crystal controlled at 2 MHz.
- ❖ Its output is divided by 10 in a digital counter to provide the 200 kHz carrier signal for the demodulator.

Sideband filters

- ❖ Two SSB filters follow the second IF amplifier.
- ❖ The USB filter passes the IF upper sideband of 200.3 to 203 kHz and rejects the lower sideband.
- ❖ The LSB filter passes the 197 to 199.7 kHz IF lower side band and rejects the upper sideband.

Balanced Demodulation

- ❖ A switch selects the desired filter and connects it to the product detector.

- ❖ The output from the detector is passed through a gated audio amplifier.
- ❖ It turns off the output to keep the noise down when the signal level drops below a preset threshold.

AGC control

- ❖ The output of second IF amplifier is rectified to provide the AGC voltage for the squelch circuit, RF and IF amplifiers.
- ❖ The figure also shows spectra in the HF receiver for an LSB signal.

2.10 SELECTION OF IF (INTERMEDIATE FREQUENCY).

Intermediate frequency (IF) should be carefully chosen as,

- ❖ High IF results in poor selectivity and therefore poor rejection of adjacent channels
- ❖ High IF results in problems in tracking of signals in the receivers.
- ❖ Image frequency rejection becomes poor at low IF or if very high.
- ❖ The chosen IF frequency should be free from radio interference.
- ❖ The broadcasters settled the standard common IF frequency is 455 KHz for AM Radio and 10 MHz for FM Radio.

2.11 AGC (AUTOMATIC GAIN CONTROL)

AGC smoothes out variations in signal strength due to fading on long distance shortwave reception, and prevents the overloading of the last IF amplifier

Principle of Operation

- ❖ Practically all receivers for amplitude modulation use Automatic gain control circuit to maintain the carrier voltage at the second detector input, approximately constant.
- ❖ A dc bias voltage applied to a certain number of RF, IF and mixer stages changes the Trans conductance of the devices, which results in variation in gain.
- ❖ The use of AGC permits tuning to stations of varying signal strengths without appreciable change in the size of the output signal.
- ❖ A receiver with AGC may be tuned from strong to weak signals without the necessity of resetting the manual volume control.

Types of AGC

There are three types of AGC,

1. Simple AGC
2. Ideal AGC
3. Delayed AGC

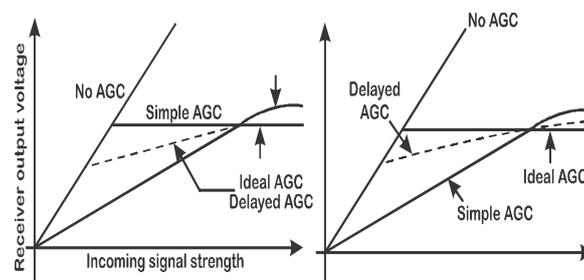


Fig.2.12 AGC

Fig 2.12 shows the variations of AGC characteristics

Simple AGC

This permits tuning to stations of varying signal strength without appreciable change in the size of the output signal.

Ideal AGC

In this, no AGC acts unless signal strength reaches a predetermined level and after this level, a constant average output is obtained irrespective of any increase in signal strength.

Delayed AGC

- ❖ It shows that AGC bias is not applied until the signal strength has reached a particular level, and after this a strong AGC bias is applied so that the output becomes more or less independent of any further increase in incoming signal strength.
- ❖ Thus the problem of reducing the receiver gain for weak signals gets solved using delayed AGC or DAGC.

Review Question

Part – A

(Two Marks)

1. Define Modulation.

In wireless communication, the low frequency information is super imposed on a high frequency signal, by using a process called modulation.

2. Define amplitude modulation.

Amplitude modulation is the process in which the amplitude of the carrier signal is varied in accordance with the instantaneous value of the modulating voltage, keeping frequency and phase angle constant.

3. State the types of modulation.

1. Amplitude modulation
2. Frequency modulation
3. Phase modulation

4. State the advantages of phase shift method of SSB Generation.

1. It produces true direct single side band modulation.
2. Easy to select the lower side band (or) upper side band signal
3. A sharp cut-off side band filter is not required.

5. State the disadvantages of SSB signal.

1. The transmitter and receiver are complicated and their performance required is of high standard.
2. The cost of SSB receiver is higher than DSB receiver.

6. What is difference between high level modulation and low level modulation?

High level modulation

In this, the modulating signal is first amplified and then modulated in high power level.

Low level modulation

In this, the modulating signal is modulated at low power level and then signal is amplified.

7. What is the Band width required for Amplitude Modulation?

Twice the frequency of modulating signal

8. What are the methods of SSB generation?

1. Filter method
2. Phase shift method.

9. Define Modulation index. What is the range of Modulation index?

It is defined as the ratio of amplitude modulating signal to the amplitude of the carrier.

$$\text{i.e., } m_a = E_m/E_c$$

Between 0 and 1

10. What are the outputs of balanced modulator?

1. USB (Upper side Band)
2. LSB (Lower side Band)

11. Define single side band.

In DSB system, the audio information is found in both lower and upper side band. To save the bandwidth, it is sufficient to transmit only one side band and rejecting the other side band and carrier. The system which adopts this technique is known as SSB system.

12. What is AGC? Mention the types of AGC.

AGC smoothes out variations in signal strength due to fading on long distance shortwave reception, and prevents the overloading of the last IF amplifier.

Types of AGC

1. Simple AGC
2. Delayed AGC

13. Mention the types of radio receiver.

1. Amplitude modulation broadcast receiver
2. Frequency modulation broadcast receiver
3. Television receiver
4. Communication receiver
5. Code receiver
6. Radar receiver

14. What is the degree of modulation?

1. Under modulation
2. Critical modulation
3. Over modulation

15. State the advantages and disadvantages of single side band.

Advantage of Single side band system

1. Band width reduction
2. Power reduction
3. Reduction in operating cost
4. Some privacy is automatically provided.
5. Improvement of signal to noise ratio from 9 to 12 dB.

Disadvantage of Single side band system

It needs Complex receiver circuit as compared to other system.

15. Why we need RF amplifier in a radio receiver?

To improve noise limited sensitivity

16. What is the value of Bandwidth of a VSB signal?

Slightly more than the highest modulating frequency

17. Write the formula for modulation index of AM signal.

$$m_a = \frac{E_c \text{ max} - E_c \text{ min}}{E_c \text{ max} + E_c \text{ min}}$$

18. Differentiate High level and low level AM transmitter.

In high level AM transmitter modulation takes place at high power level of the carrier.

In low level AM transmitter modulation takes place at low power level of the carrier.

19. Expand SSB – SC, DSB – SC, and VSB.

SSB-SC → Single Side Band Suppressed carrier

DSB-SC → Double Side Band Suppressed carrier

VSB → Vestigial Side Band

20. State the advantages of VSB.

1. Bandwidth is saved
2. 75% of the total power is saved
3. Generation of VSB is easy when carrier is present.
4. Detection is also easy

21. State the disadvantages of SSB.

1. It needs special synchronous detection process
2. It is difficult to detect or demodulate the original modulating signal from SSB signals.

22. What is the range of wave length of visible light in electromagnetic frequency spectrum?

Wave length of visible light is 0.1μm to 1 μm

23. Mention the frequency band which is suitable for satellite communication.

300 MHz to 3 GHz.

24. What do you mean by over modulation? What will be the effect of over modulation?

- ❖ When modulation index is greater than 1 the modulated wave will be distorted. This is called over modulation.
- ❖ Over modulation will distort the output AM wave.

25. Give the expression of the output voltage of Amplitude modulated wave.

$$e = E_c (1 + m_a \cos \omega_m t) \cos \omega_c t$$

26. Give the expression of the output voltage of USB and LSB of AM wave.

1. $\cos (\omega_c + \omega_m)t$ is the upper side band (USB) term of

$$\frac{m_a E_c}{2}$$

angular frequency, $(\omega_c + \omega_m)$

2. $\frac{m_a E_c}{2}$

$\cos (\omega_c - \omega_m)t$ is the lower side band (LSB) term of angular frequency $(\omega_c - \omega_m)$

Part – B

(Three Marks)

1. Draw the Structure of AM wave.
2. Represent the AM wave in frequency domain.
3. Represent the AM wave in Time domain
4. Compare low level modulation and high level modulation.
5. Compare AM, DSB, AM DSB/SC, AM–SSB/SC.
6. Mention the advantages and disadvantages of SSB system.
7. Briefly explain how VSB can generate.
8. What do you mean by AGC? Draw the characteristics of AGC and briefly explain it?
9. State the advantages and applications of VSB modulation
10. Draw the Electromagnetic frequency spectrum.
11. What do you mean by suppressed carrier modulation? What are its advantages? Mention its applications.
12. Define: modulation. Explain the need for modulation.
13. Derive the expression for AM wave and its side bands.

PART-C

(Ten Marks)

1. Draw the block diagram of Low level modulation AM transmitter and explain its working.
2. Draw the block diagram of high level modulation AM transmitter and explain its working.
3. Explain the working of SSB transmitter. Mention its advantages.
4. Explain the working of TRF Receiver. Mention its draw backs.

5. Explain the working of Super- heterodyne Receiver and Mention how is it superior to TRF receiver.
- 6 Explain the working of SSB receiver. Mention its advantages
7. (i) Explain the various types of AGC (ii) Compare the Characteristics of AM and FM.
8. (i) Compare DSB, SSB and VSB (ii) Explain the relationship between frequency and wavelength
9. Explain AM – VSB system. Mention where is it used?



UNIT - III

FREQUENCY AND PULSE MODULATION

LEARNING OBJECTIVES

On completion of this unit, the students must be able to....

- To understand the Frequency modulation process
- To learn about different types of FM transmitters
- To understand the concept of Pulse modulation
- To know the generation and detection of various pulse modulation processes

3.0 INTRODUCTION

In angle modulation, the information signal may be used to vary the carrier frequency, giving rise to frequency modulation, or it may be used to vary the angle of phase lead or lag, giving rise to phase modulation.

Since both frequency modulation and phase modulation are parameters of the carrier angle, which is a function of time, the general term angle modulation covers both.

Frequency and phase modulation have some very similar properties, but also some marked difference.

- ❖ Compared to amplitude modulation, frequency modulation has certain advantages. Mainly,
 1. S/N ratio can be increased.
 2. Interference at the receiver is more easily suppressed.
 3. Modulation process can take place at low-level power stage in the transmitter.

3.1 FREQUENCY MODULATION

Definition

Frequency modulation is a system in which the amplitude of the modulated carrier is kept constant, while its frequency and rate of change are varied by the modulating signal amplitude

3.2.1 Expression of FM

Let the modulating signal be expressed as

$$e_m = E_m \cos \omega_m t$$

Where E_m = amplitude of the modulating signal

$\omega_m =$ its angular frequency.

- ❖ After, frequency modulation, frequency of the carrier no longer remains constant but varies with time in accordance with the instantaneous value of the modulating signal, f_m .
- ❖ Thus, instantaneous frequency; f_i of the carrier after frequency modulation is given by,

$$\begin{aligned} f_i &= f_c + Kf_c e_m \\ &= f_c + Kf_c E_m \cos \omega_m t \\ \text{i.e.,} \quad f_i &= f_c (1 + K E_m \cos \omega_m t) \end{aligned}$$

Where,

f_c = frequency of the un modulated carrier

K = constant of proportionality

As the extreme value of the cosine function $\cos \omega_m t = \pm 1$,

- ❖ The maximum value of the instantaneous frequency is given by,

$$\begin{aligned} f_i &= f_c (1 + K E_m) \\ &= f_c + K E_m f_c \\ &= f_c + \Delta f \end{aligned}$$

Where, $\Delta f = K E_m f_c$

- ❖ Since, f_i deviates from f_c by the factor of Δf , Δf is called as frequency deviation of FM.
- ❖ The instantaneous amplitude of a frequency modulated carrier is given by,

$$e_i = E_c \sin (\omega_c t + m_f \sin \omega_m t)$$

Where m_f is called the modulation index for an FM wave

3.2.2 FREQUENCY DEVIATION

The change of frequency of the carrier wave with respect to the amplitude of the modulating signal is called as frequency deviation and is denoted by ' Δf '

ie.)
$$\Delta f = K E_m f_c$$

Where, K = constant of proportionality

E_m = amplitude of modulating signal

f_c = frequency of the un modulated carrier.

3.2.3 MODULATION INDEX

- ❖ **The ratio of frequency deviation Δf , to the maximum frequency of the modulating signal f_m is called as modulation index of FM. It is denoted by ' m_f '.**

Since, $\Delta f = K E_m f_c$, the frequency deviation varies with E_m , the amplitude of the modulating signal.

- ❖ So frequency of modulating signal f_m , does not affect Δf .

But since,
$$m_f = \frac{\Delta f}{f_m}$$

The frequency of f_m , affects the modulation index m_f .

ie) m_f decreases with f_m .

In FM, there is no limit for m_f . Its value varies with J coefficient.

3.2.4 Comparison of modulation index m_a of AM and

m_f of FM

m_a	m_f
1. $m_a = \frac{E_m}{E_c}$	1. $m_f = \frac{\Delta f}{f_m}$
2. m_a varies with E_m .	2. m_f also varies with f_m .
3. f_m does not affect m_a	3. f_m affects m_f
4. Maximum limit of m_a is	4. There is no limit for m_f . It varies with J coefficient.
5. Value of m_a affects S/N ratio.	5. m_f does not affect S/N ratio.

3.2.5 WAVEFORM REPRESENTATION OF FM

- ❖ In frequency modulation, the frequency of the carrier swings above and below its center or resting frequency, f_c . The amount of change in frequency, Δf is dependent on the amplitude of the modulating signal.
- ❖ The instantaneous frequency as a function of time and the wave shape of FM wave is shown in Fig 3.1.

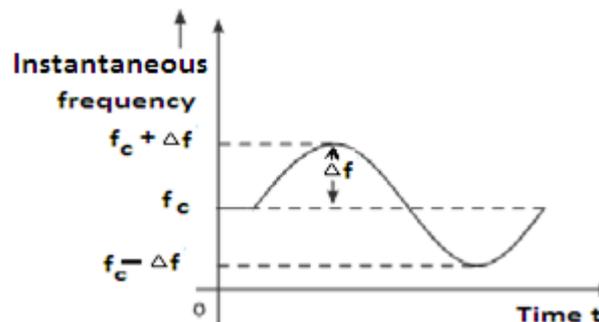


Fig.3.1 (a) Instantaneous frequency –time curve

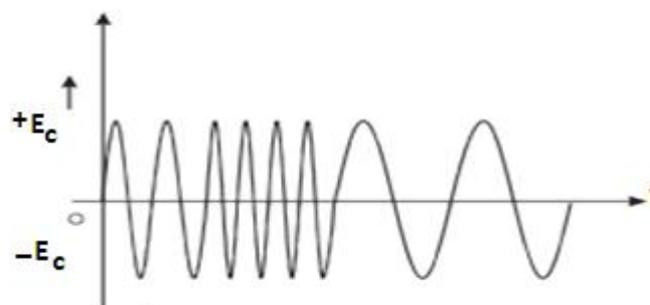


Fig. 3. 1 (b) Instantaneous voltage -time curve (FM in Time domain)

- ❖ As can be seen from the Fig 3.1, when the frequency deviation Δf is maximum (i.e.) when E_m is maximum, the frequency of FM wave is maximum.

3.2.6 FREQUENCY SPECTRUM OF FM

(Side bands produced in FM)

Different frequency components in an FM wave can be determined in the same way as followed for AM wave i.e.) by expanding the expression for the waves.

When carrier voltage e_c , is frequency modulated by a single sinusoidal voltage, e_m , the modulated carrier voltage is,

$$e_i = E_c \{ J_0(m_f) \sin \omega_c t + J_1(m_f) [\sin(\omega_c + \omega_m)t - \sin(\omega_c - \omega_m)t] + J_2(m_f) [\sin(\omega_c + 2\omega_m)t - \sin(\omega_c - 2\omega_m)t] + J_3(m_f) [\sin(\omega_c + 3\omega_m)t - \sin(\omega_c - 3\omega_m)t] + J_4(m_f) [\sin(\omega_c + 4\omega_m)t - \sin(\omega_c - 4\omega_m)t] + \dots \}$$

❖ It can be shown that the output consists of a carrier and an apparently infinite number of pairs of sidebands, each preceded by J coefficient. Where J_0, J_1, J_2, \dots are the coefficients of zero, first, second order for the Bessel functions.

❖ Bessel function gives the amplitude of the carrier ($n = 0$) and side frequencies ($n=1, 2, 3, \dots$).

A typical plot of the Bessel function is given in Fig.3.2.

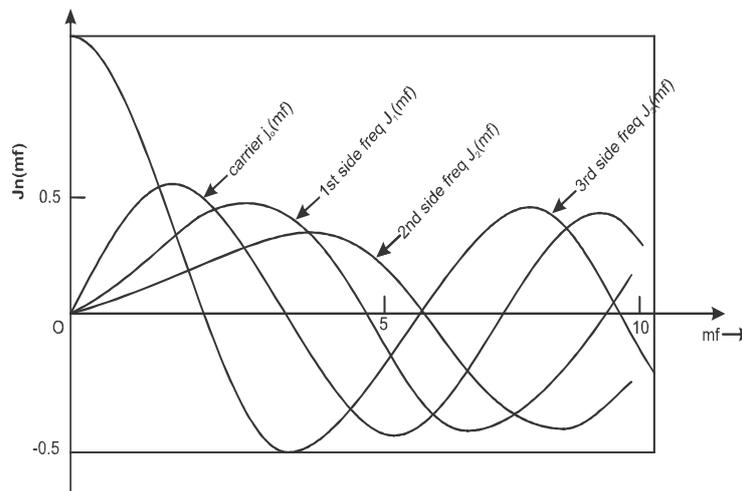


Fig.3.2

Bessel functions for the first kind (carrier) and the first three side frequencies for an FM wave of $E_{Cmax} = 1V$.

From the mathematical expression the following conclusion can be made.

- (i) FM wave has an infinite no/ of sidebands
- (ii) Sidebands are separated from the carrier by $f_m, 2f_m, 3f_m, \dots$ and reoccur.
- (iii) As n increases, J_n decreases.
- (iv) With decrease in J , the amplitude of sidebands also decreases.
- (v) The modulation index m_f , thus determines the number of significant sideband components
 - ❖ Using Bessel function table it is possible to evaluate the size of the carrier and each sideband for each specific value of modulation index m_f . When this is done, the frequency spectrum of the FM wave for that particular value of m_f may be plotted.
 - ❖ The general structure of frequency spectrum of FM is shown in Fig. 3.3

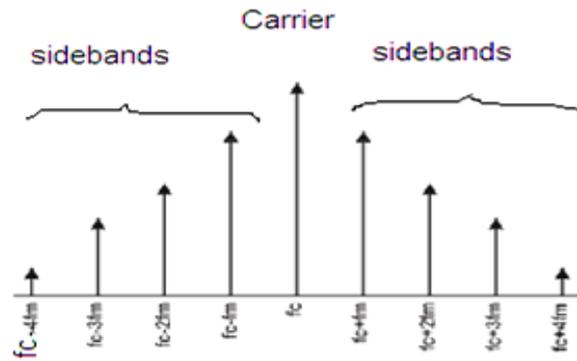


Fig.3.3

It is usual practice to notify all the side bands having amplitude more than 5% of the carrier as significant side bands and neglect the remaining side bands with less than 5% of the carrier.

3.2.7 Comparison of FM and AM.

	AM	FM
1	AM has three frequency components namely carrier USB, and LSB	FM has infinite no/. of frequency components namely carrier, as well as infinite no/. of sidebands.
2	S/N ratio is poor (small)	S/N ratio is good (High)
3	Bandwidth is only $2f_m$	FM may be narrow band or wideband. Depends on J coefficient and m_f .
4	Modulation index, $m_a = \frac{K E_m}{E_c}$	Modulation index, $m_f = \frac{\Delta_f}{f_m}$
5	Circuitry is less complex	Circuitry is more complex
6	Used in wide area coverage	Area of coverage is small (Applicable only to shortwave communication)
7	Adjacent channel interference is more	Less adjacent channel interference.
8	Noise cannot be reduced	Noise can be reduced
9	Transmitting power depends on m_a (depth of modulation)	Transmitting power is constant.
10	m_a should be less than unity (1).	No limits of m_f . But huge value will lead to increase in bandwidth.

3.3 GENERATION OF FM

- ❖ The primary requirement of an FM generator is a variable output frequency, the variation being proportional to the instantaneous amplitude of the modulating voltage.
- ❖ If the capacitance or inductance of an LC oscillator tank circuit could be varied in such a way that the variation in frequency is proportional to the voltage supplied by the modulation circuits then true FM will be obtained.
- ❖ There are a number of devices whose reactance can be varied by the application of voltage.

3.3.1 METHODS OF GENERATING FM

FM can be generated in two ways.

1. Direct Method (Varactor diode modulator)
2. Indirect Method (Armstrong method)

The most common device is the varactor diode. This method of generating FM is called direct method.

There are indirect methods of FM generation in which the frequency of the oscillator is not varied.

3.3 FM TRANSMITTER

In FM transmitters, we can use both direct and indirect FM. So for FM transmission we can use both Direct and Indirect FM transmitters.

3.3.1 DIRECT FM TRANSMITTER

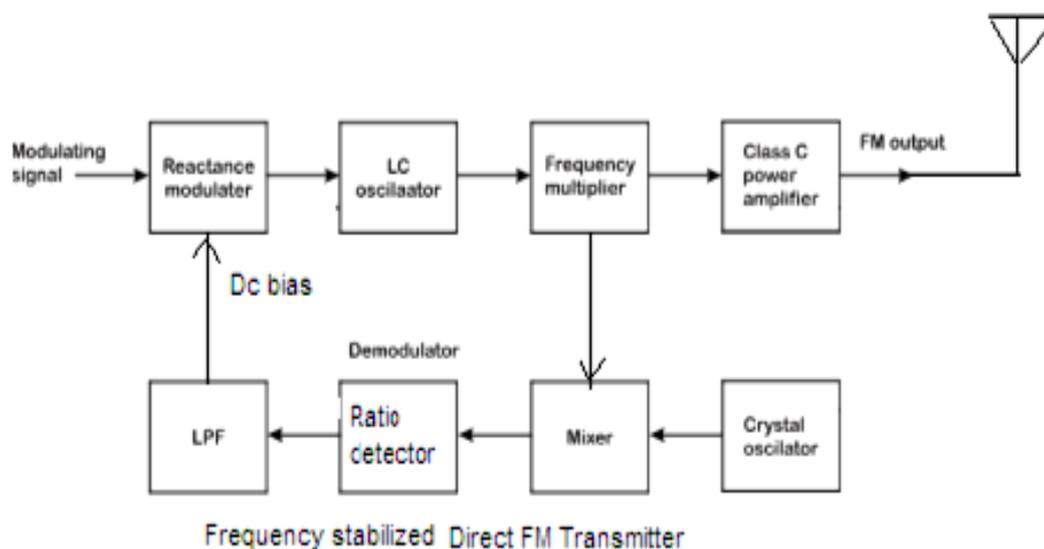


Fig.3.4

The block schematic of Direct FM transmitter is shown in Fig. 3.4.

- ❖ In this an LC oscillator may be directly frequency modulated to produce relatively large deviation.
- ❖ However, direct FM at the final carrier frequency is not feasible because of the problem of maintaining high frequency stability of the carrier while at the same time obtaining adequate frequency deviation.

- ❖ Stability can be improved by using automatic frequency control (AFC). Above fig 3.4 shows one possible arrangement.
- ❖ Here, a sample of the output signal, taken from the final driver stage, is mixed with a signal from a stable crystal oscillator.
- ❖ The difference frequency output from the mixer is fed into the discriminator circuit, where it is demodulated.
- ❖ The demodulated output consists of the original modulating signal plus any variations caused by drifting of the nominal frequency of the LC oscillator.
- ❖ The low pass filter removes the signal component while leaving the drift component which is fed back as a control bias to the oscillator.
- ❖ This is connected in such a way as to shift the nominal frequency in the desired direction to reduce the drift.
- ❖ Class c amplifiers can be used to amplify FM signals since the amplitude distortion introduced by the class. C amplifier has no effect on the modulation.
- ❖ This makes for more efficient, FM transmitters compared to AM transmitters.

3.3.2 INDIRECT FM TRANSMITTERS (ARMSTRONG METHOD)

Phase modulation may be used to indirectly frequency modulate an oscillator. This allows a crystal controlled oscillator to be used.

A very popular indirect method of achieving FM is known as Armstrong method after its inventor. In this method, the initial modulation takes place as an amplitude modulated DSB/SC signal so that a crystal controlled oscillator can be used if desired.

Following fig 3.5 shows the block schematic of Indirect. FM Transmitter using Armstrong method.

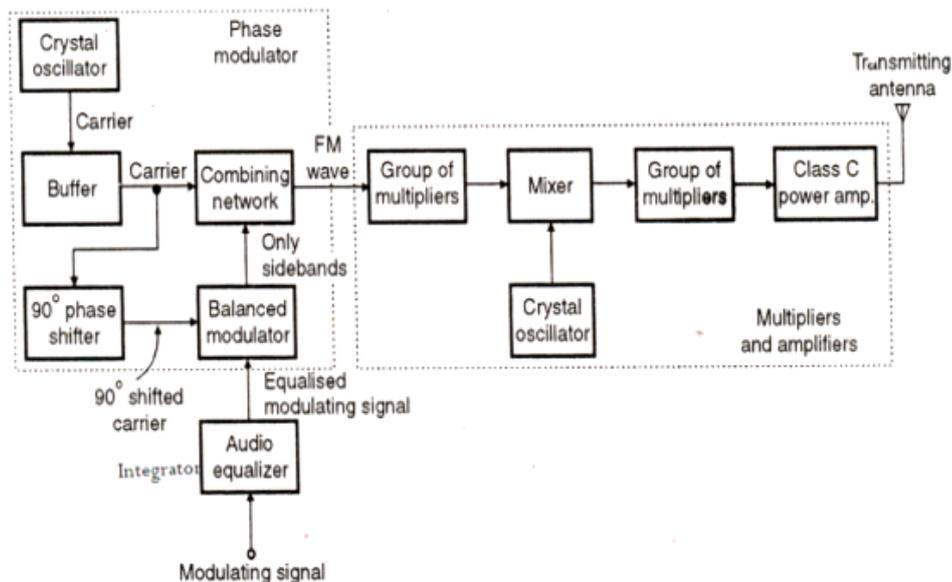


Fig.3.5

Figure fig 3.5 Indirect FM Transmitter Armstrong method.

- ❖ In this, the crystal oscillator generates the sub carrier, which can be low, say on the order of 100 KHz.
- ❖ One output of the oscillator is phase shifted by 90° to produce the sine term, which is then DSB/SC modulated in the balanced modulator by the modulating signal.

- ❖ This is combined with the direct output from the oscillator in the combining network, the result then being Phase modulated signal. The modulating signal is passed through an integrator which results in an equivalent frequency modulation.
- ❖ At this stage the frequency deviation will be low, so it is given to group of multipliers which are used to increase the peak deviation.
- ❖ Frequency multiplication can be achieved by passing the signal through a class C amplifier and tuning the output to the desired harmonic.
- ❖ With frequency multiplication the instantaneous carrier frequency is multiplied.

3.3.3 FM STEREO TRANSMITTER (BROADCAST)

Stereo FM transmission is a modulation system in which sufficient information is sent to the receiver to enable it to reproduce original stereo material.

The block diagram of a stereophonic FM transmitter is shown in fig.3.6 (a).

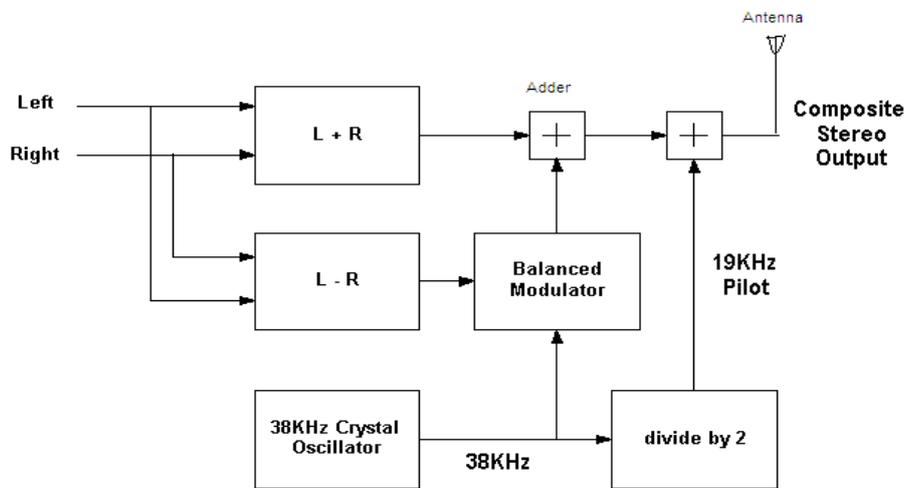


Fig.3.6 Stereo FM Transmitter

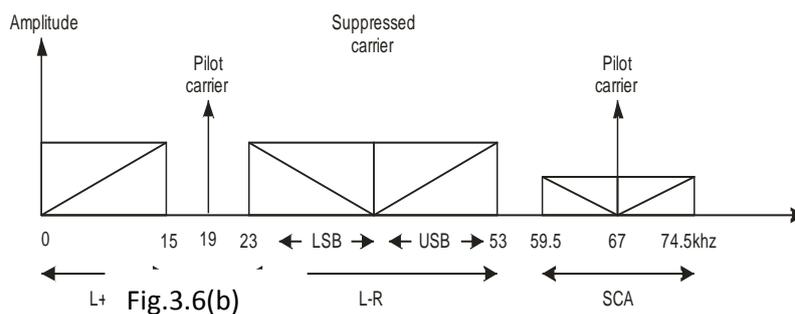


Fig. 3.11

- ❖ Two channels are used in this system. These channels left channel and right channel.
- ❖ The signal is picked up by two microphones located at right and left of the orchestra.
- ❖ The two channels are passed through a matrix which produces two outputs: Sum (L+R) and Difference (L-R).
- ❖ The sum signal occupies the range of 0 Hz to 15 KHz.

- ❖ The difference signal (L-R) is fed to the balanced modulator.
- ❖ Another input to the balanced modulator is 38 KHz subcarrier signal.
- ❖ The frequency of 19 KHz subcarrier generator is doubled by a frequency doubler.
- ❖ The 38 KHz subcarrier output of frequency doubler acts as another input to balanced modulator.
- ❖ The balanced modulator suppresses the carrier and generates two sidebands: LSB of 23 KHz and USB of 38 KHz to 53 KHz.
- ❖ Thus the sidebands produced by the difference signal occupy the frequency range from 23 KHz to 53 KHz.
- ❖ The output difference signals in the output adder preceding the frequency modulator.
- ❖ The FM output of frequency modulator is amplified by suitable amplifiers and then transmitted.
- ❖ The spectrum of stereo FM modulating signal is shown in fig 3.6 (b).

3.4 FM RECEIVER

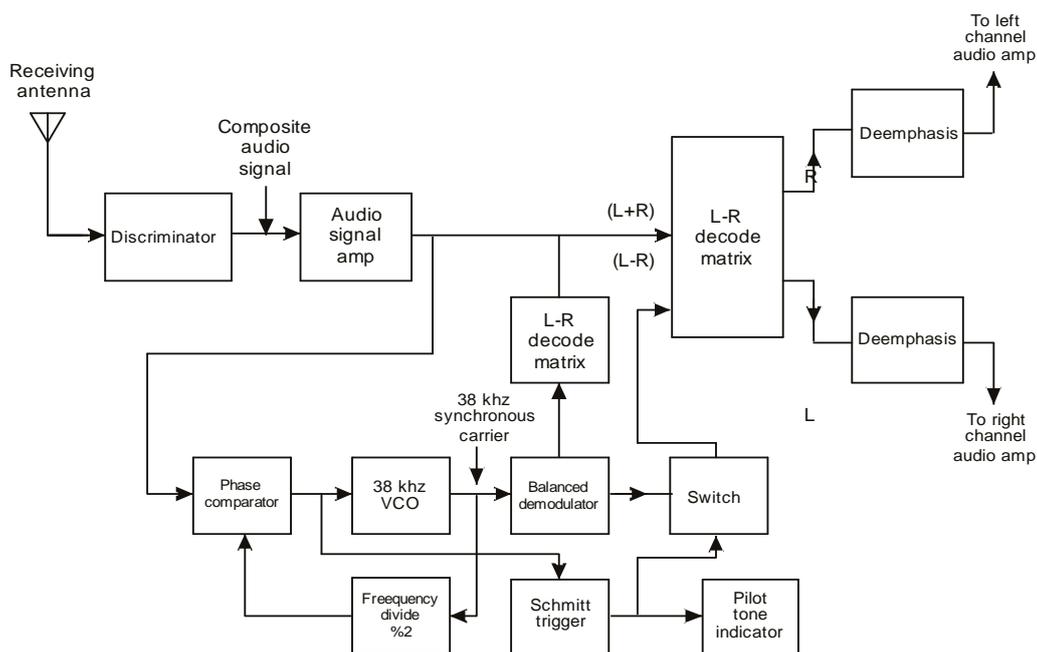
- ❖ FM broadcast receivers operate in UHF band between 88 and 108 MHz.
- ❖ They are spaced apart 200 KHz with a maximum frequency deviation of 75 KHz around the carrier frequency. Here an IF of 10.7 MHz is used.
- ❖ The figure 3.7 shows the block diagram of FM broadcast receiver.

RF tuned amplifier

- ❖ It receives the signal from the antenna. The tuning circuit of RF section selects the particular frequency. It has to receive the signal than the range of 88-108 MHz.
- ❖ The output of RF amplifier is given the mixer stage.

Local Oscillator

- ❖ The local oscillator generates a frequency which is 10.7 MHz higher than the incoming signal frequency.



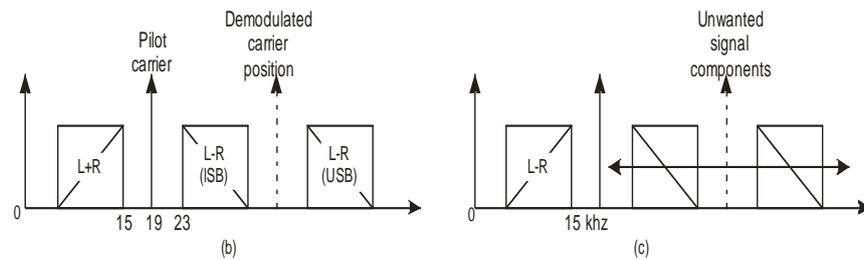


FIG.3.7

The tuning of local oscillator is ganged with the RF amplifier.

- ❖ The output of local oscillator is given to the mixer stage.

Mixer

- ❖ Mixer stage mixes the two frequencies from local oscillator and RF amplifier.
- ❖ It produces the difference frequency.
- ❖ The output of mixer is given to the IF tuned amplifier.

IF tuned limiting amplifier

- ❖ These are amplitude limiters.
- ❖ The amplitude limiters are amplifier circuits which are used to eliminate amplitude modulation and amplitude modulated noise.
- ❖ The amplitude limiter limits the amplitude of the IF signal to a predetermined value.
- ❖ Hence the IF amplifier and the limiting amplifier are tuned to give the desired band pass characteristic centered at 10.7 MHz with a bandwidth to pass the desired signal.
- ❖ The output of limiting amplifier drives the FM detector.

Discriminator

- ❖ It is the FM detector. It is normally Foster Seeley discrimination. It produces the modulating signal at its output. It incorporates AFC circuit.

De-emphasis

- ❖ It is provided in all FM receivers. In transmitter, to boost up the amplitude level of high frequency component's process called pre-emphasis is used.

Hence in the receiver to bring back the signal voltage to the original level reverse process called de-emphasis is done. It brings back the amplitude of high frequency terms to the original level.

Volume and tone control

- ❖ This control varies the volume and tone of the detected audio output.

Audio amplifier

- ❖ It amplifies the audio signal from the detector. The output of audio amplifier is given to a loud speaker.

AFC circuit

- ❖ The output of discriminator is taken and given to the AFC circuit. This circuit includes a varactor trimmer. Any deviation in IF frequency gets corrected by applying the error voltage to the varactor diode. The varactor diode in turn corrects the local oscillator frequency, thereby the IF gets corrected.

Delayed AGC detector

- ❖ AGC detector is used to control the gain of the various amplifiers within the specified limit. Here delayed AGC is used. The IF output is taken. It is rectified and the resulting DC voltage is applied to the input of IF and RF amplifiers.

3.5 COMPARISON OF PM AND FM

1. Similarities in both PM and FM, modulation index depends on the instantaneous amplitude of the modulating signal e_m .
2. If $m_p = m_f$, modulation index of PM and FM are same, then the frequency spectrum of both are same.
3. In both PM and FM, there is phase angle changes take place.
4. In both PM and FM, amplitude of the modulated carrier is same.

Differences

1. Since, $m_p = K_p E_m$, modulation index of PM does not depend on f_m (frequency of modulating signal), Since, $m_f = \frac{\Delta f}{f_c} \frac{1}{f_m}$, modulation index of FM depends upon f_m . (frequency of modulating signal)
2. For given value of f_m , phase angle of PM will differ from phase angle of FM.
3. Increase in f_m does not affect PM, but FM will be affected. There must be some limit for f_m in FM.

3.6 PULSE MODULATION

In analog modulation, some parameter of sinusoidal carrier is varied in accordance with the instantaneous value of the modulating signal.

In Pulse modulation systems, the carrier is no longer a continuous signal but consists of a pulse train, some parameter of which is varied in accordance with the instantaneous value of the modulating signal.

3.7 TYPES OF PULSE MODULATION SYSTEMS

Various types of Pulse modulation techniques are shown in Fig.3.8

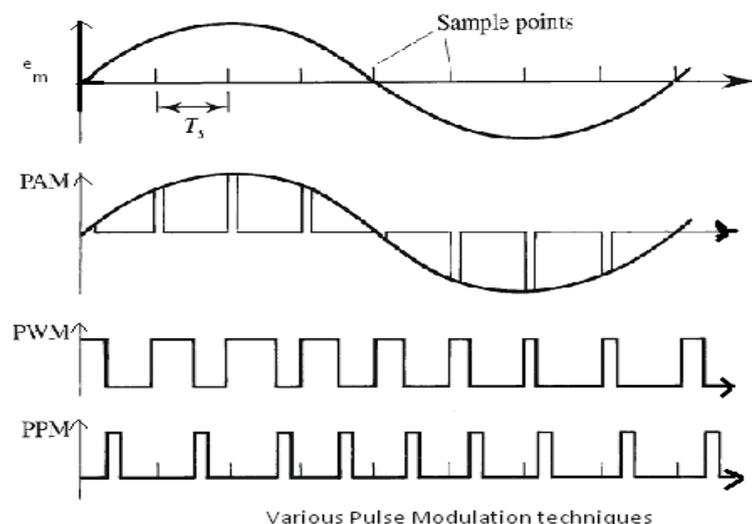


Fig.3.8

Definition

1. PAM (Pulse Amplitude Modulation)

It is a process in which the amplitude of the pulses of a carrier pulse train is varied in accordance with the modulating signal amplitude.

2. PTM (Pulse Time Modulation)

It is a process in which the timing of the pulses of a carrier pulse train is varied in accordance with the amplitude of the modulating signal.

PTM is of 2 types, namely PWM (also called as PDM or PLM) and PPM.

3. PWM (Pulse Width Modulation)

It is also called as Pulse Duration Modulation or Pulse Length Modulation.

It is the process in which the width of the pulses of a carrier pulse train is varied in accordance with the amplitude of the modulating signal

4. PPM (Pulse Position Modulation)

It is the process in which the position of the pulses of a carrier pulse train is varied in accordance with the amplitude of the modulating signal.

5. PCM (Pulse Code Modulation)

The Pulse modulation systems are not completely digital. So to get a complete digital system, the PAM signals are first quantized and then coded (usually in binary code), Thus giving rise to a Pulse Code Modulation (PCM) system.

3.8 Sampling Theorem

The sampling theorem is significant in communication systems because it provides the basis for transmitting analog signals by the use of digital techniques. The sampling theorem may be stated as follows.

Statement of sampling theorem

It states that, if the sampling rate (f_s) in any pulse modulation system exceeds twice the maximum signal frequency (f_m), the original signal can be reconstructed in the receiver, with minimal distortion.

$$\text{i.e., } f_s \geq 2 f_m$$

Where f_s = Sampling frequency

f_m = Highest frequency of the modulating signal.

It is also called Nyquist theorem because it was developed by Henry Nyquist in 1933. It determines the minimum required clock frequency for sampling a continuous time varying wave without distortion. Hence sampling rate is also called Nyquist rate.

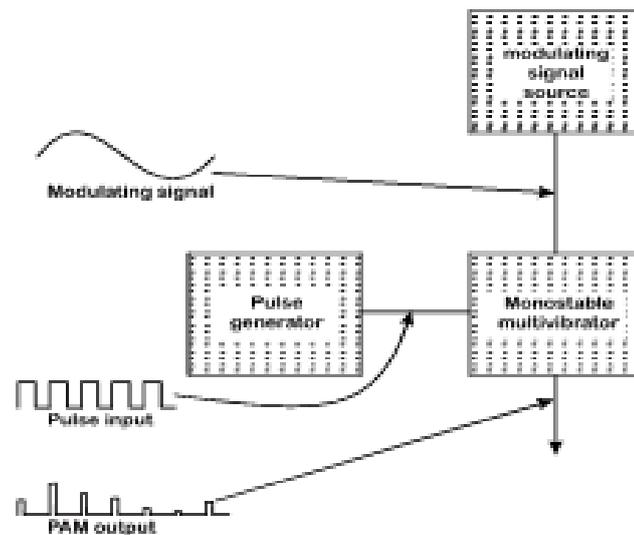
3.9 GENERATION AND DETECTION OF PULSE AMPLITUDE MODULATION (PAM)

PAM is a Pulse Amplitude Modulation system in which the signal is sampled at regular intervals and each sample is made proportional to the amplitude of the signal at the instant of sampling.

PAM MODULATION

A process for generating PAM wave form is illustrated in fig 3.9.

- ❖ The block diagrams of fig 3.9 (a) shows a pulse generator triggering a mono stable multivibrator at an appropriate sampling frequency.
- ❖ The output pulses from the multivibrator are made to increase or decrease in amplitude by the modulating signal.



(a)

- ❖ The modulating signal is superimposed upon the dc supply voltage to the output stage of the multivibrator.

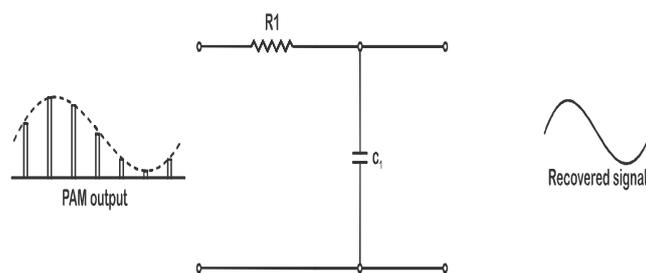


Fig.3.9 (b)

DEMODULATION (detection) OF PAM

Demodulation is accomplished simply by passing the amplitude modulated pulse through a low pass filter. This process is illustrated in fig 3.9(b).

- ❖ The PAM wave consists of the fundamental components which give the pulses their shape.
- ❖ The resistance R1 and Capacitance C1 shown in fig.3.9 form a potential divider.

- ❖ Consequently, low frequency signals (ie., the fundamentals) are passed with very little attenuation. At high frequencies, the impedance of C1 becomes quite small and the signals experience severe attenuation.
- ❖ Thus the filter output is the signal frequency.
- ❖ If necessary, more than one or more filter stage can be employed to sufficiently attenuate the pulse frequency.

3.10 GENERATION AND DETECTION OF PWM (PULSE WIDTH MODULATION)

In Pulse width or Pulse duration modulation, the signal samples must be converted to pulses which have time duration directly proportional to the amplitude of the samples. One method of producing is shown in fig.3.10

PWM MODULATION

- ❖ The system employs a comparator and a Ramp generator (Saw tooth generator) running at the required sampling frequency.
- ❖ The modulating signal is coupled to the non inverting input terminal of the comparator.
- ❖ Therefore, the voltage at that terminal is a dc level with the ac signal superimposed. The inverting input is driven by saw tooth wave from the Ramp generator.
- ❖ The output of the comparator is zero except when the message (modulating) signal exceeds the saw tooth wave.

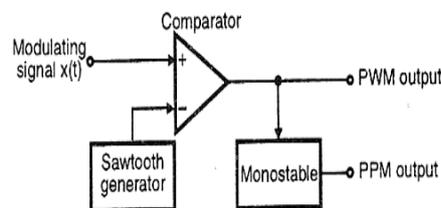


Fig.3.10 (a) PWM and PPM generator

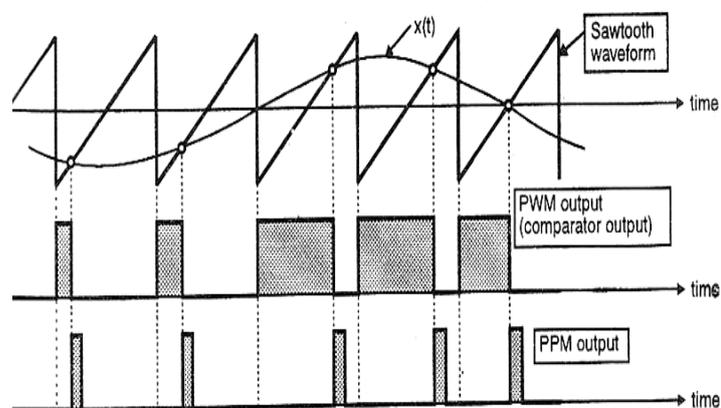


Fig.3.10 (a)

- ❖ It is seen from the fig. 3.10 that the output from the comparator is a series of positive going pulses.
- ❖ Each pulse commences at the instant of ramp wave returns to zero volts and ends when the ramp level coincides with the signal voltage.
- ❖ Thus the comparator produces PWM signal with trailing edge modulation of the pulse duration.

DETECTION OR DEMODULATION OF PWM

- ❖ Demodulation of PWM waves can be accomplished by first converting duration modulated pulse into amplitude modulated pulse.
- ❖ Then filtering can be employed to recover the original modulating signal.
- ❖ The block diagram of PWM demodulator is shown in fig 3.10.

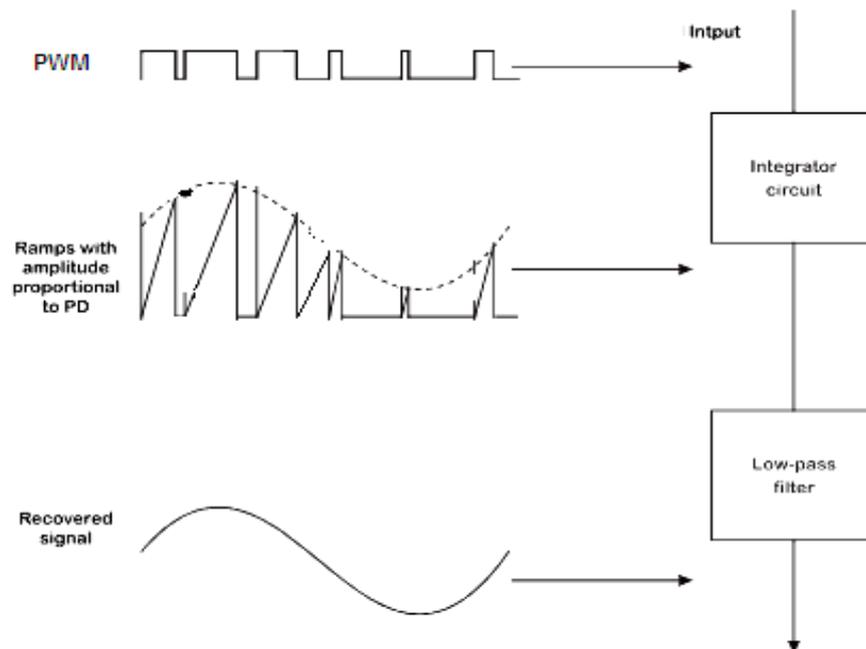


Fig.3.10(b) Detection of PWM wave

- ❖ The PWM wave is applied to an integrator which generates a ramp type output.
- ❖ The ramp commences at the start of each pulse finishes at the end of the pulse.
- ❖ Consequently the ramp and peak value is proportional to the Pulse width.
- ❖ The envelope of the saw tooth wave represents the amplitude of the original modulating signal since it is made proportional to the samples of the original signal.
- ❖ The output of the integrator is now fed to the low pass filter which removes the high frequency components and passes to the low frequency signal (modulating signal).
- ❖ The cut-off frequency of LPF should be equal to the highest frequency of the modulating signal.

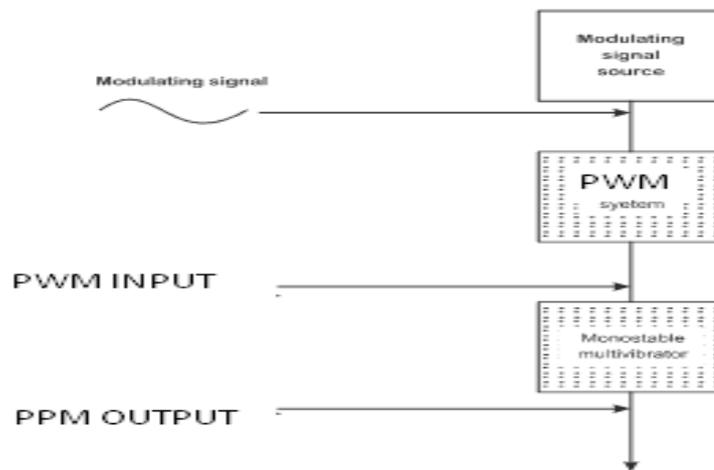
3.11 GENERATION AND DETECTION OF PPM

(PULSE POSITION MODULATION)

The simplest modulation process for Pulse Position

Modulation is a PWM system with the Monostable multivibrator.

- ❖ The monostable multivibrator is arranged so that it is triggered by the trailing edges of the PWM pulses.
- ❖ Thus, the monostable output is a series of constant width, constant amplitude pulses which vary in position according to the original signal amplitude.
- ❖ The block schematic of PPM generator is shown in fig 3.11



Detection (Demodulation) of PPM

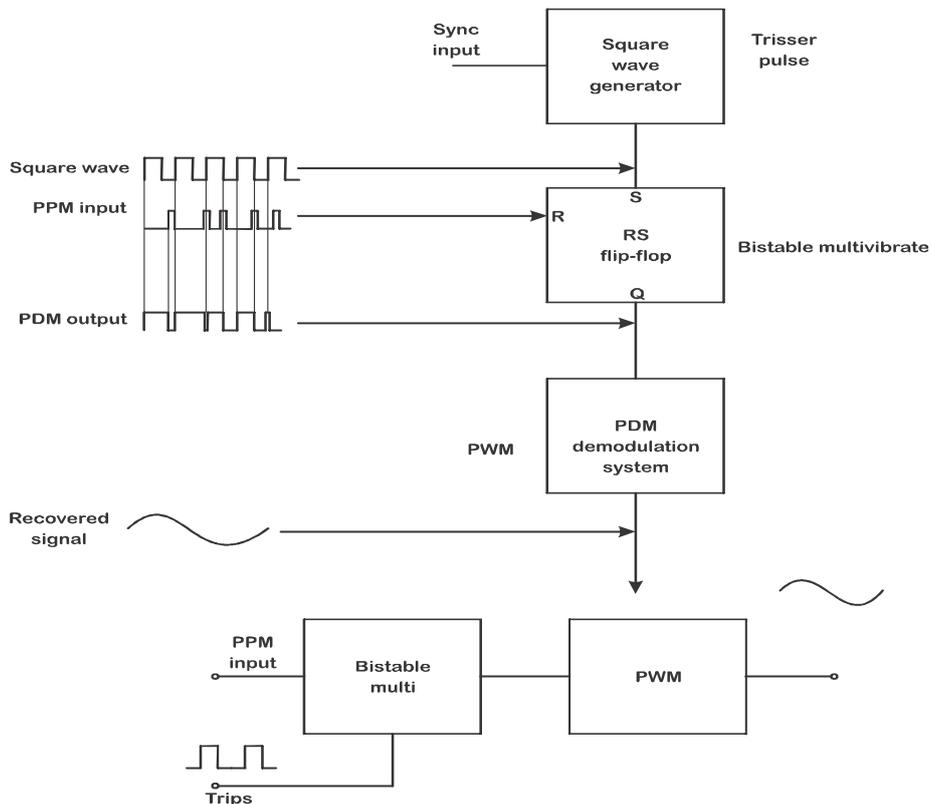


Fig.3.11

- ❖ For demodulation of PPM, a PWM waveform is first constructed by triggering an RS Flip Flop as shown in fig 3.11.
- ❖ The flip-flop is triggered into its set condition by the leading edges of a square wave which must be synchronized with the original sampling source.
- ❖ Synchronization is necessary so that the leading edge of the square wave coincide with the leading edges of the PDM wave that was employed to generate the PPM pulses as shown in fig 3.20.
- ❖ The flip-flop in fig.3.11 is reset by the leading edge of the PPM Pulses.
- ❖ The output of the flip-flop is now a PWM wave which may be demodulated by the process of PWM demodulation.
- ❖ The original signal will be produced from the PWM demodulator.

3.12 PULSE CODE MODULATION

Pulse Code modulation is the process by which an analog signal is converted to digital form in order to be transmitted in digital means.

GENERATION OF PCM (PCM TRANSMITTER)

The block diagram of a PCM transmitter is shown in fig. 3.12 (a)

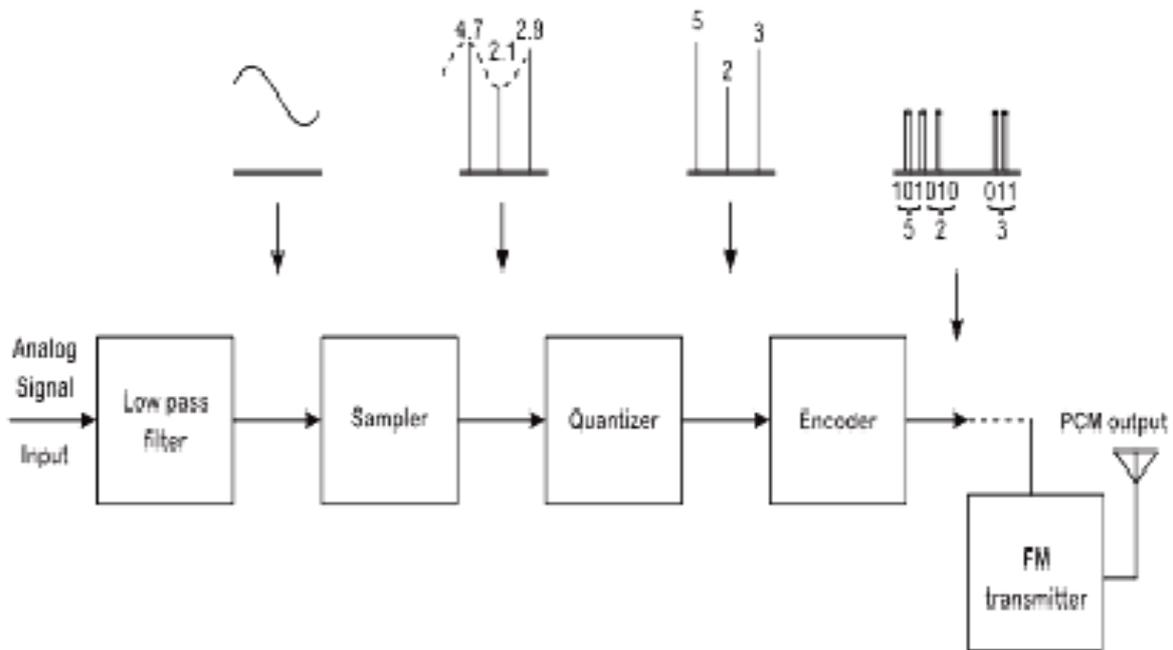


Fig.3.12 (a)

Steps involved in the process of PCM transmission:

- ❖ In order to satisfy Nyquist sampling criteria, the analog signal at the input is band limited by low pass filter.
- ❖ The band limited baseband signal is sampled at Nyquist rate $f_s \geq 2 f_m$ (or) $f_s \geq 2 w$ by the sampler.

- ❖ The sampled pulses are then quantized (Rounding off the levels) in the Quantizer.
- ❖ The encoder encodes these quantized pulses into a predetermined bit pattern before transmission.
- ❖ In essence, the quantizer and encoder perform the fundamental function of analog to digital conversion.
- ❖ Finally the encoded bits are transmitted over the channel.

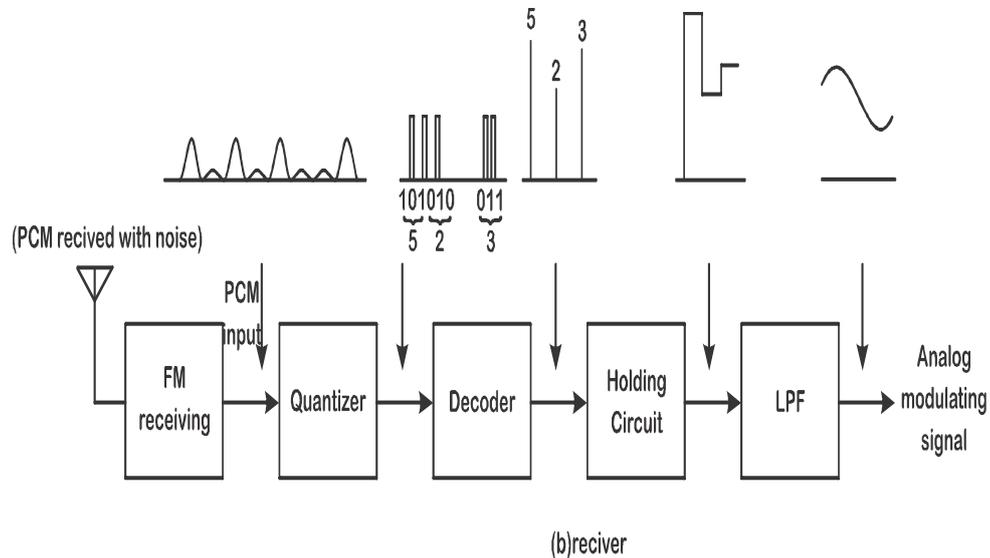


Fig.3.12 (b)

DETECTION OF PCM: (PCM RECEIVER)

The block schematic of PCM Receiver is shown in Fig. 3.12 (b)

- ❖ At the receiving end, the digitally coded analog signals are quantized, decoded and passed through a low pass filter.
- ❖ Thus the original analog signals are reproduced.
- ❖ The basic design concern of PCM systems is to minimize the impact of channel noise at the final stage of the reconstruction process.
- ❖ This is mainly achieved by the Quantizer-circuit at the receiver end.
- ❖ The Quantizer at the receiver is different from Quantizer at the transmitter.
- ❖ Here Quantizer's function is to determine whether incoming digital signal has a voltage level corresponding to binary '0' or binary '1'.
- ❖ If this recognition has been achieved, then the level of external noise has no significance on the reconstruction process.

3.13 QUANTIZATION

In PCM, the message signal is sampled and the amplitude of each sampled signal is rounded off to the nearest one of the standard levels. **The process by which the actual value of the signal is approximated to the nearest standard level is called Quantization.**

In PCM, during the quantization process the error caused due to the difference of actual and nearest standard level introduces some distortion. This is known as Quantization noise.

.14 COMPANDING

- ❖ The most effective method of achieving a better Signal-Quantization noise ratio is that of non linear quantization.
- ❖ In such a scheme, the input analog signal is compressed at the higher amplitude levels and expanded at the lower amplitude levels before quantization.
- ❖ Since the input analog signal is compressed and expanded, the scheme is called companding.

(i.e.) The process of compression at the transmitter and expansion at the receiver is called companding.

3.15 Advantages of PCM over Analog Modulation

1. At low input S/N ratio, the output S/N ratio of PCM system is better than analog system.
2. It is easy to store and time scale PCM signals.
3. PCM systems are more flexible.
4. Modulation and demodulation circuitry in PCM afford more stability and reliability since these are all digital.
5. Repeater spacing is more in long distance communication.

Disadvantages of PCM

1. PCM needs very complex encoding and quantizing
2. PCM requires larger bandwidth compared to analog systems.

3.16 Merits of FM

1. Amplitude of FM wave remains unaffected
2. In FM there is a large decrease in noise and hence increase in S/N ratio.
3. In FM, frequency allocation allows for a guard band. This reduces adjacent channel interference.
4. FM permits use of several independent transmitters on the same frequency with negligible interference.
5. In UHF band, FM operates with less noise.

Demerits of FM

1. A much wider channel, typically 200 KHz is needed.
2. Transmitting and receiving equipments are complex and costly.
3. Reception using conventional method is limited to line of sight.

3.17 Comparison of Analog and Digital Signals

	Analog signal		Digital signal
1	It is a continuous wave	1	It is a discrete wave
2	The value of an analog wave at any given instant can be anywhere within the range of the signal extremes	2	It has only two levels '0' and '1'
3	Represented by discrete signal	3	Represented by continuous time varying signal

3.18 ADVANTAGES OF DIGITAL COMMUNICATION OVER ANALOG COMMUNICATION (AM, FM)

1. It is easier to code and multiplex.
2. It is more efficient
3. Its hardware implementation is flexible.
4. It is possible to introduce new facilities in it
5. Using regenerative repeaters, accumulation of noise will be prevented.

3.19 DIFFERENTIAL PCM (DPCM)

- ❖ In DPCM, the difference between two successive signals rather than the samples themselves are quantized and coded into a PCM pulse train.
- ❖ As a result, the bit rate is considerably reduced and the signal to noise ratio is also improved.
- ❖ The basic reason for this improvement is that in speech signals large variations in the amplitude of one sample from the other is most unlikely and if these differences are quantized, it would require a fewer of bits in the transmission system.
- ❖ The outline of a DPCM modulator is shown in Fig 3.13.

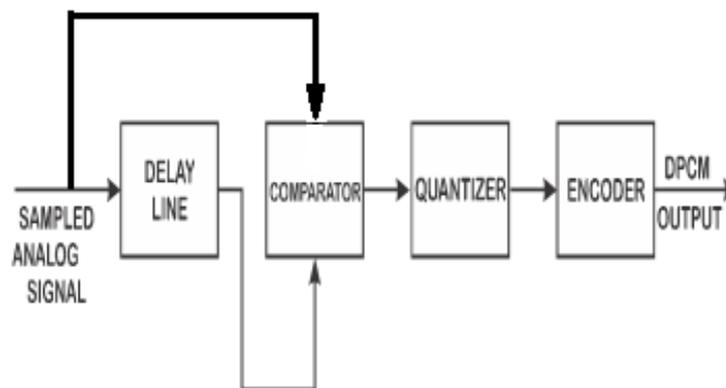


Fig.3.13

- ❖ The signals are sampled in the usual way and these samples are passed through a delay line to a comparator circuit which also gets the sampled signal input.
- ❖ The comparator output corresponds to the difference between the two successive samples. This difference is then quantized and converted into a PCM pulse train in the usual way.
- ❖ The system however is, highly complicated. The technique has been used in experimental T.V. transmission.

Review questions

PART-A

(Two Marks)

1. What is meant by the term Frequency modulation?

Frequency modulation is a system in which the amplitude of the modulated carrier is kept constant, while its frequency and rate of change are varied by the modulating signal amplitude

2. Define the term deviation ratio in FM.

The change of frequency of the carrier wave with respect to the amplitude of the modulating signal is called as frequency deviation and is denoted by ' Δf '

i.e.)
$$\Delta f = K E_m f_c$$

Where, K = constant of proportionality

E_m = amplitude of modulating signal

f_c = frequency of the un modulated carrier.

3. Write the expression for the frequency modulated carrier voltage.

The instantaneous amplitude of a frequency modulated carrier is given by,

$$e_i = E_c \sin (\omega_c t + m_f \sin \omega_m t)$$

Where m_f is called the modulation index for an FM wave

4. Write the expression for the instantaneous frequency of FM wave.

$$f_i = f_c (1 + K E_m \cos \omega_m t)$$

Where,

f_c = frequency of the un modulated carrier

K = constant of proportionality

5. What is meant by 'significant sidebands' in FM?

The side bands having amplitude more than 5% of the carrier are called as significant side bands

6. What are the merits of FM?

1. S/N ratio is good (High)
2. Less adjacent channel interference.
3. Noise can be reduced
4. Transmitting power is constant.

7. What are the limitations of FM?

1. Circuitry is more complex
2. Area of coverage is small (Applicable only to shortwave communication)
3. Huge value will lead to increase in bandwidth.

8. What are the types of Pulse modulation?

1. PAM
2. PWM
3. PPM

9. Compare analog and digital signals.

Analog signal		Digital signal
It is a continuous wave	1	It is a discrete wave
The value of an analog wave at any given instant can be anywhere within the range of the signal extremes	2	It has only two levels '0' and '1'
Represented by discrete signal	3	Represented by continuous time varying signal

10. What is Pulse width Modulation?

It is the process in which the width of the pulses of a carrier pulse train is varied in accordance with the amplitude of the modulating signal

11. What do you mean by Quantization?

The process by which the actual value of the signal is approximated to the nearest standard level is called Quantization.

12. State any two advantages of Digital communication over analog communication.

1. It is easier to code and multiplex.
2. It is more efficient
3. Its hardware implementation is flexible.
4. It is possible to introduce new facilities in it
5. Using regenerative repeaters, accumulation of noise will be prevented.

13. Give the formula for modulation index of in FM

$$\text{ie., } mf = \frac{\Delta f}{f_m}$$

14. Define PPM?

In this type, the position of pulse is changed as a function of the sampled value with reference to a standard position.

15. What are the applications of FM?

1. Stereo sound transmission involves FM
2. Sound signal of TV transmission is in Fm.
3. Multichannel broadband links of satellite communication make use of FM.

16. Define the advantage of pulse modulation?

1. The transmitted power can be concentrated in to short bursts.
2. The time interval between pulses can be filled with sample values from other message.
3. Pulse wave may contain appreciable dc and low frequency content.

PART-B

(Three Marks)

1. State the advantages and Applications of PCM
2. Compare AM and FM.
3. Compare FM and PM.
4. Draw the frequency spectrum of FM.
5. Compare m_a and m_f of AM and FM.
6. Give the expression for FM.
7. Explain the process of Quantization.
8. Represent FM in Time domain.

PART-C

(Ten Marks)

1. With neat sketch explain the block diagram of FM Stereo transmitter.
2. With neat sketch explain the working of FM Stereo Receiver.
3. Describe the process of modulation and demodulation of PAM.
4. Describe the process of modulation and demodulation of PWM.
5. Describe the process of modulation and demodulation PPM.
6. Explain the processes involved in PCM transmission.
7. Explain the working of PCM receiver.
8. Explain the working of Direct FM Transmitter.
9. Explain the working of Indirect FM Transmitter.



UNIT - IV

AUDIO SYSTEMES

4.0 ELECTRO ACOUSTIC TRANSDUCERS PRINCIPLE:

Electro acoustic transducers like microphones and loudspeakers play an important role in communication engineering. An electro acoustic transducer is one which takes energy from an electrical system and delivers energy to an acoustical system or vice versa, the wave form in the second system being substantially equivalent to those in the first system.

The transducer which is capable of converting electrical energy into acoustic energy is known as loudspeaker. On the other hand, the transducer which converts the acoustic energy into electrical energy (Audio) is known as microphone.

4.1 MICROPHONES:

Definition: Microphone is an electro acoustic transducer, which converts a sound wave into a varying electrical current.

The microphones are classified into two types. They are,

- i) Pressure Microphone and
- ii) Pressure Gradient Microphone

i) Pressure Microphone

In pressure microphones, the acoustic pressure acts on only one side of the moving element and the resulting output is proportional to the pressure applied to the moving element.

Example:

- Carbon Microphone
- Condenser Microphone
- Piezo - Electric Microphone and
- Moving Coil Microphone

ii) Pressure Gradient Microphone

In pressure gradient microphones, the acoustic pressure acts on two sides of the moving element and the resulting output is proportional to the difference between the pressure acting on two sides of the moving element.

Example:

1. Velocity Ribbon Microphone.

4.1.1 CARBON MICROPHONE:

Carbon microphone is a small enclosure filled with carbon granules known as carbon button. The button is in constant contact with a thin steel diaphragm through an electrode called plunger. When no sound waves strike the diaphragm, the resistance of the carbon button remains constant.

When sound waves strike the diaphragm, the diaphragm displaces and the plunger attached to it varies the pressure applied to the carbon button. As a result, the resistance of the carbon button changes.

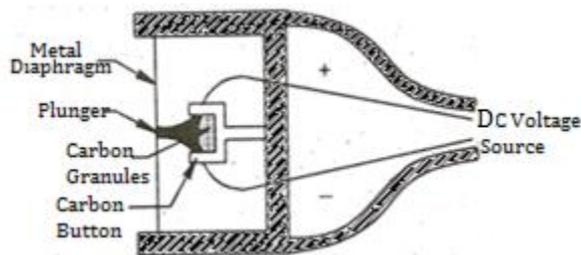


Fig.4.1. (a) Simple Carbon Microphone

The diagram of carbon microphone is shown in the fig.4.1. (a). A dc voltages source is connected to the carbon button. The carbon button is connected in series with a dc voltage source.

Thus the sound waves vary the circuit current in accordance with the variations in resistance. The frequency response characteristics of carbon microphone is shown in the fig.4.1.(b).

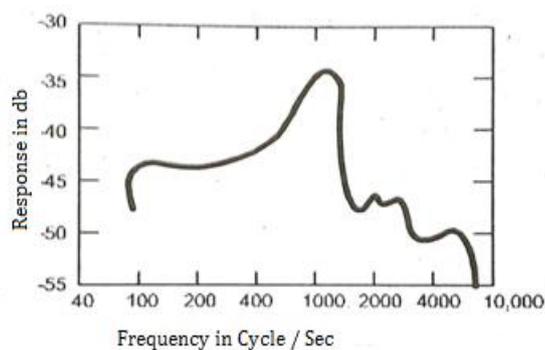


Fig.4.1.(b) Frequency Response Characteristics

Applications:

- It is used in telephones.
- It is used in radio communication.

Advantages:

- Electrical output is high.
- Low cost.
- This microphone is robust (strong).

Disadvantages:

- The frequency response is poor.
- Low fidelity.
- Noisy in operation.

4.1.2 CONDENSER OR CAPACITOR MICROPHONE:

A condenser microphone is one depends for its operation on the variations in capacitance between a fixed plate and a tightly stretched metal diaphragm.

It consists of two very thin plates, one is movable and another one is fixed. These two plates form a capacitor. It is formerly called condenser and hence it is named as condenser microphone. The moveable plate called diaphragm is separated by a small distance from the fixed plate.

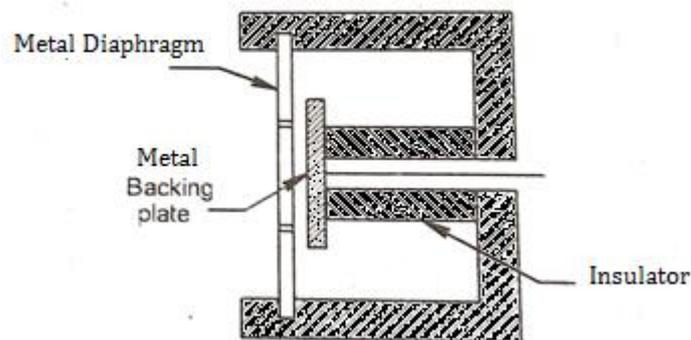


Fig.4.2.(a) Condenser Microphone

A polarizing voltage E_0 is applied between the plates. The condenser microphone is shown in fig.4.2. (a). When a sound wave strikes the diaphragm, it displaces and the distance between the two plates changes, which in turn, changes the capacitance of the microphone. The frequency response characteristic is shown in fig.4.2 (b).

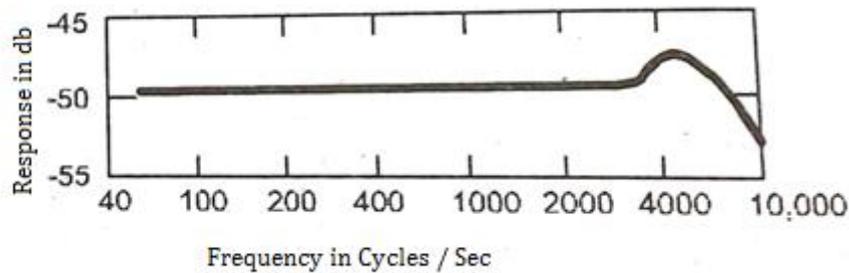


Fig.4.2.(b) Frequency Response characteristics

Application:

- Used as primary standard for calibration purpose.
- Used for high fidelity recording of sound.

Advantages:

- Excellent frequency response.
- Low distortion.
- Extremely small in size.
- High signal to noise ratio.

Disadvantages:

- It requires a built in pre-amplifier, since it has high internal impedance.
- It requires a polarizing voltage ranging from 200 to 400 volts.
- Gives low output.

4.1.3 PIEZO-ELECTRIC MICROPHONE (OR) CRYSTAL MICROPHONE:

A Crystal microphone is based on the principle of 'Piezo electric Effect' which is defined as 'difference of potential between the opposite faces of some crystals is produced when these are subjected to mechanical pressure'.

Crystal microphone is consists of a Bimorph which is nothing but two crystals connected either in series or in parallel form. If the crystals are connected in series, it gives the output result in a form of voltage. The parallel connection gives lower internal impedance.

One end of the bimorph is linked to the centre of the diaphragm by means of a driving pin. The schematic diagram of this microphone is shown in Fig.4.3 (a).

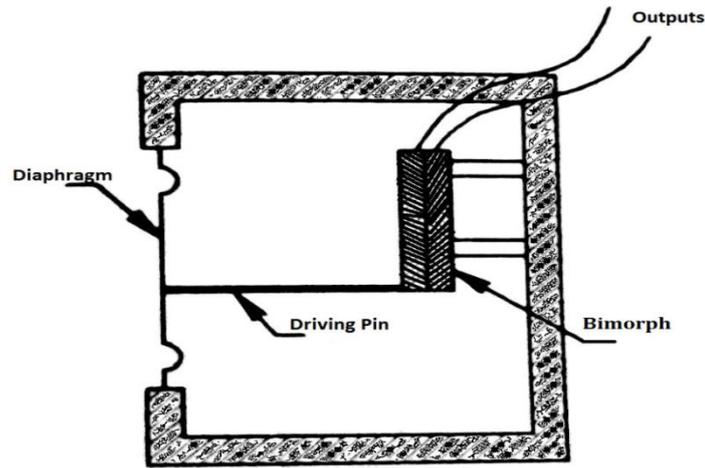


Fig.4.3.(a) Diaphragm Actuated Crystal Microphone

When the sound waves strike the diaphragm, varying pressure is applied to the crystal through the connecting pin, and a varying voltage is produced between the plates. The frequency response characteristics of piezo-electric microphone is shown in Fig.4.3(b).

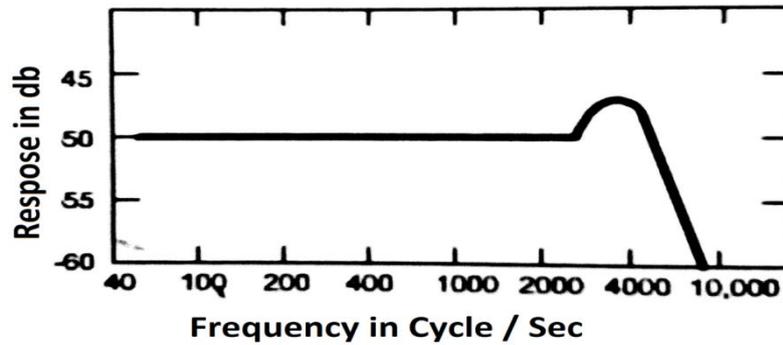


Fig.4.3.(b) Frequency Response Characteristics

Applications:

- Used in public address system.
- Used in hearing aids.
- Used in sound level meters.

Advantages:

- High sensitivity.
- Frequency response is good.
- Low cost.
- Small size.
- It is a non directional microphone.
- Gives large output.
- Polarized source is not needed.

Disadvantages:

- Affected by temperature and humidity.
- Large mechanical impedance of its vibrating elements.
- Not suitable for hot climates as the crystal loses the piezo electric properties.

4.1.4 MOVING COIL (OR) ELECTRO DYNAMIC MICROPHONE

A moving coil microphone is consist of a coil of fine wire is suspended in a strong magnetic fields and is rigidly attached to the back of diaphragm. The diaphragm is corrugated to make it more robust and freely movable. The moving coil or electro dynamic microphone is shown in Fig.4.4(a).

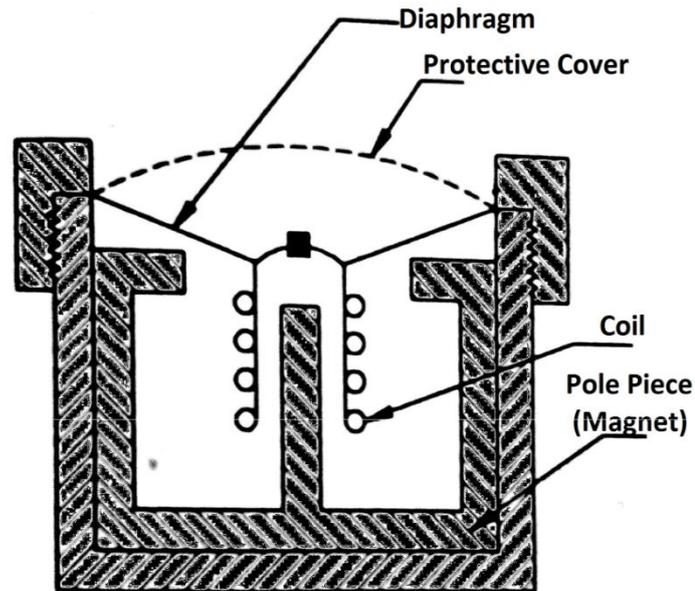


Fig.4.4.(a) A Moving Coil Microphone

When the sound waves strike the diaphragm, it moves back and forth taking the coil with it. The motion of the coil cuts the magnetic lines of force and induces an a.c. voltage in the coil.

The frequency of the a.c voltage is the same as the frequency of the sound waves. The amplitude of this voltage is proportional to the sound waves air pressure. The frequency response characteristics are shown in Fig.4.4 (b).

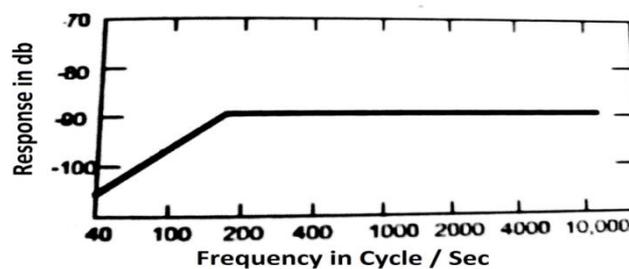


Fig.4.4.(b) Frequency Response Characteristics

Application:

- Used for indoor and outdoor recording system.

Advantages:

- Internal impedance is low.
- Frequency response is constant.
- It requires no external voltage.
- Light in weight.
- Immune to mechanical vibration, temperature and moisture.

Disadvantages:

- Open circuit voltage sensitivity is low.

4.1.5 VELOCITY RIBBON OR PRESSURE GRADIENT MICROPHONE:

Velocity ribbon microphone works on the principle of pressure gradient which implies that driving force acting on a moving element is proportional to the difference in pressures acting on its two sides. The velocity ribbon microphone is shown in Fig.4.5 (a).

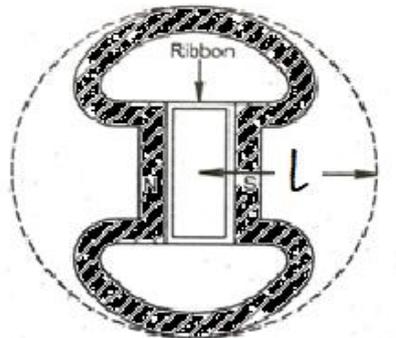


Fig.4.5.(a) Velocity Ribbon Microphone

It consists of a light corrugated metallic ribbon suspended between the magnetic pole pieces n and s, and freely accessible to acoustic pressures on both sides. The resulting driving force is proportional to the difference between the pressure acting on both sides of the diaphragm (Ribbon).

This structure is mounted in a circular baffle of radius l , which in effect determines the length of the air path between the two sides of the ribbon. When a sound wave impinges upon the ribbon, the ribbon moves back and forth in accordance with the velocity of sound.

It cuts the lines of force between the poles of the magnet and a voltage is induced in the ribbon. This voltage is very small, but it can be stepped up by a transformer. The frequency response characteristic is shown in Fig.4.5 (b).

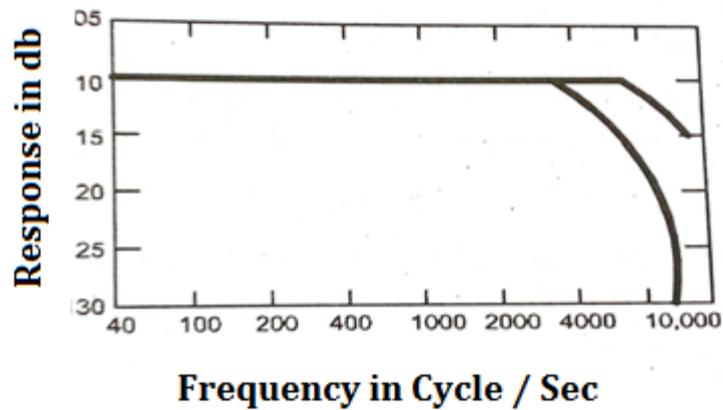


Fig.4.5.(B) Frequency Response Characteristic

The term 'directivity' in microphone indicates how much signal it receives from each direction i.e, 0 to 360°. The bi-directional characteristic of velocity ribbon microphone is shown in Fig.4.5 (c). The bi-directional mike receives maximum signal from 0° and 180°, and no signal from 90° and 270°.

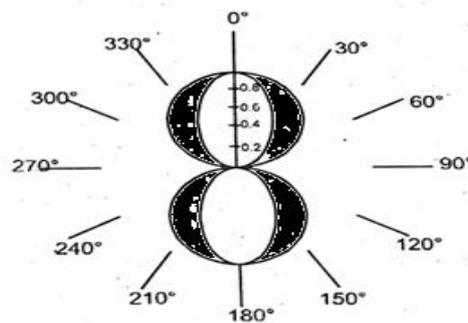


Fig.4.5.(C) Bi-Directional Characteristics of Velocity Ribbon Microphone

Application:

- Used in studio work.

Advantages:

- Frequency response is better than moving coil type.

Disadvantages:

- It requires built-in transformer, since its internal impedance is very low.

4.1.6(a) COMPARISON OF CONDENSER MICROPHONE AND PIEZO ELECTRIC MICROPHONE:

S.NO	CONDENSER MICROPHONE	PIEZO ELECTRIC MICROPHONE
1	Operation depends upon the variation of capacitance.	Operation depends upon piezo electric effect.
2	Mass controlled system.	Stiffness controlled systems.

3	Needs a polarizing source ranging from about 200 to 400 volts	Polarizing source is not needed.
4	It has very high capacitive impedance, and needs a built in pre-amplifier.	It exhibits a high and capacitive impedance and can be directly connected at the input circuit of a JFET or tube amplifier.
5	It gives very low output.	It gives larger output.
6	Extremely small in size and hence used in commercially available magnetic tape recorder.	Smaller in size.
7	Used as primary standard for calibration purpose and high fidelity recording of sound.	Used in public address system, hearing aids system and sound level meters.

4.1.6(b) COMPARISON OF CARBON MICROPHONE AND CONDENSER MICROPHONE :

S.NO	CARBON MICROPHONE	CONDENSER MICROPHONE
1	Operation depends on the variation in resistance of a carbon button.	Operation depends on variation of the capacitance of a capacitor.
2	Stiffness controlled system.	Mass controlled system.
3	High electrical output.	High internal impedance and low output hence pre- Amplifier is needed.
4	It requires any polarized voltage.	It requires polarizing voltage ranging from about 200 to 400v.
5	Used for telephone and radio communication purpose.	Used for high fidelity recording of sound and used as primary standard for calibration purpose.

4.2.0 LOUDSPEAKER:

Definition: Loudspeaker is an electro acoustic transducer, which converts electrical signal into sound signal.

CHARACTERISTICS OF AN IDEAL LOUDSPEAKER:

1. Electro acoustic efficiency approaches 100 percent.
2. Acoustic output is independent of frequency over the entire audible range.
3. Harmonic and inter modulation distortion are not present in the output.
4. Input signal are faithfully reproduced.
5. It is capable of producing a non direction radiation pattern.
6. It must be of small size having the required acoustic output.

TYPES OF LOUDSPEAKER:

1. Dynamic cone type moving coil loudspeaker.
2. Horn type moving coil loudspeaker.
3. Electrostatic type loudspeaker.

4.2.1 DYNAMIC CONE TYPE MOVING COIL (OR) DIRECT RADIATOR (OR) ELECTRODYNAMICS LOUDSPEAKER:

It works on a principle that when a current carrying conductor is placed in a magnetic field, force is induced in the conductor.

Construction:

The constructional details of direct radiator speaker or dynamic loud speaker or moving coil loud speaker is shown in Fig.4.6(a).

Permanent Magnet:

The magnetic structure is made up of a soft iron frame which forms pole piece and a strong permanent magnet. It is used to produce high flux density across the annular air gap.

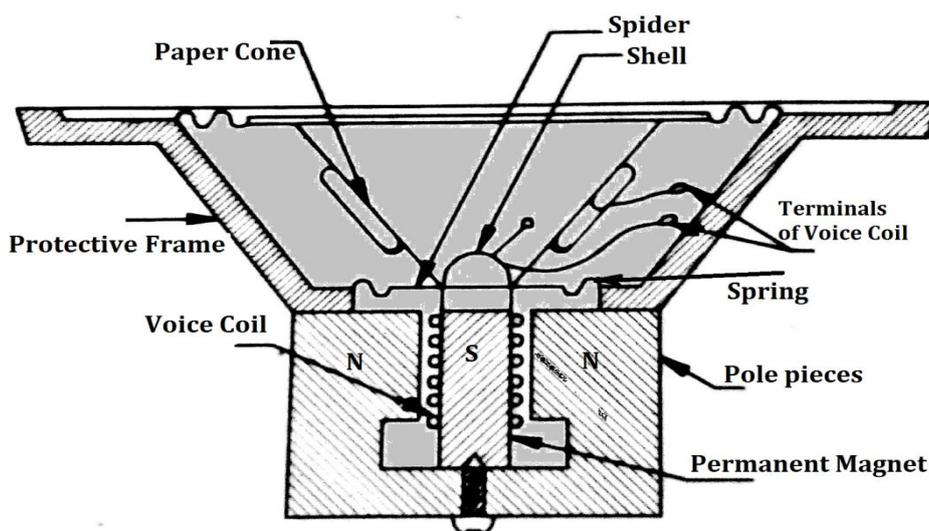


Fig.4.6.(a) Electromagnetic Loud Speaker

Voice coil (or) speech coil:

This coil is wound on a cylindrical form held in the annular air gap of the magnetic structure.

Spider:

The spring action of the corrugated spider mounting keep the voice coil centered.

Diaphragm (cone):

It is a paper cone attached to the voice coil which moves alternatively, compresses and expands the air in front of the speaker. The corrugations give reasonable flexibility of movement. Similarly the coil end of the cone is positioned by a corrugated paper commonly known as spider, so that the coil can move freely in the gap.

Baffle:

The simplest form of baffle is a large flat board with a hole at the centre through which cone can operate.

Dust cap:

It prevents the dust entering into the air gap between the voice coil and magnet.

Operation:

Voice frequency electrical current passes through the voice coil. When a positive current is passed through the coil, a force is generated between the coil and the permanent magnet and this moves the coil axially out of the pole structure. It is restrained lightly by the spring action of the corrugated spider mounting which keeps the coils centered. When the current reverses, the direction of the magnetic force also reverses and the coil is pulled back into the pole structure.

When the cone moves forward, it compresses the air in front of it with a corresponding rarefaction at its back. When the cone moves backward, the reverse process happens. If there is nothing else present except the cone, the compressed air in front of the cone can move around the edge of the cone to the rear where the pressure is less. This causes the air pressure in front of the cone to return to normal. Under this circumstance, the sound output produced is considerably reduced. The reduction in sound output is very large at low frequency because the time interval between successive compressions and rarefaction is then large.

This effect can be overcome by the use of some form of a baffle. Sound waves from the front of cone are prevented by the baffle from moving to the back. The use of baffle therefore improves radiation at low frequencies. Larger the area of the baffle, the better is the low frequency radiation.

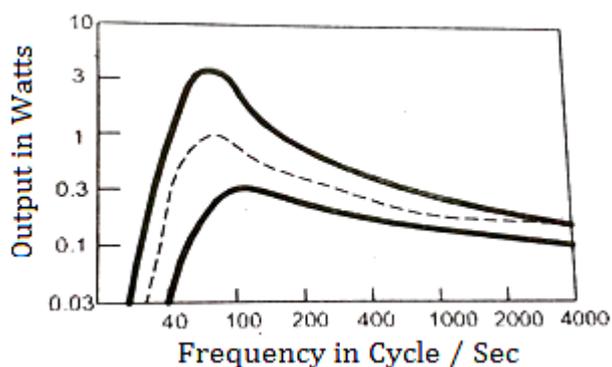


Fig.4.6.(b) Frequency Response Characteristics

The frequency response curve is shown in Fig. 4.6 (b). The ac load impedance of the voice coil measured at the terminals is a function of the magnetic field force, coil size and efficiency of the acoustical coupling of cone.

Typical values are low, ranging from 1Ω to 300Ω with 4, 8 and 16Ω speakers are commonly used. Power rating ranges from few mw to several hundred watts. Lower power rating speakers are used in portable receivers and the higher ones are used in auditorium of outdoor.

Application:

- Used in all audio systems

Advantages:

- Compact in size and low cost.
- Better response in audio range.

Disadvantages:

- Low efficiency in low frequency.
- Narrow directivity pattern.
- Low power handling capacity.

4.2.2 HORN TYPE LOUDSPEAKER:

Sound output, particularly at low frequency, can be increased by attaching properly shaped horn to a small piston line sound source. The cone of the speaker is used to improve the efficiency of coupling between the coil motion and the surrounding air. Higher efficiency is obtained when the output from speaker unit is fed into the throat of an acoustical horn. The horn acts as an impedance transformer and also as a radiator. Fig.4.7 shows the structure of a horn type loudspeaker.

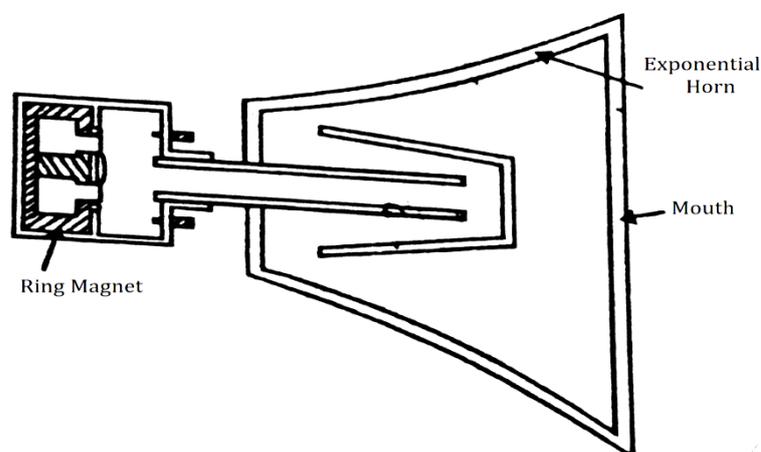


Fig.4.7. Horn Type Loud Speaker

Construction:

In the annular air gap of a heavy permanent pot magnet a moving coil made up of a former one to which a thin aluminum ribbon is wound. Edge wise is made to have its movement. This driver unit is used with a horn baffle. The driver unit is the same type of structure as the cone type speaker except the cone is omitted. The horn structure is filled back on itself to conserve physical space because an unfolded horn might be as much as 2m in length and 1m across the mouth.

Working:

Horn act as an impedance matching device connecting the diaphragm of high impedance with air of low impedance. An exponential horn acts as a high pass filter. Higher efficiency is obtained when the output from the speaker is fed to the throat of a horn.

The good reproduction requires a long horn which has a smaller taper as well as a larger area. These units have power handling capability of more than 100watts operating at an input impedance of 8 to 16 ohm. The horn loaded loud speakers have a far greater efficiency than cone type loud speaker.

Application:

- Announcement system.

4.2.3 DIFFERENCE BETWEEN CONE AND HORN TYPE LOUDSPEAKER:

S.NO	CONE TYPE LOUD SPEAKER	HORN TYPE LOUD SPEAKER
1	Power handling capacity ranges from few milli watts to several 100 watts.	Power handling capacity is more than 100 watts.
2	At low frequency cone type loud speaker is less efficient.	At low frequency horn type speaker is more efficient.
3	It is not so.	The horn works as an acoustic transformer and transfer low amplitude pressure vibration in the region of the diaphragm into vibration of high amplitude at low pressure.
4	Diaphragm is lager in size.	Diaphragm is smaller in size.
5	Sound effect of this speaker is less.	Sound effect of this speaker is very high and efficient.
6	It requires an external impedance matching circuit.	The system itself is an impedance matching circuit.
7	More than one loudspeaker is required to cover different octaves of the sound spectrum.	One loudspeaker works for many octaves of the sound spectrum.

4.2.4 ELECTRO -STATIC LOUDSPEAKER:

An electrostatic loud speaker is a loudspeaker design in which sound is generated by the force exerted on a membrane suspended in an electrostatic field.

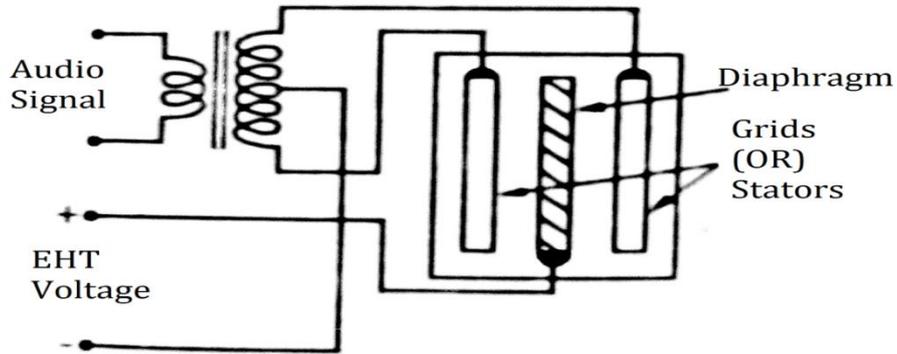


Fig.4.8.ELECTRO STATIC LOUDSPEAKER

Construction:

Fig 4.8 shows the construction and connections of an electrostatic loudspeaker. The electrostatic loud speaker consists of a thin flat diaphragm and two conductive grids (or) stators. The diaphragm is sandwiched between two electrically conductive grids with a small air gap between the diaphragm and grids. The diaphragm is usually made from a polyester film (thickness 2 to 20 μm) with exceptional mechanical properties. The grids are nothing but perforated metal sheets.

Working:

By means of the conductive coating and an external high voltage supply the diaphragm is held at a DC potential of several kilovolts with respect to grids. The grids are driven by the audio signal. The front and rear grid are driven in antiphase. As a result a uniform electrostatic field proportional to the audio signal is produced between both grids. This causes a force to be exerted on the charged diaphragm, and its resulting movement drives the air on either side of it.

It is a high impedance device. Therefore impedance matching is necessary in order to use a normal amplifier. Most often a transformer is used. It must provide a constant transformation ratio over the entire audio frequency range and so avoid distortion.

Advantages:

- Extremely light weight of the diaphragm and high frequency response.
- More accurate stereo reproduction of recordings.

Disadvantages:

- Lack of bass response.
- Sensitivity to ambient humidity level.

4.2.5 HIGH FIDELITY (HI-FI) ELECTROSTATIC SPEAKER:

A high fidelity sound is that it is a reproduced sound with high degree of similarity to that of the original or live sound that was travelled from a source and has undergone several conversions through a system or several systems.

Hi fidelity is said to be achieved when the sound that is reproduced has negligible distortion from the original, where it has little external noise, and when the volume levels and room acoustical effects are pleasing to hear. This reproduced sound might even be more pleasing to the listener at the output of the system than the original live sound would have been, if heard at its source.

4.2.5 (A) WOOFER, MIDRANGE AND TWEETER:

Fidelity of a receiver is its ability to reproduce different frequency components. Single speaker cannot reproduce all frequency components correctly.

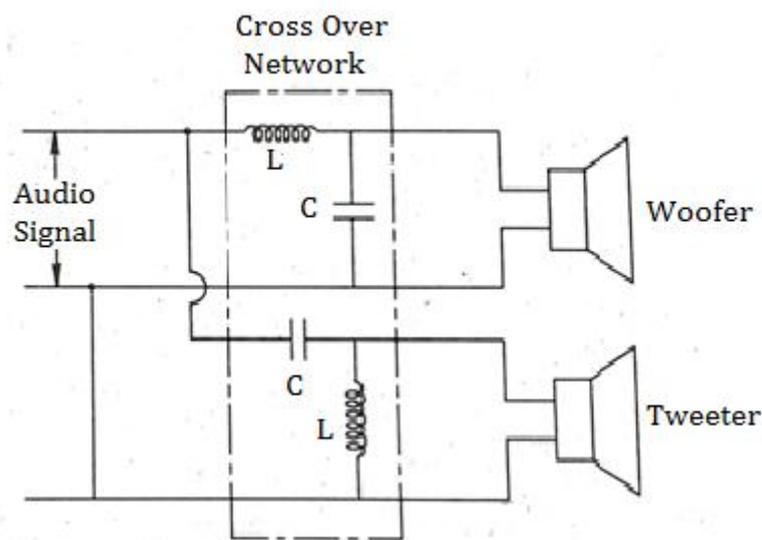


Fig.4.9.Connection of Hi-Fi Speakers (or) Two Way Cross Over Network

Woofer:

- To reproduce low frequency signal, cone diameter should be large is called Woofer.

Tweeter:

- To reproduce high frequency signal, cone diameter should be small is called tweeter.

Driver or Squawker:

- To reproduce the mid frequencies between 300Hz to 5000Hz, the driver or squawker loudspeakers are lying between the tweeter and woofer.

Fig 4.9 shows the connection of woofer and tweeter loudspeakers. It is a two way cross over network.

Frequency Response Characteristics:

The frequency response of woofer, tweeter and squawker is shown in Fig.4.1

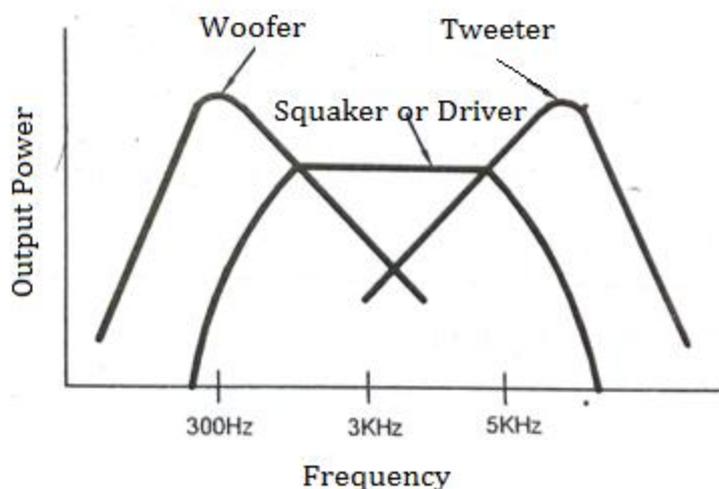


Fig.4.10.Frequency Response Characteristics

4.2.5. (B) SPECIFICATION AND RANGE OF WOOFERS AND TWEETERS:

Loud speakers are available in circular, oval and hexagonal shapes. The speakers are specified with

- a. Diameter of the speaker frame.
- b. Type.
- c. Wattage rating.
- d. Voice coil impedance.

S.NO	CHARACTERISTICS	WOOFER	TWEETERS
1	Frequency Range	Low Frequency (0-300)Hz	High Frequency above 5000Hz
2	Diameter of cone used	Large	Small
3	Type Generally used	Dynamic	Electrostatic
4	Wattage Rating	High	Low
5	Voice Coil Impedance	Low	High
6	Cone Depth	More	Less
7	Cone Type	Stiff Cone and Heavy Frame	Hard Cone

4.2.6 CROSS OVER NETWORK:

To improve the quality and performance of Amplifiers and radio receivers, various control circuits are used. These circuits are called cross over network. Cross over networks are used in multi-way speaker systems to split the input signal into separate frequency bands. The cross over network for a three way system is shown in Fig. 4.11.

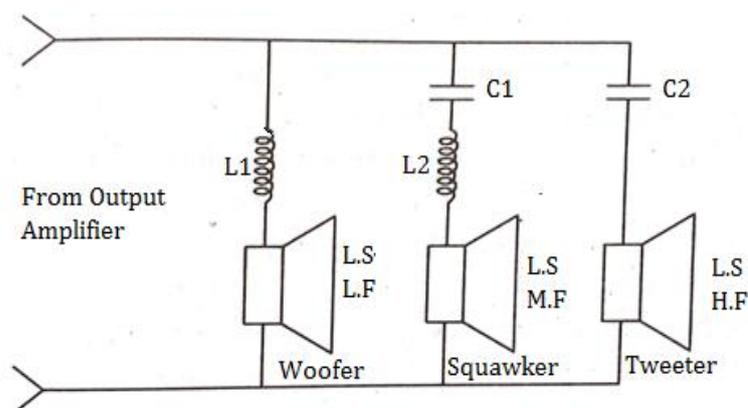


Fig .4.11.Three Way Cross over Network

The mid range of frequency forms a band between the low pass and high pass frequencies. The low pass filter is formed by an inductor L_1 , and the high pass filter is formed by a capacitor C_2 . The band pass filter is formed by an inductor L_2 and capacitor C_1 .

The capacitor, C_2 connected in series with tweeter prevents low and mid frequencies from reaching the tweeter. Similarly inductance L_1 in series with woofer prevents high frequencies from reaching the woofer. Inductance L_2 and capacitor C_1 placed in series with squawker circuit prevents low and too high frequencies from reaching the squawker.

4.2.7 SURROUNDS SOUND SYSTEMS:

Surround sound, using Multichannel audio, encompasses a range of techniques for enriching (expanding and deepening) the quality of sound reproduction, of a recorded source. In this system the additional recorded sound channels are reproduced by using additional discrete speakers. The three dimensional (3D) Sphere of human hearing can be virtually achieved with audio channels above and below the listener.

Surround sound technology is used in both cinema a home theater system, video game consoles and PCs.

1. Creating Surround Sound:

Surround sound is created in several ways. The first and simplest method is using a surround sound recording microphone technique and/or mixing-in surround sound for playback on an audio system using speakers. In this system make to play audio from different

directions for the listeners. The second method, is processing the audio with psychoacoustic sound localization method to simulate a two dimensional (2D) sound field with headphones. A third approach based on Huygens principle, to reconstruct the record sound field wave fronts within the listening space of an 'audio hologram' form.

2. Mapping Channels to Speakers:

Surround sound systems rely on the mapping of each source channel to its own loud speakers. Matrix systems recover the number and content of the source channels and apply them to their respective loudspeakers. In discrete surround system, the transmission medium allows for (at least) the same of channels of source and destination. The one-to-one, channel-to-speaker, mapping is not the only way of transmitting surround sound signals.

3. Bass Management:

Surround sound systems may make use of bass management, the fundamental principle of which is that bass content in the incoming signal, irrespective of channel, should be directed only to loudspeakers capable of handling it.

4. Surround Sound Specification:

Each specification description represents the number of channels, not the number of speakers. The different channel surround systems are explained below.

5. Low Frequency Effects (LFE) Channel:

It was originally developed to carry extremely low 'sub – bass' cinematic sound effects on their own channel. In the original movie theater implementation, the LFE was a separate channel fed to one or more subwoofers.

Types:

1.3.0 Channel Surround (Analog multiplexed , Dolby Surround):

In this system 3 audio channels are extracted from a specially encoded two channel source. Fig.4.12 (a).

- Two channels for the speakers at the front – left (L) and right (R)
- One channel for surround speaker or speaker at the rear – surround (S)

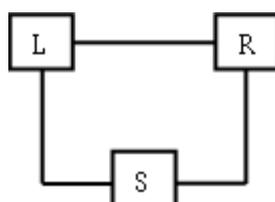


Fig.4.12. (a)

Placement (3 Speakers in Total):

Three identical speakers placed equidistant around a central listening position. If two rear speakers are used they should also be placed above ear height, slightly behind the listening position.

2.4.0 Channel Surround (Analog Multiplexed / Discrete, Quadraphonic) :

In this system 4 audio channels are extracted from either a specially encoded two - channel source or a 4 - channel source. Fig.4.12(b).

- Two channels for speakers at the front – left (L) and right (R)
- Two channels for surround speakers at the rear – surround right (RS) and surround left (LS)
- Describes the early matrixes system and discrete quadraphonic surround systems.

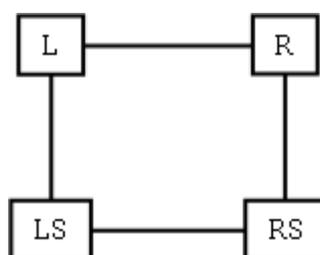


Fig.4.12.(b)

Placement:

This system is used for music only. All speakers should be at $\pm 45^\circ$. All speakers should be placed above ear height.

1. 5.1 Channel Surround (3-2 Stereo) (Analog Matrixed , Dolby Pro Logic – II):

In this system five audio channels and one LFE channel are extracted from a specially encoded two-channel or a stereo source. Fig.4.12(c).

5.1 Surround sound may be referred to as 3-2 stereo. the term 3-2 refers to 3 front speakers and 2 rear speakers.

- Two channels speakers at the front – left (L) and right (R)
- One channel for speaker at the centre - centre (C)
- Two channels for surround speakers at the rear – surround left (LS) and surround right (RS)
- One low frequency effects channel (LFE).
- Describes the Dolby Pro Logic – II matrixed surround system.

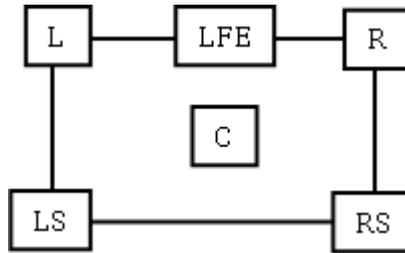


Fig.4.12.(c)

Placement:

The left and right speakers are located at $\pm 30^\circ$. The rear speakers should be positioned approximately $\pm 110^\circ$.

2. 6.1 Channel Surround :

In this system 6 audio channels and one LFE channel are extracted from either a specially encoded two channels or stereo source. Fig.4.12(d).

- Two channels for speakers at the front – left (L) and right (R)
- One channel for speaker at the centre – centre (C) side left (LS) and side right (RS)
- Two channels for surround speakers at the side – side left (LS) and side right (RS)
- One channel for surround speakers at the rear – back surround channel (BS)
- One low frequency channel to drive a sub – woofer (SW)

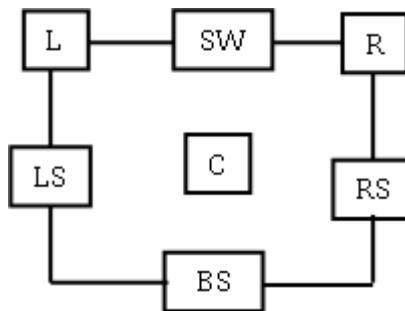


Fig.4.12.(d)

3. 7.1 Channel Surround :

In this system seven audio channel are extracted from an 8 channel source. This system is used in home entertainment system. Fig.4.12(e).

- Two channels for speakers at the front – left (L) and right (R)
- One channel for speaker at the centre – centre (C)
- Two channels for surround speakers at the sides – left surround (LS) and right surround (RS)
- One low frequency effects (LFE) channel.

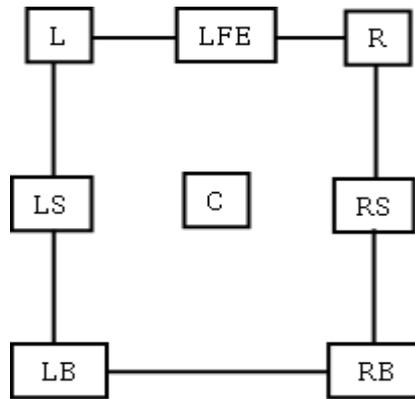


Fig.4.12.(e)

4. 10.2 Channel Surround :

10.2 Channel Surround contains 14 discrete channels.

- Five front speakers – left wide, left, centre, right and
- Five surround channels – left surround diffuse, left surround direct, back surround, right surround diffuse and right surround direct.
- Two LFE channels – LFE left and LFE right
- Two height channel – left height and right height.

4.3 AUDIO RECCORDING AND REPRODUCTION:

A microphone converts sound signal into electrical signals called audio signals. There are two methods for recording audio signal,

- (1) Recording on tape, and
- (2) Recording on disc.

Recording on tape is done in the form of tiny magnets and that on disc is done in the form of tiny pits. Reproduction of signals from the magnetized tape is done by using the principle of electromagnetic induction. Reproduction from the disc is done either by sensing capacitance between the stylus and the pits or by making use of light. Light from the pits is diffracted light is taken as '1' and the reflected light is taken as '0'.

4.3.1. COMPACT DISC (CD) SYSTEM:

Compact disc (CD) stores wide variety of data. The signals stored in compact disc are in high density digital format. Hence the signals recorded are accurate representation of the original audio signal. Compact disc is also used to store like text, picture image, audio, video, software etc. In earlier days, the compact disc was used to store audio tracks. The standard CD is 12 cm or 5 inch in diameter. It has three layers,

1. Transparent Substrate:

This layer is made up of a poly carbonate wafer [plastic disc].

2. Thin metallic Layer:

The wafer base is coated with a thin metallic layer of aluminium alloy. This metallic layer contains a long sequence of pits of about 0.5 micrometer diameter. The reflective aluminium film portion of the disc is usually read by the CD drive.

3. Outer Layer of Protective Acrylic:

The aluminium film is further covered by a plastic coating called protective lacquer layer that protects the underlying data. The layout of CD is shown in the Fig.4.13.

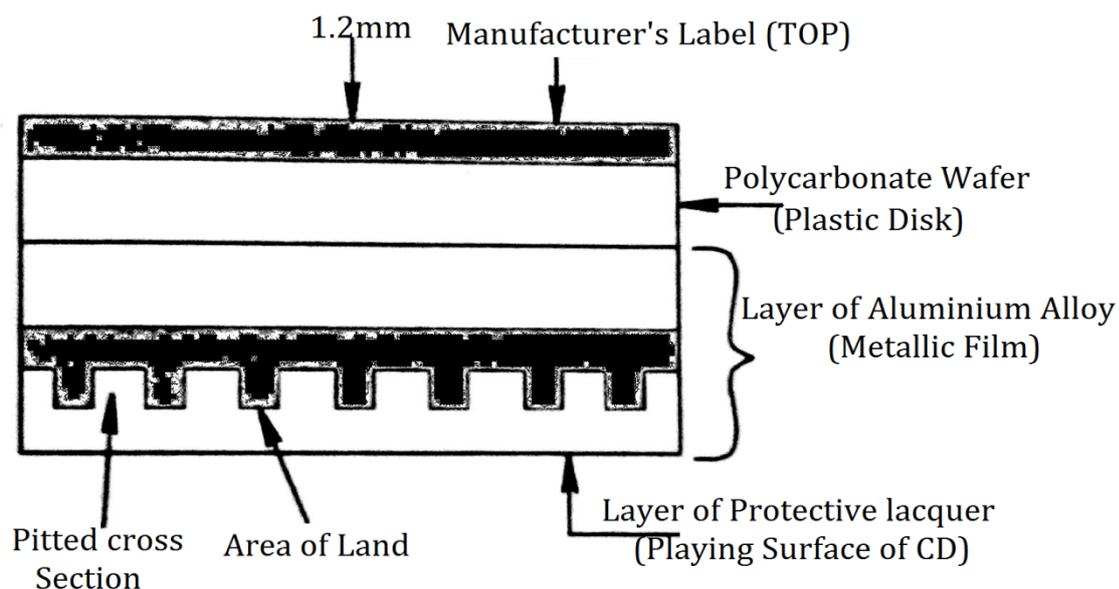


Fig.4.13.Layout CD-ROM Disc

When CD is manufactured, a disc of glass is coated with photo resist id sensitive to Laser light. During recording, the Laser light expose a fine helical track on the photo resist material. The width of the track is normally $0.5\mu\text{m}$. The 1's in the digital data are recorded onto the photo-resist surface by switching the Laser beam ON and OFF alternately as each 1 occurs.

No 0's are recorded onto the disc they are regenerated within the CD player. While recording 1's the photo-resist surface of the disc is exposed and a series of pits and lands are created. Thus while recording the digital data '1', pits are produced on the surface of the CD disc.

Similarly while recording the digital data 0, there is no change on its surface. So the pits are having the data of 1 and the other places are having the data 0. The length of the pit or land depends on the period or distance between 1's.

1. CD RECORDING :

The signal to be recorded on CD is first amplified and then converted into digital signal by using sample and hold circuit, and ADC. The ADC output is also applied to Laser beam generator. The signal from the crystal oscillator and Laser beam generator are applied to the control circuit and for controlling the servo system.

The servo system controls the disc rotation through motor, also controls the track and focus of the Laser beam generator. The block diagram of CD recording system is shown in the figure 4.14.(a).

After recording, the unexposed photo-resist material is chemically removed, thereby producing a helical track across the surface of the glass disc. This becomes the glass master to produce large volumes of CD's.

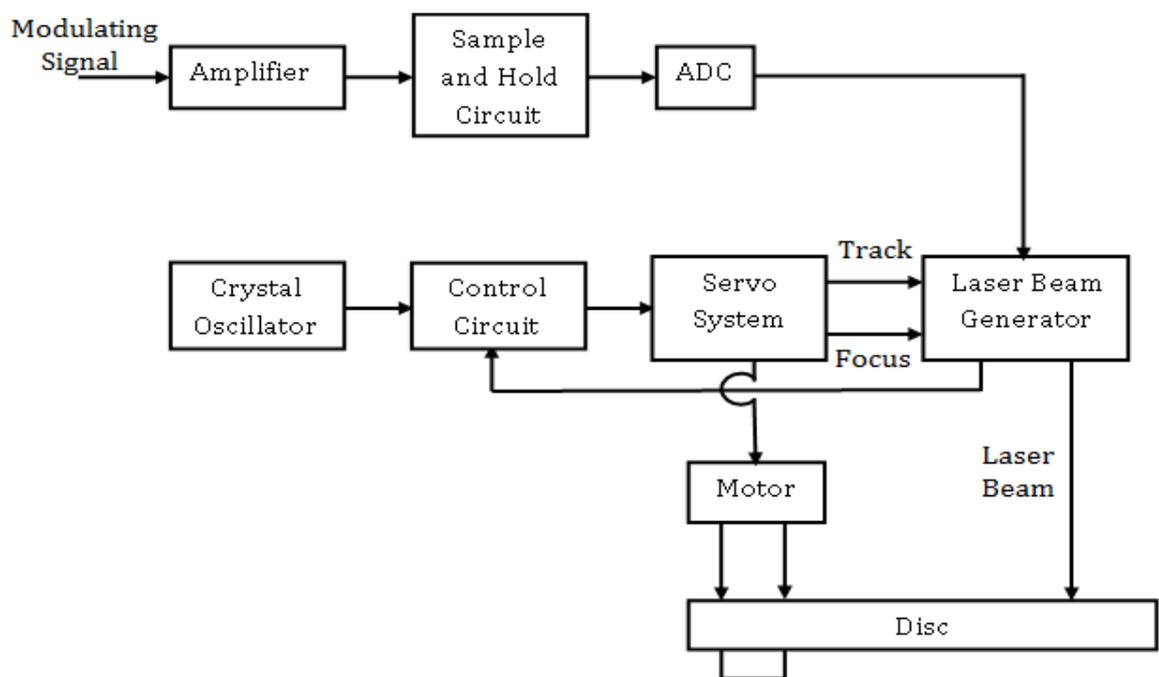


Fig.4.14.(a) Block Diagram of CD Recording

2. CD REPRODUCTION :

The data retrieval system basically consists of the following stages. They are

1. A servo mechanism- It rotates the CD
2. A laser head - It will move radially. The laser head can emit and detect laser beam of 70nm wave length.

When the disc rotates, the laser beam is focused onto the playing surface, where the laser is reflected by the 'lands' and scattered by the 'pits' i.e., whenever

there is a pit-to-land or land-to-pit change, there is a corresponding change in the amount of light reflected. Thus laser beam detect the pit boundaries.

A pit boundary represents a '1' and its absence represents a '0'. Strong reflection of the light indicates no pit, i.e., land. A receiver of light receptor detects where the light is strongly reflected or where it is absent or diffused. Absent or diffused light reflection is caused by the recorded pits in the CD. The block diagram of CD reproduction system is shown in the fig 4.14.(b).

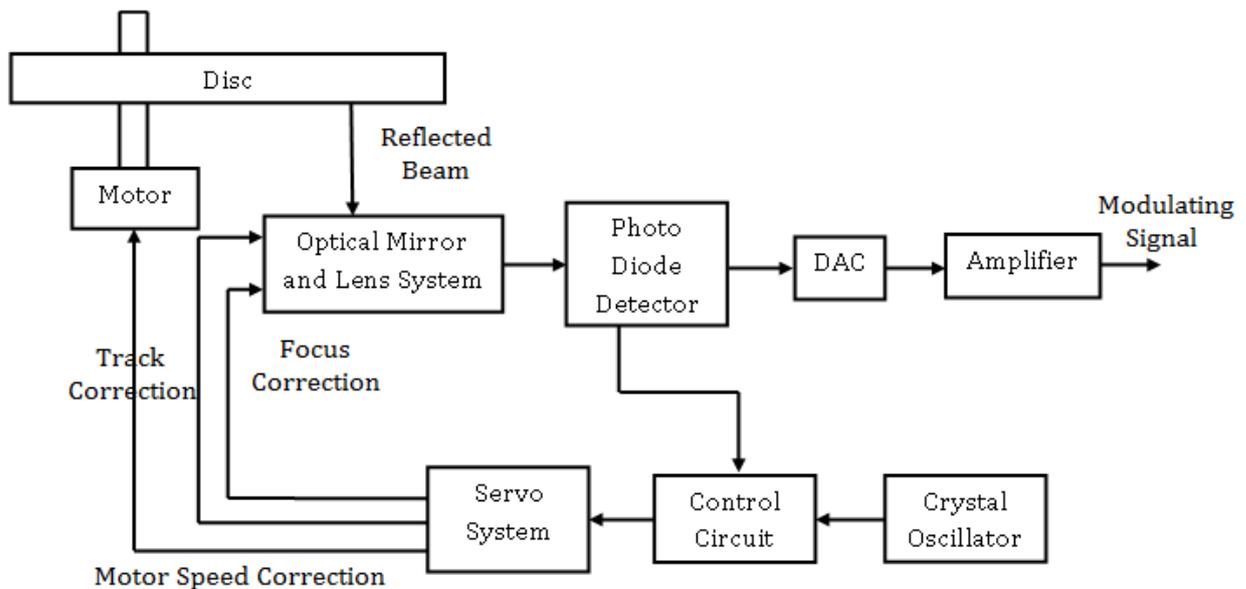


Fig.4.14.(b) Block Diagram of CD Reproduction System

The clock signal obtained from the disc is compared with a crystal oscillator signal. The output of the control circuit is used to control the servo system. The servo system controls the motor speed and also the track correction and focus corrections of optical mirror and lens system.

The reflected beam from the disc is applied to the photo detector. It converts the light variations into digital signal. The digital output is then converted in to analog signal by using DAC and properly amplified. Thus we can get the original modulating signal.

The clock signal coming from the disc and the signal coming from the crystal oscillator are applied to the control circuit. By using the two signals the control circuit controls the servo system. The speed of the motor is controlled by the servo system. The servo system also controls the optical mirror and lens system and also the focusing of the rays.

CD data encoding uses two methods. They are

- CIRC – Cross -Interleave Read – Solomon Code.

- EFM – Eight to Fourteen Modulations.

Advantages of Compact Disc:

1. Error detection in the disc is possible using certain codes.
2. The surface of CD is not affected by dust, grease and scratches because it is covered by the transparent plastic.
3. Reproduced sound has good quality with less background noise.
4. High signal to noise ratio.
5. Stereo Effect is preserved well.
6. High channel separation.
7. Small size.
8. Excellent frequency range.
9. Low distortion.
10. Does not exist flutter.
11. Program selection is possible in a compact disc.

4.3.2. MP3 SYSTEM:

MP3 is a compression system for music. This compression method helps to reduce the space required to store a song on a CD. The goal of the MP3 format is to compress a CD quality song by a factor 10 to 14, without losing the CD sound quality. This allows one MP3 CD to store 100 to 150 songs compared to 10 to 15 songs on an audio CD.

MP3 is an abbreviation of MPEG audio layer – 3. This is an MPEG compression system to compress sound. MPEG stands for Motion Picture Experts Group (USA) that has established standards for compressing moving picture. MP3 format uses characteristics of the human ear to compress the music. Some of these characteristics are,

- There are certain sounds that the human ear cannot hear.
- There are certain sounds that the human ear hears much better than others.
- If there are two sounds playing simultaneously we hear the louder one but cannot hear the softer one

Using facts like these about the human ear, certain parts of a song can be eliminated without significantly hurting the quality of the song for the listener. Then by compressing the rest of the song with compression techniques we can shrink the song by a factor of 10 to 14.

The schematic diagram of MP3 player is shown in the Fig. 4.15.

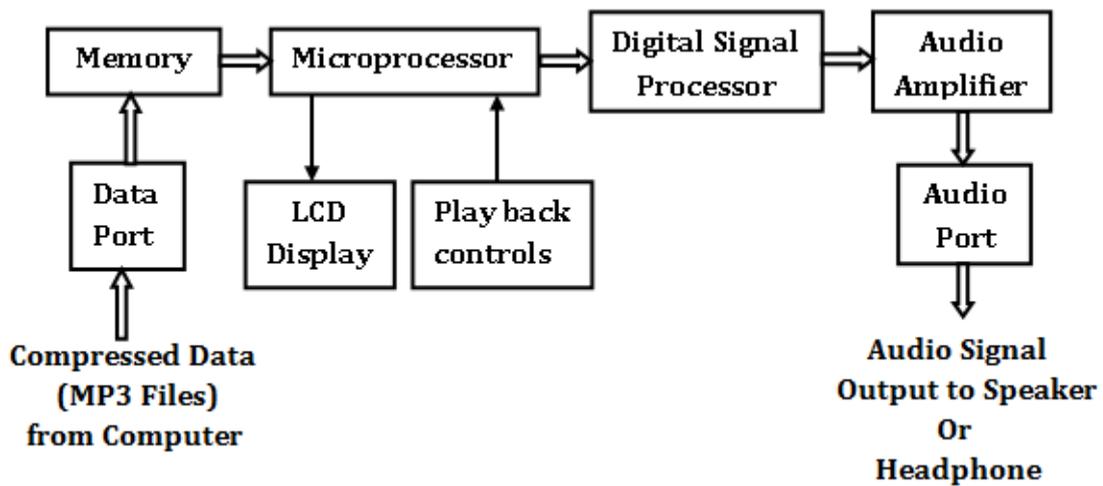


Fig. 4.15. Schematic Diagram of MP3 Player

Data Port:

The data port is connected to the parallel port of the computer. It is used to transfer data to the MP3 player.

Memory:

The MP3 files are saved in the player’s memory. Any one of the following solid state memories can be used.

- Internal flash memory
- Compact flash cards
- Smart media cards
- Memory stick

The advantages of using solid state memory is that there are no moving parts, and no moving parts means better reliability and no skip in the music.

Microprocessor:

The microprocessor is the brain of the player. It monitors user input through the playback controls, displays information about the current song on the LCD panel. It also sends directions to the DSP chip that tells it exactly how to process the audio.

Digital Signal Processor (DSP):

The DSP pulls the song data from memory, applies any special effects or equalization, and sends it to the audio amplifier.

The DSP runs a decompression algorithm to uncompress the compression of the MP3 files. Then a digital – to – analog converter converts the digital data back into sound signals.

Audio Amplifier:

The audio amplifier boosts the strength of the audio signal and sends it to the audio port.

Audio Port:

The audio port connects the audio signal output to the pair of headphones. Once you have downloaded the MP3 files into your portable MP3 player, you are ready to take your music anywhere. Most MP3 players are small and light weight.

4.3.3. DVD SYSTEM:

DVD stands for Digital Versatile Disc through commonly called Digital Video Disc. A DVD is very similar to a CD but has a much larger data capacity. A standard DVD holds about nearly seven times more data than a CD does.

- **DVD ROM:** It can only be read and not written.
 - **DVD-R:** It can only record data once and then function as a DVD ROM.
 - **DVD-RW & DVD RAM:** Both can record and erase data multiple times.
- Digital Versatile Discs (DVDs) provide superb Video, Audio and data storage and access – all on one disc.

DVD Audio:

DVD – audio is a format for delivering high –fidelity audio content on a DVD. It offers many channel configuration options at various sampling frequencies. Compared with the CD format, the much higher capacity DVD format enables the inclusion of considerably more music and for higher audio quality.

4.3.3.1. DVD PLAYER:

A DVD player is similar to CD player but needs manufacturing precision because of very fine encoding and extremely small track thickness. The schematic diagram of DVD player is shown in the figure 4.16.

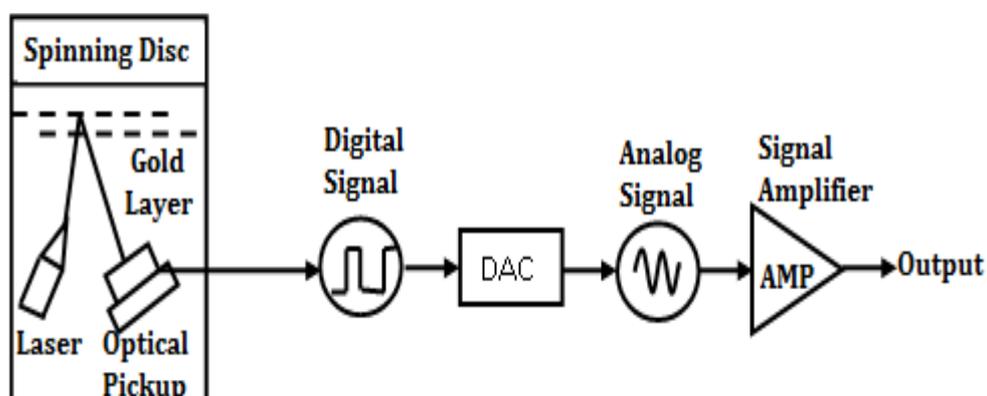


Fig.4.16 Schematic Diagram of DVD Player

The fundamental components of a DVD player are Drive motor, Laser and Lens System, Tracking Mechanisms, DAC and Signal Amplifier.

1,2,3,41 Drive Motor:

A drive motor is used to spin the disc. The disc is rotating with the rate of speed of 200 – 500 rpm depending on which track area is being read.

1,2,3,42 Laser and Lens System:

The laser and lens system focus the lights on the spinning disc (track) and read them. The light from the laser has a smaller wavelength.

1,2,3,43 Tracking Mechanism:

The tracking machine can be used to move the laser assembly, so that the laser's beam can follow spiral track.

1,2,3,44 DAC (Digital to Analog Converter):

This electronic circuitry converts the digital signal into analog signal.

1,2,3,45 Signal Amplifier:

It raises the signal to the desired level by using signal amplifier.

Advantages:

- The quality of picture and sound in DVD is better than on a video tape.
- It is high quality and maintained long time.
- East to maintain.
- Easy to store data.
- It is possible to watch a favorite song or scene again and again if desired.

4.3.4. STEREOPHONIC SOUND SYSTEM:

Stereophonic system is the science of reproducing, broadcasting or recording sound enabling the listener to hear the sound with its original spatial characteristics of direction and depth.

Let us consider that we are sitting in the centre of the hall to listen to a live orchestra as shown in Fig.4.17. The sound reaching our ears would not be identical. Instruments playing to the left of centre are heard by the left ear a fraction of a second sooner than by the right ear.

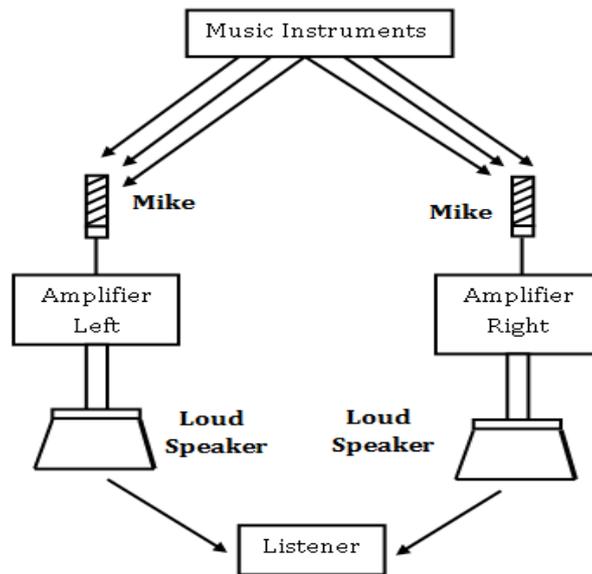


Fig.4.17 Stereophonic Systems

Only the sound of the instrument playing in dead centre reaches both ears at same instant. Hence these differences in time of arrival of various sounds enable us to distinguish the direction of each source. In addition, the sounds reflected from the walls, give rise to an echo effects.

The echoing sounds are sorted out by two ears, blending with the earlier received direct sound and these directional echo effects help us to judge the depth of the sound.

In monophonic system the sound usually comes from only one location. Hence the listener is unable to realize the original sound. By placing two mikes in front of the orchestra, one for left ear and the other for right ear, the effects of two directional, hearing are simulated as shown by the differences in length of the sound path lines in the figure. To increase the effects of directional hearing, several mikes may be used.

The sound patterns picked up by the left and right microphones are amplified by separate amplifiers and fed to the corresponding loudspeakers. The listener sitting in the hall between the two loudspeakers hears the differences between the two sound patterns.

4.3.5. HI-FI SYSTEM PRINCIPLE:

High fidelity (or) Hi-Fi is the name given to any audio system that can reproduce / record sound with a quality that resembles as closely as possible the original sound before recording or reproduction.

The word “Fidelity” means faithfulness. In audio systems it is used to indicate faithful reproduction of sound. The ideal fidelity should have the following characteristics.

1. The signal to noise ratio should be infinite.
2. It should be no frequency distortion.
3. It should be no non-linear distortion.
4. It should be no spatial distortion.
5. Environmental conditions should be simulated where the sound is being reproduced.

No sound system can be so perfect as to give ideal fidelity. The best fidelity systems are less than perfect but when sound is reproduced with a degree of similarity to the original or live sound, it can be said to be of high fidelity or simply ‘Hi-Fi’.

It is the following requirements are fulfilled, a sound system can reproduce Hi-Fi sound.

1. Signal to noise ratio should be better than 50db.
2. Frequency response should be flat over the frequency range of 40Hz to 15 KHz.
3. Non – linear distortion should not be more than 1%.
4. The system should process dynamic range of at least 80db.
5. Stereophonic effect should be provided.
6. Environmental conditions should be such as to eliminate the external noise in the listening room, and to give the desired reverberation time.

The microphones, recording amplifiers and devices, pick-ups, playback amplifiers and loudspeakers, all contribute to the high fidelity of the reproduced sound. Reflections, reverberation, ambient noise also affect fidelity. High fidelity can be achieved by using low noise components and proper design of the circuits and devices.

4.3.6. DTS (DIGITAL THEATER SYSTEM):

DTS also known as Digital Theater System. In order to achieve good quality sound a format or codec should be developed for sound system. Dolby lab has developed a new codec (encoder) called 5.1 channel system. This system encodes the audio as five primary channels plus a special LFE (Low Frequency Effect) channel for sub – woofer. Encoders and decoders support many channel combinations and stereo, four-channel.

Working:

Information of time code is optically imaged on to the film. An optical LED reader reads the time code data from the film and sends it to the DTS processor. The DTS processor uses this time code to synchronize the projected image with the sound track audio. The actual audio is recorded in compressed form on standard CD-ROM at

a bit rate of 1103Kbits/S. the processor also acts a transport mechanism, as it holds and reads the audio discs. Newer units can generally hold three discs, allowing a single processor / transport to handle two-disc film soundtracks along with a third disc containing sound for trailers.

DTS audio is stored on a separate set of CD-ROM media, whose greater storage capacity gives the potential to deliver better audio fidelity. When the DTS audio track in the film is encoded at its highest bit rate (1536 Kbits/ Sec) most audio programs can be recorded.

Play Back:

Both music and movie DVDs allow delivery of DTS audio tracks. Nearly all DVD player / Receivers manufactured today can decodes DTS. For PC play back, many software players support the decoding of DTS.

In addition to standard 5.1 channel DTS surround, other technologies in DTS are also available. They are,

1. DTS 70mm: this is a process designed for play back in motion picture theaters quipped with 70mm projection and 6-track stereo phonic surround sound.
2. DTS-ES (DTS Extended surround system).

REVIEW QUESTIONS AND ANSWERS

PART-A

(TWO MARKS)

1. State Acoustic Principle.

The study about the changes in the sound waves are said to be acoustic. In acoustical system, the oscillations in sound pressure are picked up by some form of receiving electro acoustic transducer called microphones.

2. What is a Microphone? Give its Types.

Microphones are used for converting sound signals into electrical (audio) signals. It is operated only in air. **Two types:**

- Pressure microphones &
- Pressure gradient microphones.

3. What is a Pressure Microphone? Give Examples.

In pressure microphones, the acoustic pressure acts only one side of the moving element & the resulting output is proportional to the pressure applied to the moving element.

Example: carbon microphone, condenser microphone
Piezo electric microphone & moving coil microphone.

4. What is a Pressure Gradient Microphone? Give Examples.

In pressure microphones, the acoustic pressure acts only one side of the moving element & the resulting output is proportional to the difference between the pressure acting on the two sides of the moving element. Example: Velocity ribbon microphone.

5. What are the Advantages & Disadvantages of Carbon Microphone?

Advantages: Electrical output is high, low cost. It is robust.

Disadvantages: Frequency response is poor, Hissing sound is produced.

6. What is the Application of Carbon Microphone?

It is widely used in telephone & radio communication.

7. What are the Advantages & Disadvantages of Condenser Microphone?

Advantages: low distortion, small in size, high signal to noise ratio.

Disadvantages: high internal impedance requires voltage ranging from 200 to 400 volts.

8. What is the Application of Condenser Microphone?

- It is used in acoustic research for standard calibration
- Purpose & for high fidelity recording of sound.

9. What are the Advantages & Disadvantages of Piezoelectric Microphone?

Advantages: high sensitivity, small size, low cost, frequency response is good

Disadvantages: not suitable for hot climates as the crystal loses the piezoelectric properties.

10. What is the Application of Piezoelectric Microphone?

It is used in public address system & for hearing aids.

11. What is the Advantages of Moving Coil Microphones?

- Light weight
- Requires no external voltage
- Internal impedance is low

12. What is Velocity Ribbon Microphone?

It works on the principle of pressure gradient which implies that driving force acting on a moving element is proportional to the difference in pressures acting on its two sides.

13. What is the Use of Velocity Ribbon Microphone?

- It is used particularly in studio work.

14. What are the Advantages & Disadvantages of Velocity Ribbon Microphone?

Advantage: Response curve is better than moving coil type.

Disadvantage: It requires a built in type transformer, its internal impedance is low.

15. Define Loudspeaker.

Loudspeaker is an electro acoustic transducer, which converts electrical signal into sound signal.

16. Compare Carbon Microphone & Condenser Microphone.

S.No.	Carbon Microphone	Condenser Microphone
1	Stiffness Controlled System	Mass Controlled System
2	High Electrical Output	Low Electrical Output
3	Operation depends upon variation of resistance of carbon button	Operation depends upon variation of Capacitance of a capacitor

17. Compare Condenser Microphone & Piezoelectric Microphone.

S.No.	Piezo Electric Microphone	Condenser Microphone
1	Stiffness Controlled System	Mass Controlled System
2	High Electrical Output	Low Electrical Output
3	Operation depends upon piezoelectric effect	Operation depends upon variation of Capacitance of a capacitor

18. What are the Characteristics of Loudspeaker?

- Efficiency approaches 100 percent
- Harmonic & inter modulations not present
- Input signal are produced faithfully

19. What are the Types of Loudspeaker?

- Dynamic cone type moving coil loud speaker.
- Horn Type moving coil loudspeaker
- Electrostatic type loudspeaker

20. What are the Advantages & Disadvantages of Dynamic Cone Type Loudspeaker?

Advantage: Compact in size, low cost & better response in audio range

Disadvantage: Low efficiency, low power, narrow directivity

21. What is the Application of Dynamic Cone Type Loudspeaker?

- It is used in all audio systems

22. What are the Advantages of Horn Type Loudspeaker?

- Sound effect of this speaker is very high & efficient
- Diaphragm is smaller in size
- Power handling capacity is more than 100 watts.

23. What is the Use of Electrostatic Type Loudspeaker?

- This type of loud speakers used in high fidelity applications.

24. Compare Cone Type & Horn Type Loudspeaker.

S.No.	Cone Type Loud Speaker	Horn Type Loud Speaker
1	Sound effect of this speaker is less	Sound effect of this speaker is very high & efficient
2	Diaphragm is larger in size	Diaphragm is smaller in size
3	Power handling capacity is from few milli watts to several 100 watts	Power handling capacity is more than 100 watts

25. Give the Speaker Specifications.

- Diameter of the speaker frame
- Type & Wattage rating
- Voice coil impedance

26. Compare Woofer & Tweeter.

S.No.	Woofer	Tweeter
1	It is used for the reproduction of low frequencies	It is used for the reproduction of high frequencies
2	It has large diameter cone	It has small diameter cone
3	High wattage rating with more cone depth	Low wattage rating with less cone depth

27. Define Cross Over Network.

- To improve the quality & performance of amplifiers & radio receivers, various control circuits are used. These circuits are called cross over networks.

28. What is CD? Mention its Layers.

CD could store wide variety of data formats like text picture, video, software etc. It is 12cm or 5inch in diameter. It has three layers: Transparent substrate, thin metallic layer, outer layer of protective acrylic.

29. What are the Advantages of CD?

- High signal noise ratio
- Small size & low distortion
- Excellent Frequency range

30. What is MP3?

MPEG-1 Audio Layer 3 commonly referred to as MP3. It is a digital encoding format using a form of lossy data compression. MPEG stands for Moving Picture Experts Group. It is for compressing moving picture. It is an audio specific format.

31. What is DVD?

DVD could store wide variety of data formats like text picture, audio, video, software etc. It needs manufacturing precision because of very fine encoding & extremely small track thickness.

32. What are the Characteristics of Hi-Fi System Principles?

- Signal to noise ratio should be finite
- No frequency distortion
- No non linear distortion
- No spatial distortion

33. Expand DTS.

Digital Theater Sound

34. What is Hi-Fi System?

When the sound reproduced with a degree of similarity to the original or live sound, it can be said to be of high fidelity or simply Hi-Fi.

35. Define Stereophonic System.

A stereophonic system is a two channel system of sound recording & reproduction, which gives the feeling of depth & direction to the reproduced sound.

36. What is Bit Rate?

The bit rate is used to specify how many kilobits the file may use per second of audio as in when ripping a CD to MP3 format.

37. What is Surround Sound System?

Multichannel audio technique is used for enriching the quality of sound reproduction of a recorded source. It is for 3D sphere of human hearing achieved with audio channels.

PART B

(THREE MARKS)

1. Comparison of Carbon and Condenser Microphone.
2. Comparison of Condenser and Piezo Electric Microphone.
3. Draw and explain Electrostatic Type Loudspeaker.
4. Difference between Cone Type and Horn Type Loudspeaker.
5. Draw the Woofer, Midrange and Tweeter & frequency Response with Specification.
6. Difference between Woofer and Tweeter.
7. Draw and explain Cross Over Network
8. Draw and explain 3-Way Cross over Network.
9. What is Surround Sound System? Explain Creating Surround Sound
10. What is Surround Sound System? Explain Mapping Channels to Speaker
11. What is Surround Sound System? Explain Surround Sound Specification.
12. Explain Three Layers of Compact Disc System.
13. Explain MP3 System.
14. Explain Stereophonic Sound System.
15. Explain HI-FI System Principle.
16. Explain DTS.

PART C

(TEN MARKS)

1. Draw and explain Carbon Microphone. With application, advantages and disadvantages.
2. Draw and explain Condenser Microphone. With application, advantages and disadvantages.
3. Draw and explain Piezo-Electric Microphone. With application, advantages and disadvantages.
4. Draw and explain Moving Coil (Electrodynamics) Microphone. With advantages.
5. Draw and explain Velocity Ribbon (Pressure Gradient) Microphone. With application, advantages and disadvantages.
6. Comparison of Carbon and condenser microphone.

7. Comparison of Condenser and Piezo electric microphone.
8. Draw and explain Dynamic Cone type moving coil loudspeaker. With application, advantages and disadvantages.
9. Draw and explain Horn type loudspeaker. With application, advantages and disadvantages.
10. Draw and explain Electrostatic type loudspeaker. With application, advantages and disadvantages.
11. Difference between cone type and Horn type Loudspeaker.
12. Draw and explain Woofer, Midrange and Tweeter & frequency Response with Specification.
13. Difference between Woofer and Tweeter.
14. Draw and explain Cross over Network and 3-Way Cross Over Network.
15. What is Surround Sound System? Explain i) Creating Surround Sound
ii). Mapping Channels to Speaker iii). Surround Sound Specification.
16. Draw and Explain Three Layers of Compact Disc System.
17. Draw and Explain CD Recording.
18. Draw and Explain CD Reproduction with advantages
19. Briefly explain MP3 System.
20. Draw and Explain DVD Player.
21. Draw and Explain Stereophonic sound system.

UNIT - V

VIDEO SYSTEMS

5.1. INTRODUCTION TO TELEVISION:

Tele means 'distance' and **vision** means 'seeing'. Thus in general television means 'seeing from a distance'. **Broadcasting** means 'to send in all direction'.

The picture or scene in the form is optical in nature. So we have to convert it into electrical signal before transmitting it to a long distance. At the receiver side we have to convert the electrical signal into optical signal to view in the screen.

Camera tubes are used to convert optical signal into electrical signal and picture tubes are used to convert electrical signal into optical signal.

5.1. (a) TV FUNDAMENTALS:

Television means "to see from a distance". The first television was given by J.L. Baird in UK and C.F. Jenkins in USA around 1927. The fundamental aim of a television system is to extend the sense of sight beyond its normal limits, along with the sound associated with the scene being televised. TV system is an extension of the science of radio communication, used for entertainment and news programs.

The microphone converts sound waves to corresponding electrical variations for the audio signal. Similarly, the camera tube converts its light input to corresponding electrical variation for the video signal. The camera tube is to "VIDEO" and the microphone is to "AUDIO". At the end of the system, the picture tube converts the video signal voltage into the light variations and the loud speaker reproduces the original sound.

Most of the television systems, the picture signal is amplitude modulated and sound signal is frequency modulated before transmission. The carrier frequencies are suitably spaced and the modulated outputs radiated through a common antenna. Thus each broadcasting station can have its own carrier frequency and the receiver can then be tuned to select any desired station.

MONOCHROME TELEVISION:

5.1.1. Scanning Principles:

Scanning is a process, in which it converts the charge image produced inside the TV camera tube into a variable electrical signal. The TV picture is scanned in sequential series of horizontal lines one under the other. It is the same way as we would read a text page to cover all the words in one line and all the lines on the page. All the picture elements are scanned in successive order, from left to right and from top to bottom, one line at a time.

The scene is scanned rapidly both in the horizontal and vertical directions simultaneously to provide sufficient number of complete pictures or frames per second to

give the continuous motion. The frame repetition rate is 25 per second in most television systems.

The sequence for scanning all the picture elements is as follows. They are,

1. The electron beam sweeps across one horizontal line, converting all the picture elements in that line. This time is called trace. It is done by horizontal scanning.
2. At the end of each line, the beam returns very quickly to the left side to begin scanning the next horizontal line. The return lines are called retrace or fly back. No picture information is scanned during retrace because both the camera tube and picture tube are blanked out for this period. Thus the retraces must be very rapid, since they are wasted time in terms of picture information.
3. When the beam is returned to the left side, its vertical position is lowered so that the beam will scan the next line down and not repeat the same line. This is accomplished by the vertical scanning motion of the beam.

Types of scanning:

- a. Horizontal Scanning
- b. Vertical Scanning
- c. Interlaced Scanning

(A). HORIZONTAL SCANNING:

The linear rise of current applied to the horizontal deflection coils, deflects the electron beam across the screen with a continuous, uniform motion for the trace from left to right. At the peak of the rise, the saw tooth wave direction decreases rapidly to its initial value. These fast reversals produce the retrace or fly back. The start of horizontal trace is at the left edge of the raster. The finish is at the right edge, where the fly back produces retrace, back to the left side.

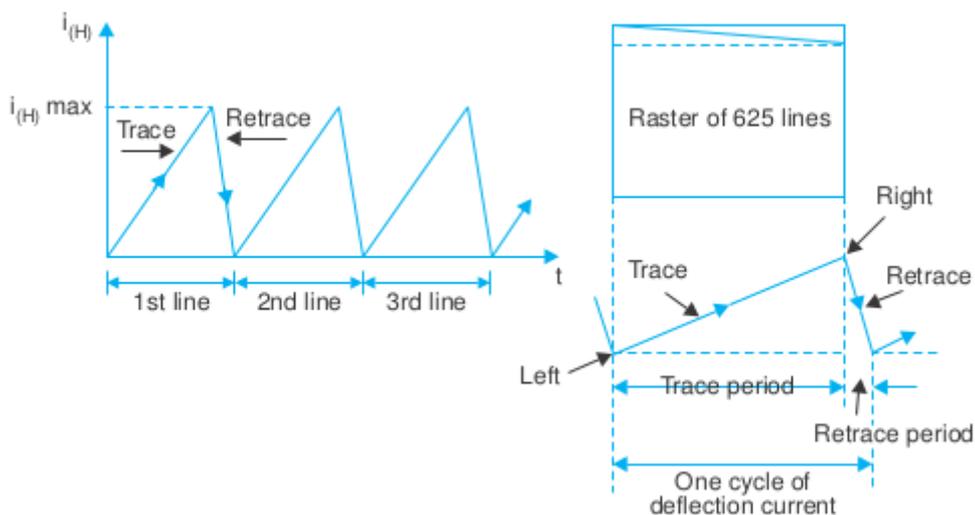


Fig.5.1. Waveform of Horizontal Deflection Coils

The horizontal deflection coil waveform is shown in Fig.5.1. Note that 'up' on the saw tooth wave corresponds to horizontal deflection to the right. The dark line indicates the useful scanning time and the dashed lines correspond to the retrace time. The number of scanning lines for one complete picture should be large in order to include the greater number of picture elements and thus more detail. In the 625 line system, the horizontal scanning frequency is 15,625Hz.

(B). VERTICAL SCANNING:

The saw tooth current applied to the vertical deflection coils moves the electron beam from top to bottom of the raster at a uniform speed, while the electron beam is being deflected horizontally. Thus the beam produces one below the other while moving from top to bottom. The vertical scanning waveforms are shown in Fig.5.2.

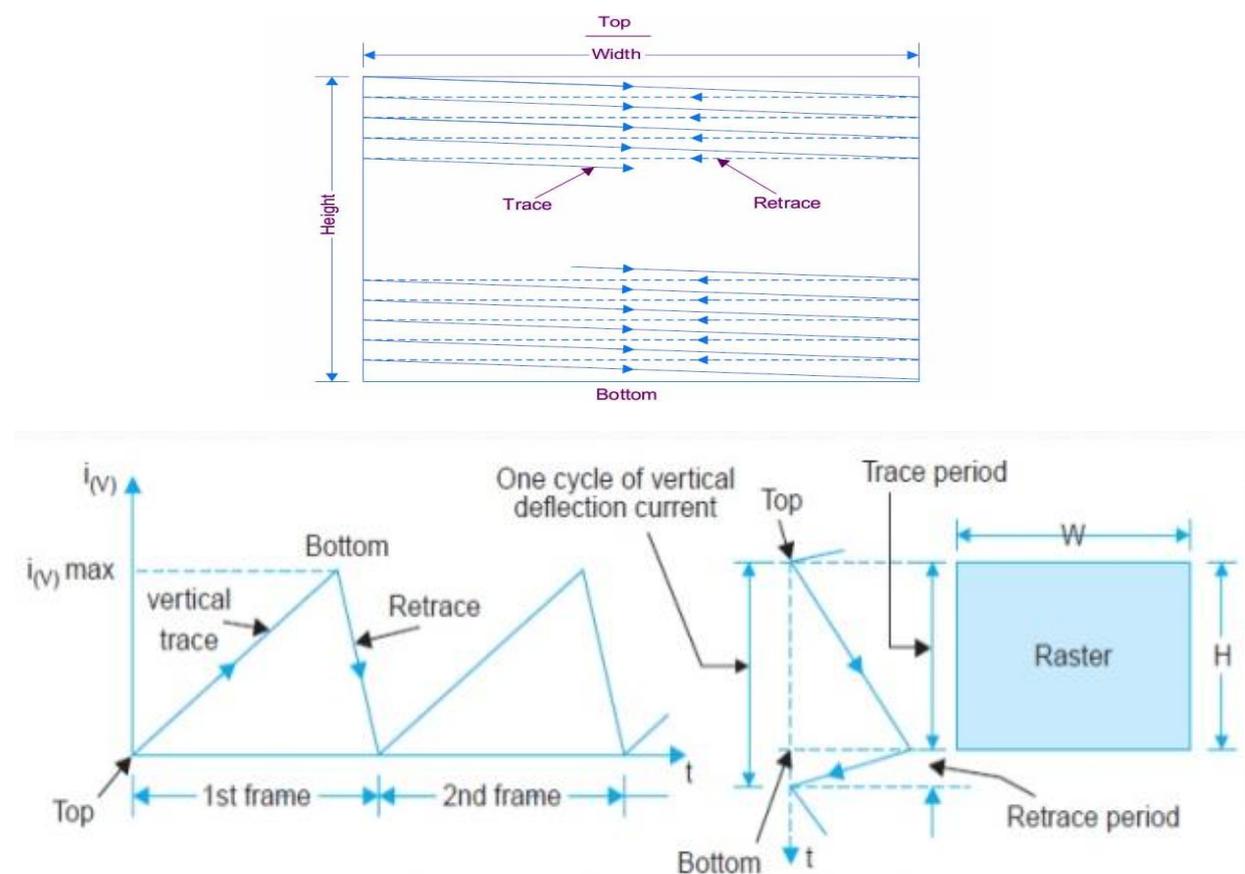


Fig.5.2. Vertical Deflection Waveform

As shown in the Fig, the trace part of the saw tooth wave for vertical scanning deflects the beam to the bottom of the raster. Then the rapid vertical retrace returns the beam to the top. The maximum amplitude of the vertical sweep current brings the beam to the bottom of the raster.

During vertical retrace the horizontal scanning is continuous and several lines get scanned during this period. By using the scanning beam, the information is converted into electrical signal. The vertical scanning frequency is 25Hz, i.e., 25 frames are scanned in one second.

During both the horizontal and vertical retrace intervals, the scanning beams at the camera tube and picture tube are blanked and no picture information is either picked up or reproduced.

(C) FLICKER:

The scanning rate of 25 frames per second in television picture is not rapid enough to allow the brightness of one picture or frame to blend smoothly into the next. By this effect the screen is blanked between successive frames. This results in a definite flicker of light as the screen is made alternatively bright and dark.

In progressive scanning, all the lines in the frame are being scanned in progressive order from top to bottom. There would be only 25 blank-outs per second, hence **Flicker** effect is produced. Scanning of 50 complete frames per second makes 50 blank-outs per second. Now this fast variation is not detected by human eyes, and hence flicker effect is avoided. By this effect the video frequencies corresponding to the picture elements are doubled in a line. This type of problem is solved by using the method of Interlaced Scanning.

(D) INTERLACED SCANNING:

In this television pictures, an effective rate of 50 vertical scans per second is utilized to reduce flicker. This is accomplished by increasing the downward rate of travel of the scanning electron beam, so that every alternate line gets scanned instead of every successive line. When the beam reaches the bottom of the picture frame, it quickly returns to the top, to scan those lines that were missed in the previous scanning. Thus the total numbers of lines are divided into two groups called '**Fields**'. Each field is scanned alternatively. This method of scanning is known as '**Interlaced Scanning**'. The representation of interlaced scanning is shown in the Fig.5.3. In the 625 line system, for the interlaced scanning.

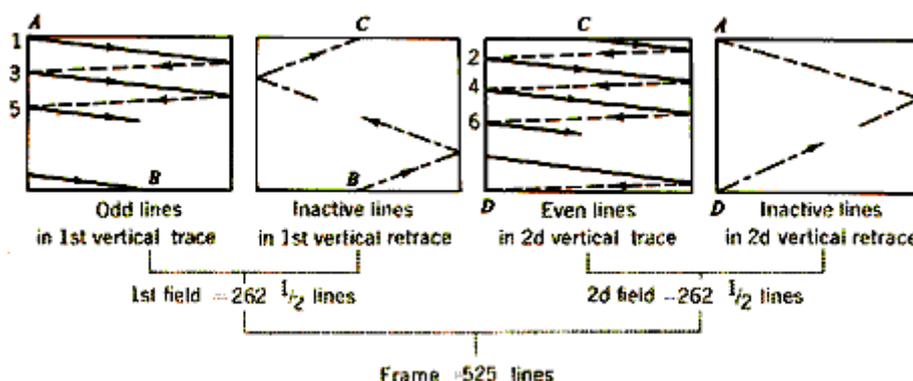


Fig.5.3. Interlaced Scanning

The Total Lines of each Frame are divided into Two Fields:

- Odd field and
- Even field

The first and following odd fields contain the odd lines in the frames. The second and all the even fields include the even scanning lines. Given two fields per frame and 25 complete frames scanned per second, and frame repetition rate is 50 per second. In fact, doubling the vertical scanning frequency from the 25 Hz frame rate to the 50Hz field rate, makes the beam scan every other line in the frame. Each field contains 312.5 lines.

The beam starts at A, and sweeps across the frame with uniform velocity to cover all the picture elements in one horizontal line. At the end of this trace, the beam retraces rapidly to the left side of the frame and to begin scanning the next horizontal line. Note that the horizontal line slopes downward in the direction of scanning because the vertical deflection signal simultaneously produces a vertical scanning motion, which is slow compared with horizontal scanning.

After line '1' is scanned, the beam is at the left side, ready to scan line 3, omitting the second line. This line skipping is accomplished by doubling the vertical scanning frequency from 25 Hz to 50Hz. Now the electron beam scans all the odd lines, than finally reaching a position, such as point B in the figure, at the bottom of the frame. At time B, the vertical retrace begins because of fly back on their vertical saw tooth deflection signal. Then the beam returns to the top of the frame to begin the second or even fields.

Now the beam moves from B up to C, travelling a whole number of horizontal lines. During this period 20 horizontal lines are placed. These 20 lines are known as inactive lines, because the scanning beam is cut-off during this period. Thus the second field starts at the middle of the raster. After scanning a half-line from point C, the beam scans line 2 in the second field.

Then the beam scans the even lines that were omitted during the scanning of the first field. The vertical scanning motion is exactly the same as in the previous field. As a result, all the even lines in the second field are scanned, down to point D. Points D and B are a half line away from each other because the second field starts at a half line point.

The vertical retrace in the second fields starts at point D. From here, the vertical fly back causes the beam to return to the top. Since there are a whole number of vertical retrace lines, the beam finishes the second vertical retrace at A. the beam will always finish the second vertical retrace where the first trace started because the number of vertical retrace lines is the same in both fields. All odd fields begin at point A. all even fields begin at point C. this process is continued at a fast rate of 50 times a second, which avoids flicker satisfactorily.

5.1.2. HORIZONTAL AND VERTICAL BLANKING:

The purpose of the blanking pulses is to make invisible the retraces required in scanning. The blanking pulses are rectangular pulses placed during the vertical and horizontal retrace intervals with amplitude of 75% of its black level.

The horizontal blanking pulses are used to blank out the horizontal retrace from right to left for each line. The frequency of horizontal blanking pulse is 15,625Hz. Similarly the vertical blanking pulses are used to blank out the vertical retrace from bottom to top for each field. The frequency of vertical blanking pulse is 50Hz. The time needed for horizontal blanking is 18.75 percent of each horizontal line. Similarly the time needed for vertical blanking is 6.4 percent of each vertical field.

5.1.3. HORIZONTAL AND VERTICAL SYNCHRONIZATION:

The transmitter and receiver scanning in step, special synchronizing signals must be transmitted with the picture information for the receiver. These timing signals are rectangular pulses that are used to control both camera and receiver scanning. These rectangular timing signals are called **Synchronizing Pulses**.

The synchronizing pulses are transmitted as part of the complete picture signal for the receiver, but they occur during the blanking time when no picture information is transmitted.

A horizontal synchronization pulse at the end of each line determines the start of horizontal retrace. Vertical synchronization at the end of each field determines the start of vertical retrace.

Without the vertical field synchronization, the reproduced picture at the receiver does not hold vertically, it rolls up or down on the picture tube screen. If the horizontal scanning lines are not synchronized the picture does not hold horizontally, it slips to the left or right.

5.1.4. ASPECT RATIO:

The width to height ratio of the picture frame is called the **aspect ratio**. It is standardized at 4:3, this aspect ratio makes the picture wider than its height by a factor of 1.33.

$$\text{Aspect ratio} = \frac{\text{Width of the Frame}}{\text{Height of the Frame}} = \frac{4}{3}$$

In practical use, a picture is viewed normally in rectangular format. So we can get best view of an object in rectangular format. Our eyes can view with more ease and comfort, when the width of the picture is more than its height. The resolution of the retina is also concentrated mostly in horizontal plane.

The usage of rectangular frame in motion pictures also has the aspect ratio of 4:3. This enables direct television transmission of film programs without wastage of any film area. Due

to the above reasons, the width should be four times and the height should be three times. Only such a dimension could produce a satisfactory, natural view of the image.

5.1.5. COMPOSITE VIDEO SIGNAL:

The composite video signal consists of,

1. Signal corresponding to the desired picture information
 - a. Luminance signal alone in the case of monochrome CVS
 - b. Chroma signal alone in the of colorplexed CVS
2. Horizontal and vertical blanking pulse
3. Horizontal and vertical sync pluses
4. Equalizing pulse

The blanking and sync pulses are added during retrace intervals with an amplitude of 75% and 100% levels, to obtain higher efficiency of picture transmission.

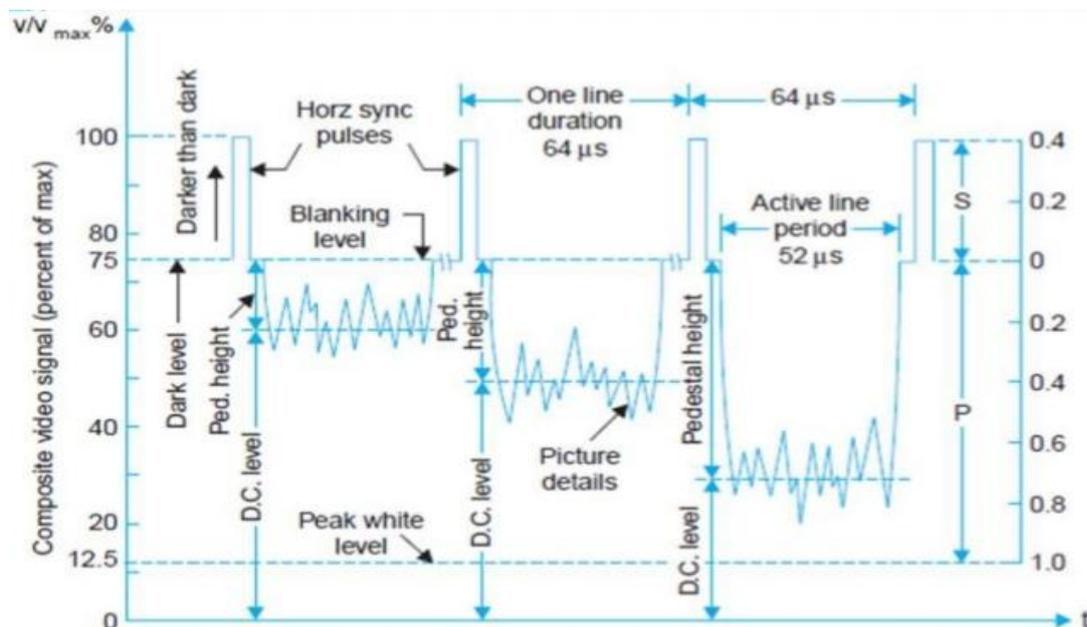


Fig.5.4 Composite Video Signal of Three Different Levels

The horizontal sync pulse is needed at the end of each active line period, whereas a vertical sync –pulse is required after each field is scanned. Fig 5.4 shows the details of composite video signal of three different levels, each corresponding to a different brightness level of the scene.

During the forward scan, the video signal level changes between black and white levels occupying intermediate grey level also, depending upon the picture brightness in the scanned area. Thus the picture information may vary between 12.5 percent to about 67.5 percent of the composite video signal.

The picture details about white areas are placed in lower amplitudes, and the dark (blank) areas are placed in higher amplitudes. The lowest 10 percent of the voltage range is not used, to minimize noise effects.

Average Brightness means, the average of value of brightness variation of horizontal lines in one frame. The distance between average brightness and blanking level is called **pedestal height**. Similarly the distance between zero level and the average brightness is called **DC level**. The higher amplitude composite video signal represents dark picture and lower amplitude represents bright picture.

5.1.6. TV BROADCASTING STANDARDS:

We have to send both the audio and video signal for TV transmission. We have to modulate both the signal for long distance transmission. At the receiver we have to properly demodulate them. So we have to standardize some factors like field frequency, lines per frame, line frequency, video bandwidth, channel bandwidth, audio and video modulation and color system to be used.

Based on TV standard available in different countries, we have to use the following systems. They are,

1. FCC (Federal Communication Commission)
2. NTSC (National Television System Committee)
3. CCIR (International Radio Consultative Committee)
4. PAL (Phase Alternation with Line)
5. SECAM (Sequential Couleurs (Color) a Memory)

COMPARISON OF TV STANDARDS:

The TV standards of different countries are compared in table.5.1.

S.NO	standards	Western, Europe, Middle East, India & most Asian countries (CCIR B)	North and south America including US, Canada, Mexico and Japan (NTSC)	England (PAL)	USSR	SECAM (FRANCE)
1	Lines Per Frame	625	525	625	625	625
2	Frames per Second	25	30	25	25	25
3	Field frequency (Hz)	50	60	50	50	50
4	Line Frequency (Hz)	15625	15750	15625	15625	15625
5	Video bandwidth (MHz)	5 Or 6	402	5.5	6	6
6	Channel	7 Or 8	6	8	8	8

	bandwidth (MHz)					
7	Video modulation	Negative	Negative	Negative	Negative	Positive
8	Picture modulation	AM	AM	AM	AM	FM
9	Sound signal modulation	FM	FM	FM	FM	AM
10	Color system	PAL or CCIR B	NTSC	PAL	SECAM	SECAM
11	Color subcarrier in MHz	4.43	3.58	4.43	4.43	4.43

5.1.7. BLOCK DIAGRAM OF TV TRANSMITTER:

In a TV transmitter, we have to transmit both video and audio signal. So we have two type of transmitters, they are

1. Visual transmitter or exciter and
2. Aural transmitter or exciter

Here visual transmitter is used for transmitting video signals and aural transmitter is used for transmitting audio or sound signal. Outputs of these two transmitters are combined before transmitting through common antenna. The combining network is called as **diplexer**.

Before transmission, the video signal is amplitude modulated (AM) and audio signal is frequency modulated (FM). After modulation and proper power amplification they are fed to a common antenna through the diplexer.

Initially a camera is used to convert optical image into electrical signal. After adding sync pulses and blanking pulses we get composite video signal which is to be AM modulated. Similarly, a microphone (MIC) is used to convert audio signal into electrical signal. After pre-emphasis and proper amplification we have to FM modulate the audio signal. The block diagram of TV transmitter is shown in the Fig.5.5.

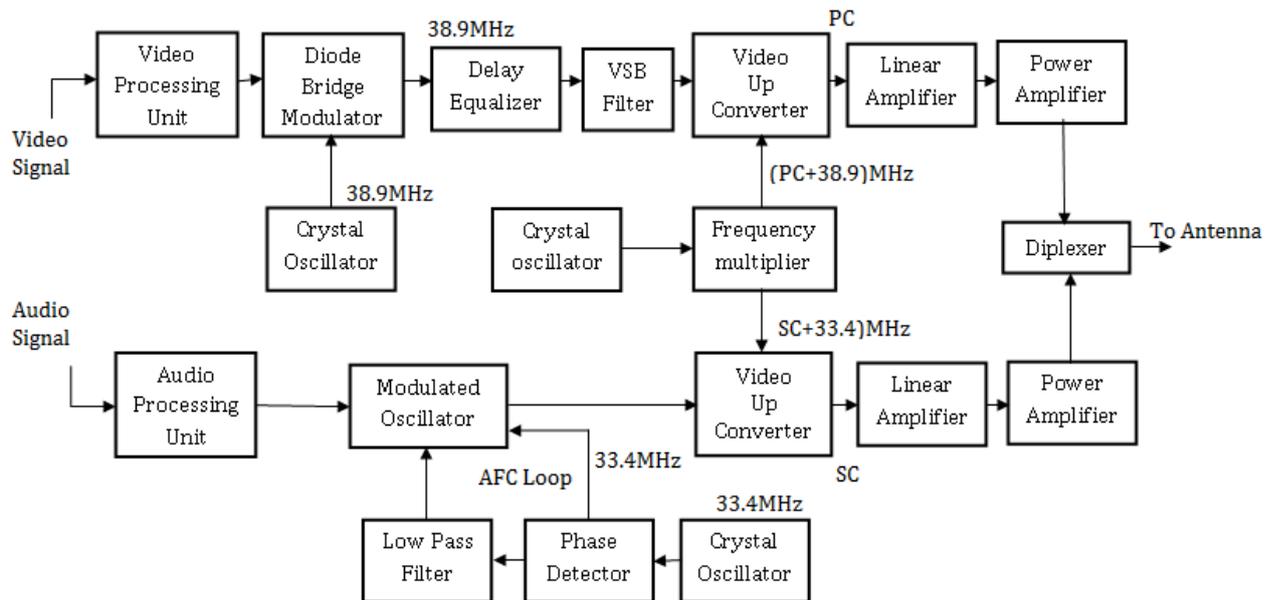


Fig.5.5. IF Modulated TV Transmitter

a. Visual Transmitter or Exciter:

This unit consists of a video processing unit, a visual modulator, VSB filter and phase compensator or delay equalizer and frequency converter.

1. Video Processing Unit:

The video signal input of 1Vp-p is processed in a stabilizing amplifier and sync regenerator to minimize the effects of hum and noise, sync compression and other distortions, to correct it into the standard form. Controls for linearity correction, gain, black set-up, sync level, picture sync ratio, sync stretching etc are provided here to reconstitute the pulses that may have been distorted.

2. Diode Bridge Modulator or Visual Modulator:

A diode bridge modulator or a diode balanced modulator is commonly used for low level modulation. It modulates the incoming standard video signal to its video IF signal with a frequency of 38.9MHz. It is an amplitude modulator.

3. VSB Filter:

The VSB filter is designed with lumped components, typically consisting of four sections of low pass ladder network. It attenuates the signal frequencies beyond 1.25MHz above the carrier by more than 20db.

4. Phase Delay Compensator of Equalizer:

The phase compensator is built-in with the VSB filter to compensate phase distortion.

5. Video up Converter:

The second crystal oscillator generates a frequency equal to the picture carrier PC plus the IF of 38.9MHz. This is heterodyned with the VSB modulated IF carrier in the up converter, which also employs a diode bridge modulator. A band pass filter is also provided at the output of the up converter. It is tuned to picture carrier frequency, so that only the VSB modulated picture carrier passes through the filter.

6. Power Amplifier:

The up converter frequencies are amplified in linear amplifiers employing ceramic tetrodes in grounded grid configuration. This stage increases the input signal power to the required transmitted power.

b. Aural Transmitter or Exciter:

Aural exciter consists of audio processing unit, an aural modulator, AFC, Audio Up Converter and Amplifier.

1. Audio Processing Unit:

The audio signal is given a pre-emphasis by a high pass network with the time constant of $50\mu\text{s}$, and the signal is amplified to make the signal at standard level.

2. Aural Modulator:

The signal is then fed to a crystal controlled oscillator modulator for frequency modulation. The audio signal voltage is applied to a varactor diode in parallel with a tuned circuit of an LC oscillator. It generates a centre frequency equal to the sound IF carrier of 33.4MHz.

3. Automatic Frequency Control (AFC):

The center frequency of the LC oscillator is stabilized by an Automatic Frequency Control (AFC) circuit that compares this frequency with the crystal driven stable frequency. The 33.4MHz oscillator frequency and the crystal controlled 33.4MHz frequency were compared in a phase detector and produce an output voltage proportional to the difference between the two frequencies. When the two frequencies are identical a 90° out of phase, the detector produces a zero output voltage. Otherwise, some error voltage is produced at its output, which drives the oscillator frequency towards its output, which drives the oscillator frequency towards 33.4MHz. Hence, by continuous looping, stabilization is easily achieved.

4. Frequency up Converter:

The frequency modulated IF of 33.4MHz is then fed to a mixer, where it is heterodyned with a frequency equal to $(SC+33.4)$ MHz to obtain the difference frequency output at the desired sound carrier frequency of the channel.

5. Output Power Amplifier:

The mixer output is amplified in linear class A, class AB or Class B stages that drive the final drive the final output stage. The final stage may employ ceramic tetrodes operated as narrow band class C amplifier.

6. Diplexer:

A Common Impedance Notch (CIN) diplexer unit combines the outputs of both the visual and aural transmitters and feeding a common broad band transmitting antenna system. The CIN diplexer consists of two units of 3db directional couplers and two units of aural notch cavities suitably connected by co-axial transmission lines. It also maintains sufficient isolation between the visual and aural inputs.

5.1.8 BLOCK DIAGRAM OF TV RECEIVER:

Television receiver is to pick up the RF signals transmitted from the television stations and process signal from one TV station to produce visible picture and audible sound which in all respects to the original telecast. The block diagram of Monochrome TV receiver is shown in the Fig.5.6.

1. Receiving Antenna Section:

All TV antennas sections are mounted in horizontal position for better reception and favorable signal to noise ratio. Half wave length antenna is used for VHF band. For this Yagi and Yagi-Uda antennas are used. Fan dipole, Rhombic and Parabolic Reflector Antennas are used for Ultra High Frequency (UHF) band.

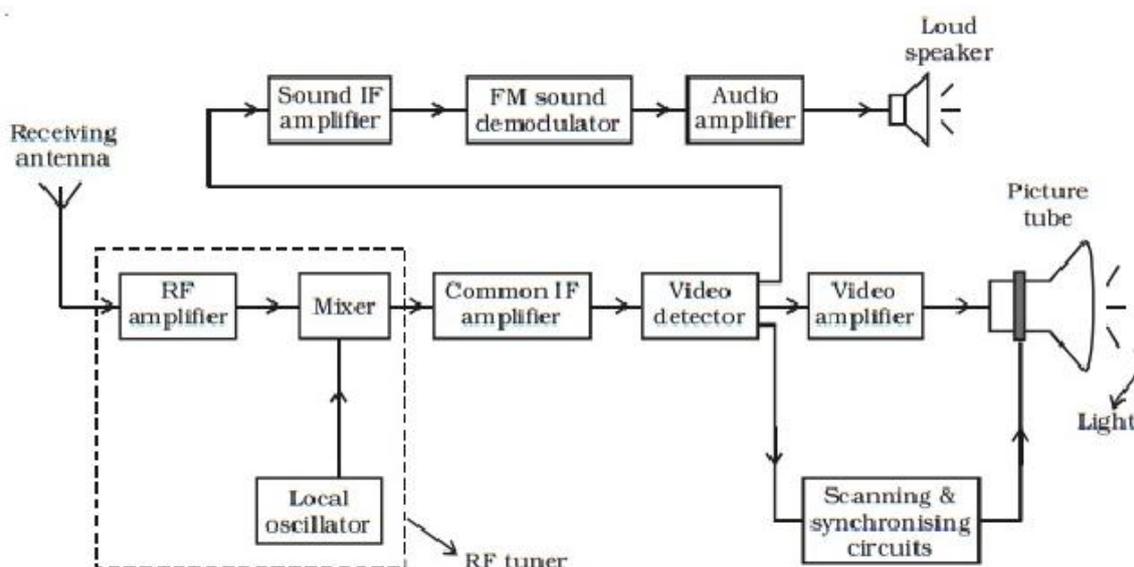


Fig.5.6. Block Diagram of TV Receiver

2. RF Tuner Section:

RF Tuner section consists of RF amplifier, mixer and local oscillator. It is normally mounted on a separate sub-chassis, called the 'Front End' or 'RF Tuner'. It is used to amplify both sound and picture signals picked up by the antenna, and to convert the carrier frequencies and their associated bands into the intermediate frequencies and their side bands. The standard IF frequencies for the 625-B system are, Picture IF = 38.9MHz and sound IF = 33.4MHz. The tuning of RF and oscillator tuned circuit is preset for switching in different channels. A fine tuning control is also provided in this section, for obtaining best picture results on the screen.

3. Common If Amplifier:

The main function of this section is to amplify the modulated IF signal over its entire bandwidth. This section needs overall gain of about 8000. This section is also called Video IF amplifier, since composite video signal is the envelope of the modulated picture IF signal. Generally 3 or 4 transistorized IF stages are needed for getting required gain.

4. Video Detector:

Video detector is a rectifier cum high frequency filter. This circuit is used to recover the original video signal from the modulated carrier signal. The detector may use either series circuit or shunt circuit. Semiconductor diodes are generally used for linear detection without distortion.

5. Video Amplifier:

The amplifier of the composite video signal at the output of the video detector is not large enough to drive the picture tube directly. Hence further amplification is necessary, and this is provided by the video amplifier. The video signal on application to the picture tube varies the intensity of its beam. The gain control of the video amplifier constitutes the **contrast control**. This signal is then applied to the picture tube, to reproduce original picture details.

6. Scanning and Synchronizing Circuit:

The sync pulses include horizontal, vertical and equalizing pulses. These are part of composite video signal, present at the top 25 percent of the signal amplitude. These are separated from the composite video signal by the sync separator.

The vertical and horizontal sync pulses can be separated from each other by using RC filters. A low pass filter, connected across the incoming sync pulse train will develop appropriate trigger pulses for synchronizing the vertical oscillator.

Similarly, a high pass filter will deliver sharp differentiated pulses for the horizontal oscillator from the same pulse train. This differentiated (Spiked) output is fed to the AFC

circuit. This output holds the horizontal oscillator at the correct frequency. The use of AFC circuit ensures correct synchronization even in the presence of noise pulses.

The actual movement of the electron beam in a picture tube is controlled by magnetic fields produced by the vertical and horizontal deflection coils. Hence proper vertical and horizontal driving voltages must first be produced by synchronized oscillators and associated wave shaping circuits. For vertical deflection, the frequency is 50HZ and the horizontal deflection is at 15,625Hz. The during waveforms thus generated are applied to power amplifiers which provide sufficient current to the deflection coils to produce full raster on the screen of the picture tube.

7. Picture Tube:

Picture tube is used to reproduce the original picture information or picture details.

8. Sound IF Amplifier:

The amplitude of the inter carrier sound signal at the output of the video detector is very low. So at least two stages of sound IF amplification is provided before feeding it to the FM detector. Each IF stage is a tuned amplifier with a centre frequency of 5.5MHZ.

The frequency modulated sound signal has some amplitude variations, both because of beat formation at the video detector and due to somewhat unequal amplification at the RF and IF stages of the receiver. This type of interference can be eliminated by using AM limiting circuits.

9. FM Sound Demodulator:

The FM sound demodulator recovers the original sound signal form the frequency modulated sound carrier signal. In general FM sound detectors may be classified as

- a. Foster-seeley discriminator
- b. Slope detector
- c. Ratio detector
- d. Quadrature detector
- e. Differential peak detector
- f. Phase-locked loop detector

10. Audio Amplifier:

This amplifier magnifies the signal to the required level, and to reproduce the original sound by using loudspeaker.

11. Loud Speaker:

Loud speaker is used to reproduce the original sound signal or sound information.

5.2.0 COLOR TELEVISION:

Three Color Theory:

In monochrome (Black and White) system, we are transmitting only the luminance (Brightness) information. But in color system, we have to transmit color information also in addition to brightness information. Regarding our eye, all optical information is separated into three colors by cones and they are combined at the optical centre of the brain to identify the original color.

The three basic colors called as primary color are,

- a. Red (R)
- b. Green (G) and
- c. Blue (B)

To get different color shading, we have to mix the primary color with different intensities. The two types of mixing used in general are,

- a. Additive mixing and
- b. Subtractive mixing

5.2.1. PRINCIPLES OF COLOR SIGNAL TRANSMISSION AND RECEPTION:

Three different types of color systems are used today. They are,

1. The American NTSC (National Television System Committee) system.
2. The German PAL (Phase Alternation by Line).
3. The French SECAM (Sequential Couleures A memory) system.

The 625 lines CCIR B monochrome system and PAL color system are adopted in India. In television the color system should be compatible with black and white system. Compatible implies that,

- i. The color television signal must reproduce a B/W picture on a monochrome receiver without any modification of the receiver circuitry, and
- ii. A color receiver must be able to produce a B/W picture from a normal monochrome signal (reverse compatibility). To achieve this color signal must meet the following requirement.
 1. It should occupy the same bandwidth as the corresponding monochrome signal.
 2. The location and spacing of picture and sound carrier frequency should remain the same.
 3. The color signal should have same luminance (Brightness) information as would a monochrome signal transmitting the same scene.

4. The composite color signal should contain color information together with the ancillary signals needed to allow this to be decoded.
5. The color information does not affect the picture reproduced on the screen of a monochrome receiver.
6. The system should employ the same deflection frequencies and sync signals as used for monochrome transmission and reception.

5.2.2. PAL COLOR TV RECEIVER:

A Color TV receiver may be regarded as conventional black and white receiver plus a chroma section and color picture tube. The block diagram of PAL color TV receiver is shown in the Fig.5.7. The various sections of this receiver are discussed below.

1. VHF and UHF Tuner:

VHF and UHF tuner is provided at the front of the TV receiver. It contains tuned circuit for selecting the desired channel. The signal received by the antennal is amplified and converted into IF signal, it is then applied to the video IF amplifier. Automatic frequency tuning facility is also adopted in tuners for proper reproduction of color.

2. Video IF Amplifier:

The strength of output signal derived from the tuner is not sufficient to drive the video detector, so it is amplified to the required level by using cascaded IF amplifiers.

3. Video Detector:

The video detector recovers the original video signal from the modulated colorplexed composite video signal.

4. Sound Section:

Sound section contains Trap Circuit, Limiter, FM Detector, Audio Amplifier, Power Amplifier and Loudspeaker. This stage first recovers the original sound signal from the modulated sound carrier signal. The output of the FM detector is properly amplified and applied to the loud speaker for reproduction of sound.

5. Automatic Gain Control Circuit:

The Automatic Gain Control (AGC) circuit maintains the output signal as constant amplitude irrespective of its input signal strength.

6. Deflection Circuit:

The sync separator recovers the H and V sync pulses. They are processed by proper oscillator and amplifiers. Finally applied to the V and H deflection coils for deflect the electron beams in V and H directions simultaneously.

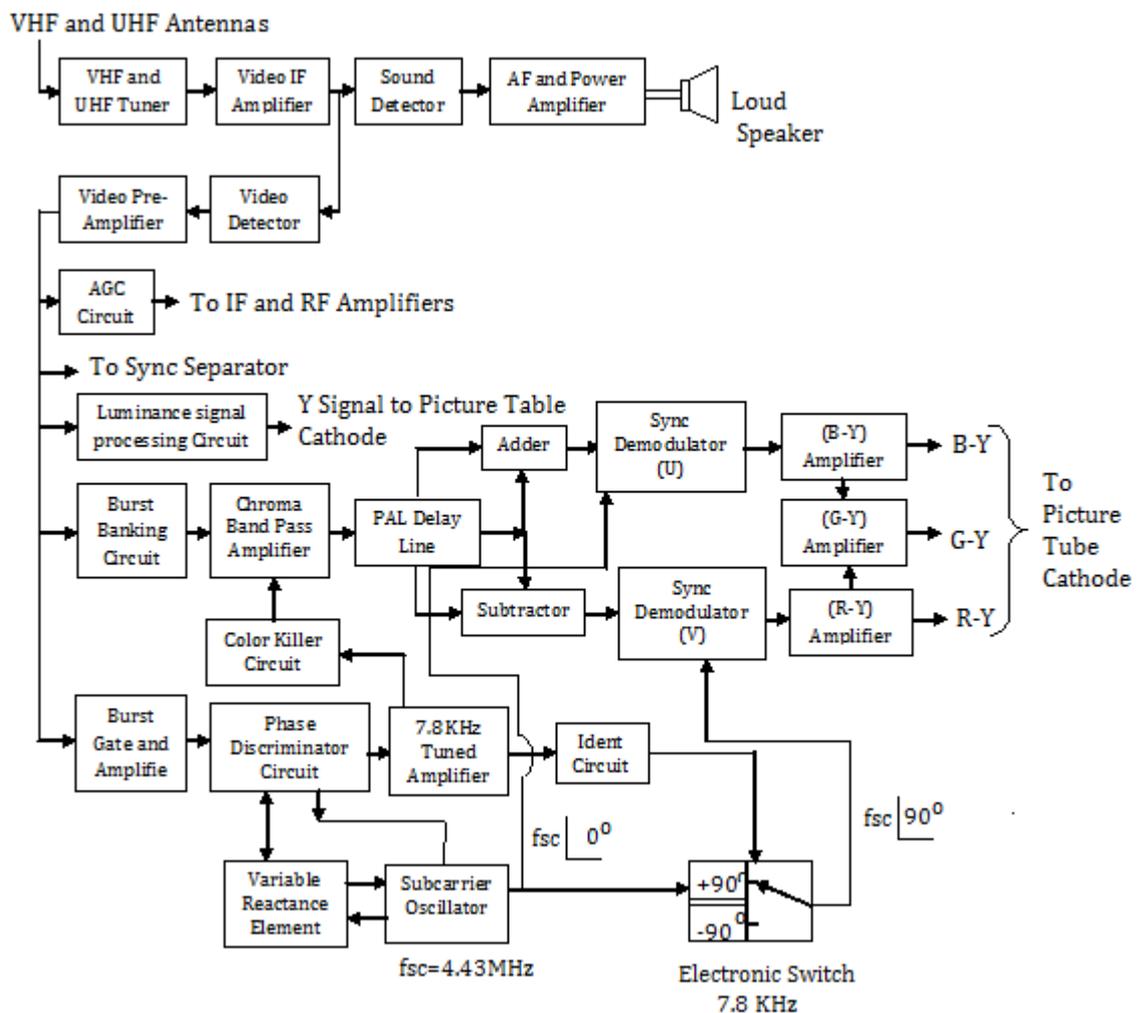


Fig.5.7. Block Diagram of PAL Color TV Transmitter

7. Luminance Signal:

The chrominance and luminance signal take separate paths after the video detector and later rejoin in the matrix section. The luminance signal processing network recovers the luminance (Y) signal from the composite video signal. The negative going Y signal (-Y) is usually applied to the cathodes of color picture tube.

8. Color Signal Processing:

The signal available at the output of video amplifier is properly amplified before feeding it to the various sections.

a). Chrominance Band Pass Amplifier:

The chrominance band pass amplifier selects the chrominance signal and rejects other unwanted components of the composite signal.

b). Burst Banking Circuit:

This circuit prevents the signal flow to the chrominance band pass amplifier during color burst intervals.

c).Burst Amplifier:

The burst gate amplifier separates the color burst signal from the chrominance signal, and also amplifies it to the required level. The frequency of this signal is 4.43 MHz.

d).Subcarrier Generation and Control:

The primary purpose of this section is to produce a subcarrier of correct frequency. Subcarrier oscillator is a crystal oscillator, used for producing subcarrier signal with the frequency of 4.43 MHz. The phase discriminator and variable reactance elements combine, to act as an Automatic Phase control circuit. It receives subcarrier signal and burst signal.

If the subcarrier oscillator frequency is exactly correct, its phase is arranged to be shifted by 90° with reference to the incoming burst signal. The output of APC circuit is applied to a 7.8 KHz tuned amplifier. This circuit takes off the 7.8 KHz ac component superimposed on the output signal. The output of this circuit is applied to the identification circuit and color killer circuit.

The identification circuit is used to control the electronic switch for alternately reversing the phase of subcarrier output before applying it to the V demodulator. When a color signal is being received, the 7.8 KHz component is available at the APC circuit of the reference subcarrier oscillator. Now the usual operation is done by chrominance band pass amplifier.

During monochrome signal reception, there is no 7.8 KHz. Therefore the color killer circuits switch off the chrominance band pass amplifier.

9. Separation of U and V Modulation Products:

This section contains PAL delay line circuit, adder, subtractor, V and U sync demodulators, difference signal amplifiers with matrix network. The chrominance signal, produced at the output of chroma band pass amplifier is applied to one of the inputs of adder and subtractor circuits. The same signal makes a delay by using PAL delay line, and is also applied to the other inputs of adder and subtractor circuits.

The output of adder is the U information. Similarly the output of subtractor is V information. The output of adder and subtractor consists of two independent double sideband, suppressed carrier RF signals. These signals are applied to U and V synchronous demodulators respectively. The color subcarrier signal from the reference oscillator is directly applied to the U sync demodulator.

Similarly the same signal through an electronic switch, for making an $+90^\circ$ or -90° line phase shifted signal is applied to V sync demodulator. The U demodulator recovers the original B-Y signal. Similarly the V demodulator recovers the original R-Y signal. The matrix

network combines the two signals and produce G-Y signal. These color difference signals are applied to the corresponding grids of color picture tube

5.2.3. COLOR CCD CAMERA:

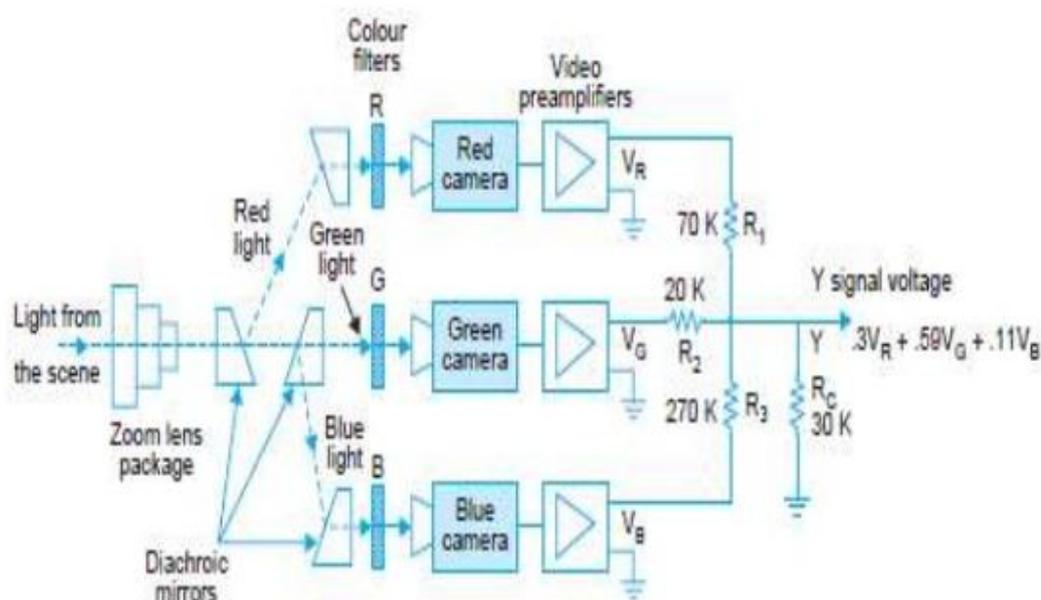


Fig.5.8. Color Camera Tube

A color TV camera consists of three camera tube, one for each three primary colors. In general, color TV camera is used to split the received optical images into the three primary colors. A group of mirrors (prisms) called as **diachronic mirrors** are used for this purpose. These mirrors are coated with a special material, that reflects a particular color light and allow other color to pass through it. That is, red diachronic mirror reflects red color and allows other colors to pass through it.

The arrangement of diachronic mirror, camera tubes and other components inside a color camera are shown in Fig.5.8.

The light from the scenes focused by the zoom lens package on the diachronic mirror setup. The mirrors split the light falling on it, into basic primary color light (red, green and blue). Then these lights are passed through the color filters (Trimming Filters) which can provide precise primary color images.

Then the filtered color lights are converted into video signals using three vidicon camera tubes. Each camera tube develops a video signal voltage, which is proportional to the color intensity received by it. Three video preamplifiers are used to amplify the signals coming out of camera tube. The output signals are referred as V_R (Red), V_B (Blue) and V_G (Green) signals.

A master deflection oscillator is used to simultaneously scan all the three camera tubes. As shown in the Fig.5.7, a resistance network is used to generate the luminance (Brightness) signal and color signal according to the Grassman's Law.

That is, the resistance values are selected such that luminance signal Y is given by,

$$Y=0.3 V_R + 0.59 V_G + 0.11 V_B$$

Instead of denoting each time as V_R, V_G and V_B , we simply denote the above terms as R, G and B. Thus,

$$Y=0.3 R + 0.59 G + 0.11 B$$

For example, if red color is focused on the lens package then we have only red signal.

$$Y = \frac{30}{(30+70)} V_R = 0.3 V_R$$

To achieve the compatibility condition we have to transmit the brightness information also along with the color signal information. That is why here luminance (Y) signal is generated in addition to the color signal.

5.3 LCD DISPLAY UNIT:

Liquid crystal displays (LCD) of small to large size have been developed for TV receivers in color as well as monochrome. LCDs in video display units are now available in varying sizes from a couple of inches to several inches, also in color. LCD TV is television sets that use LCD display technology to produce images.

An LCD screen essentially shines a backlight through a layer of liquid crystals when voltage is applied, it twist the varying amount of light pass through color filters, to eventually create an image on the screen. LCD televisions produce a black and colored image by selectively filtering a white light. The light is typically provided by series of cold cathode fluorescent lamps (CCFLs) at the back of the screen. Millions of individual LCD shutters are arranged in a grid. It will open and close to allow particular amount of white light through it.

Each shutter is paired with a colored filter to remove all but the red, green and blue (RGB) portion of light from the original white source. Each shutter filter pair forms a single sub-pixel. The sub-pixels are so small, the three sub pixels with red, green and blue color filters together to produce a single spot of color, called **Pixel**.

The shade of color is controlled by changing the relative intensity of the light passing through the sub pixels. A typical shutter assembly consists of a sandwich of several layers deposited on two thin glass sheets forming the front and back of the display. The rear sheet starts with polarizing film, the glass sheet, the active matrix components and addressing electrodes and then the director. The Fig.5.9.shows the general concept of LCD display unit.

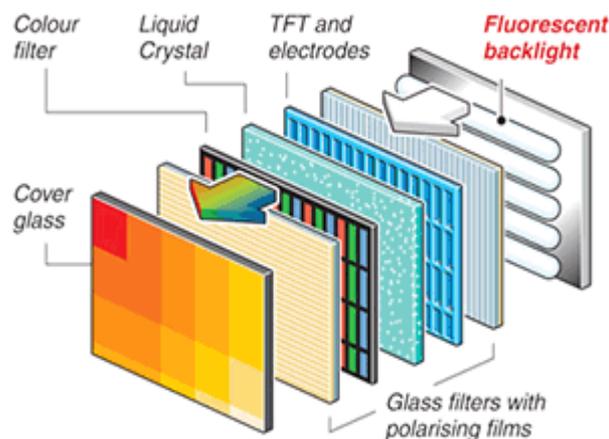


Fig.5.9. LCD Display Unit

The front sheet is similar but lacks the active matrix components, replacing those with the patterned color filters. The liquid crystal is placed between the two sheets in a patterned plastic sheet that divides the liquid into individual shutters. To produce complete television, the shutter assembly is combined with control electronics and back light.

The backlight can be provided by a single lamp, or number of separate lamps. A diffuser or frosted mirror is used to spread out the light. Each pixel consists of a column of liquid crystal molecules suspended between two transparent electrodes and two polarizing filters. The axes of polarizer are perpendicular to each other. Without the liquid crystals between them, light passing through one would be blocked by the other.

The liquid crystal twists the polarization of light entering one filter to allow it to pass through the other. Before applying an electrical charge the liquid crystal molecule are in a relaxed state. Charges on the molecule cause these molecules to align themselves in a helical structure or twist.

Light passing through one filter is rotated as it passes through the liquid crystal, allowing it to pass through the second polarized filter, and the assembly is transparent.

When an electric charge is applied to the electrodes, the molecules of the liquid crystal align themselves parallel to electric field, thus limiting the rotation of entering light. If the liquid crystals are completely untwisted, the light passing through them will be polarized perpendicular to the second filter, and thus completely blocked.

The pixel will appear unlit. By controlling the twist of the liquid crystal in each pixel, light can be allowed to pass through in varying amount, correspondingly illuminating the pixel. LCD screen enables good color reproduction and high contrast. These can be built in larger sizes than possible with CRT screens. Though large, they are light in weight and can be installed easily, even hung on a wall.

Advantages:

- Compactness.

- It is a thinner and light weight device.
- Low cost.
- It consists of some microwatts for display in comparison to some mill watts for LED's.
- It consumes less amount of power compared to CRT and LED.

Disadvantages:

- Low contrast and brightness.
- Require additional light sources.
- Range of temperature is limited for operation.
- Low reliability.
- Speed is very low.
- LCD's need an AC drives.

Applications of Liquid Crystal Display

- Liquid crystal thermometer.
- Optical imaging.
- The liquid crystal display technique is also applicable in visualization of the radio frequency waves in the waveguide.
- Used in the medical applications.

5.4 LED DISPLAY UNIT:

The LED TV more advanced version of LCD TV. While both type of TV on LCD technology and have flat-panel designs. LED TV's are slimmer, brighter, and more detailed than your traditional LCD TV. LED TV's function by illuminating LCD panels with LED backlighting. However, not all LED TV's utilize the same type of LED backlighting.

The LED television sets have defined the viewing experience in a whole new way and with best viewing. The LED screens are even lighter and thinner than LCD screen. The working principle of the LED TV's is not much different from the LCD televisions. The LED TV's can be considered to be a better modification of the LCD television sets. The LED TV's have minimized the drawbacks of the LCD TV's.

The fluorescent tubes in the LCD television were space consuming and heavy and sometimes they do not provide sufficient color quality. These drawbacks have been replaced by the LED television witch use an array of LED's to provide the backlight as shown in the Fig.5.10. LED televisions are more space saving and hence the LED TV's are thinner and lighter than the LCD televisions. The LEDs are also capable of displaying the purest black and white picture.

There are two types of LED TV's at present in the market. One is back lit, which receives the back lighting through the numerous LED's that re placed at the back of the television panel. The other one is edge lit which receives the light from the LED's that are placed along the edges of the television panel. The edge lit LED TV's are further thinner than the back lit ones. So edge lit LED TV's are also less power consuming and more efficient.

An LED TV illuminates its LCD panel with light emitting diodes. LEDs consist of small semiconductors, which glow during expose to electric current. When electric current is applied to small specific area of liquid crystal layer, the LED TV can alter the alignment of liquid crystal.

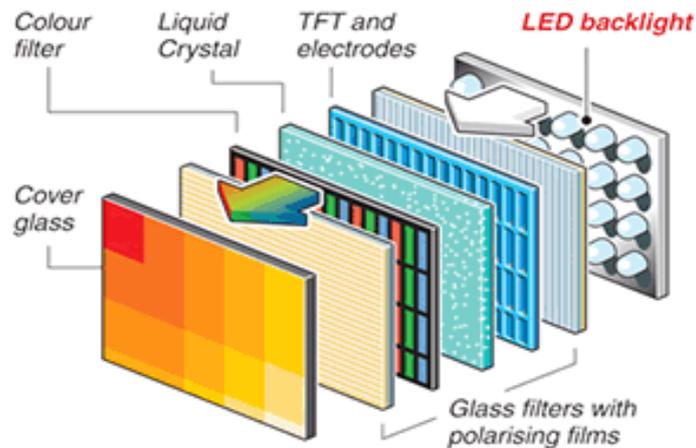


Fig.5.10. LED Television Screen

This current actually flows between LED anodes, which are positively charged electrodes, and LED cathodes, which are negatively, charged electrodes. In contrast, a traditional LCD TV utilizes fluorescent lamps for backlighting. These lamps function by using mercury vapory to create ultraviolet rays, which in turn cause the phosphor coating of the lamps to glow.

LED backlit LCD's are self illuminating, unlike pure LED systems, there are several methods of back lighting an LCD panel using LEDs, including the use of either white or RGB (Red , Green and Blue) Led arrays behind the panel, and edge Led lighting, which uses white LEDs arranged around the inside frame of the TV and a light diffusion panel to spread the light evenly behind the LCD panel.

Features of LED Display:

1. Lower power consumption
2. More balanced color saturation
3. Produces image with greater dynamic range
4. Thinner panel
5. Better heat dissipation
6. Brighter display
7. Better contrast
8. More available
9. Less environmental pollution

5.5 PLASMA DISPLAYS:

A Plasma display Panel is a type of flat panel display now commonly used for large TV displays. Many tiny cells located between two panels of glass hold an inert mixture of noble gases. The gas in the cell is electrically turned into plasma which then excites phosphors to emit light.

The normal gas atoms release electrons when they are stimulated. The loss of the negatively charged electrons means that the atoms become positively charged and the gas is said to be ionized. When this process has reached a certain level the gas has become plasma. Plasma display will glow when an electrical current is passed through a gas (Zenon or Neon). The gas is contained in hundreds of thousands of tiny cells positioned between two plates of glass.

Long electrodes are also sand-witched between the glasses plates on both sides f the cells. The address electrodes are behind the cells along the rear glass plate. The schematic diagram of plasma display is shown in the Fig.5.11.

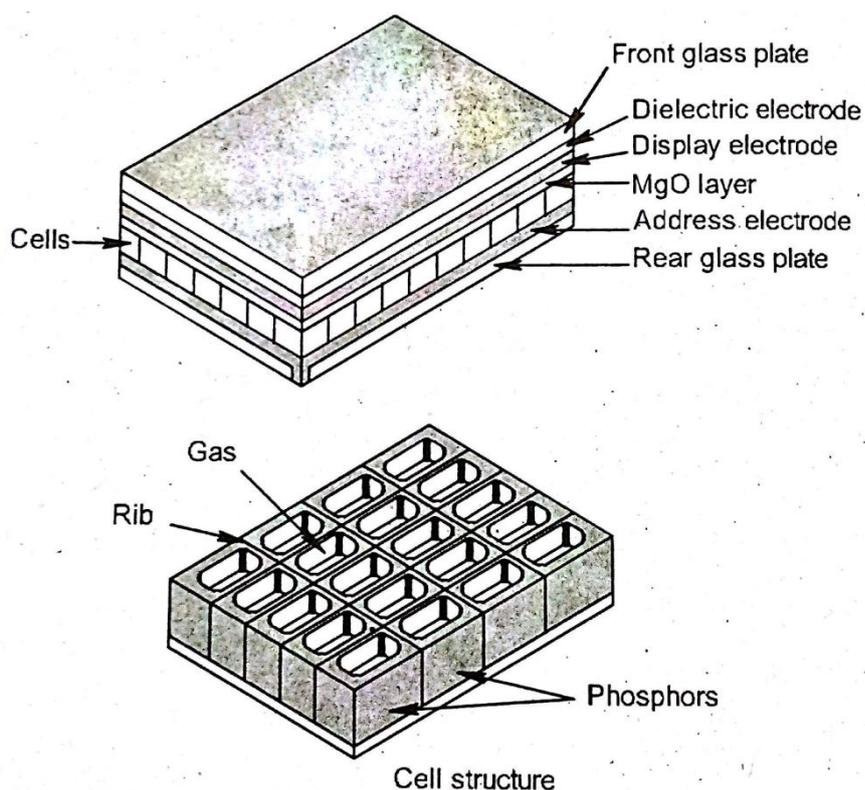


Fig.5.11. Construction of a Plasma Screen

The display electrodes are covered by a dielectric material and magnesium oxide (protective layer). This assembly is placed above the cell, along the front glass plate. Both sets of electrodes extend across the entire screen. The display electrons are arranged in horizontal rows along the screen and the address electrodes in vertical columns. The

horizontal and vertical electrodes from a basic grid. To ionize the gas in a particular cell, the plasma display's computer charges the electrodes that intersect at that cell.

When the intersecting electrodes are charged on application of small voltage, and electric current flows through the gas in the cell. The current creates a rapid flow of charged particles. They stimulate the gas atoms to release ultraviolet photons. The released ultraviolet photons interact with phosphor material coated on the inside wall of the cell. When an ultraviolet photon hits a phosphor atom in the cell, one of its electrons jumps to a higher energy level and the atom heats up.

When the electron falls back to its normal level, it releases energy in the form of visible light photon which illuminates the screen. The color phosphors in a plasma display give-off color light when they are excited. Each pixel is made up of three separate sub pixel cells with different color (R, G, and B) phosphors. Their colors blend to create overall color of the panel. By varying the pulse of current flowing through the cells, the control system can increase or decrease the intensity of each sub pixel color.

By this effect hundreds of different combinations of red, green and blue color are created. In this way the control system can produce colors across the entire visible spectrum. Plasma screens are thin and fragile type. They are heavy and consume a lot of power.

Advantages:

- Capable of producing superior contrast ratio.
- A very large, very thin screen can be produced.
- The image is very bright and has wider viewing angles.
- Less visible motion blurs.
- Fast response time.

Disadvantages:

- Consume more power.
- Shorter life span
- Image brightness range is narrower.
- Does not work at high altitudes.
- More expensive plasma compatible sensor must be used.

5.6 HANDY CAM:

A camcorder is a generic term for a portable device for recording of audio and video onto a storage device within it. The term camcorder is a combination of 'camera' and 'recorder' in one unit. It is also called as handy cam. Now a day's digital cam corders use mini DVD or DVD for storage.

It shoots and records motion and still pictures both. Handy cams create high quality movies or documents. The handy cam uses charge coupled devices (CCD) and an image

sensor. It converts the object light (image) collected through the lens system into a video signal.

The video signal is converted into the format used by the handy cam, then the image and sound data are recorded on to the videotape or memory chip. There are two types of handy cam, analog and digital. Analog handy cam record video signals on to a video tape as a continuous track of magnetic patterns. Digital handy cam converts analog information into digital form, recording video and audio in digital form.

Digital handy cam is expensive, because it has the following features.

1. Longer recording.
2. Offer 16 bit sound with higher resolution
3. Easy to edit and archive.

Digital video handy cam record images in digital mode and use smaller tapes. DVD handy cam record video digitally onto 8cm DVDs, which can hold an hour of video. Many handy cam use flash memory card in addition to the main recording media to save still shots and web quality video clips. The CCD sensor works as the brain of the handy cam.

It scans the object light and converts it into electric signals. The effectiveness of the CCD sensor depends upon the size of the CCD sensor, number of active pixel elements, technology used in CCD circuitry and number of CCDs. Most handy cam use a single CCD to scan and transmit the entire color information. One sensor handy cam converts the image collected by the lens system into three images, one for each basic color (Red, Green and Blue).

The resulting images are then converted into digital data by their respective CCDs and recorded on the digital video. This results in improved quality of recording. Three CCD handy cams capture more information and record color better and more accurately. The lens often the most expensive component for good image quality. The object information contains color and brightness information must go through the lens prior to processing. If the lens is not able to accurately direct light on to the CCD sensor, the recorded images will not be sharp.

Handycam with a higher signal to noise ratio produce better images in dim light. Many Handycam come with built-in light source for recording in dim conditions. The light can be automatically switched on in dark conditions. Many handycam incorporate a liquid crystal display screen and view finder LCD and view finder facilitate immediate play back.

A zoom lens package is also provided at the front side of the handycam. Optical zoom offers better image quality than high magnification digital zoom. Recording a fast moving target can lead to shaky images. Optical image stabilizers use motion detectors and special lens arrangement to stabilize the images.

Most handycam record interfaced video with odd even field's but with progressive scan mode. This produces high quality still images. The micro phone attached to the handycam is

used to pick up the sound signal. It is mounted in the front or top of the handycam. It is used for recording presentation or speeches.

5.7 CLOSED CIRCUIT TELEVISION (CCTV) SYSTEM:

Closed-circuit television (CCTV), also known as video surveillance, is the use of video cameras to transmit a signal to a specific place, on a limited set of monitors. In CCTV, the video signal at the camera output is directly fed to a monitor at a remote location by means of a cable. Since the video signal is not modulated and is to be transmitted through a cable.

The television monitor is basically a TV receiver without RF and IF sections. All professional studio cameras have a view finder. It is used to enable the camera operator to have a real time observation of what he is viewing. In a typical CCTV set up, one video camera may feed more than one monitor. Projection TVs with large screens can also be used in place of monitors.

The schematic diagram of CCTV is shown in the Fig.5.12. CCTV is used in large number of applications like education, industry, medicine, traffic control, and business and so on. One of the applications of CCTV technology is in the concept of one teacher can teach too many class rooms.

A lecture delivered before cameras could be viewed in a number of class rooms on different monitors or projection TVs. CCTV finds with applications at Air ports, Railway stations, harbors etc. for public announcements like arrival and departure time. CCTV network ensures remote observation of patients in a ward at the central nursing station.

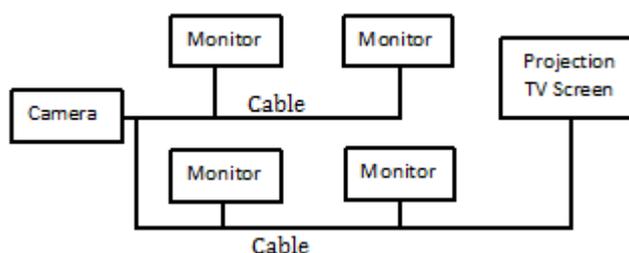


Fig.5.12 Schematic Diagram of CCTV

In another application, in surgery, all phases of an operation can be transmitted in LIVE to large group of medical researchers outside the operation theatre. It can be used as a surveillance tool in stores, banks and other important places. It can be used as a door monitor, baby sitter or as an observer of a sick person in bed. In Industry it is used for remote inspection of materials.

Application:

- Medicine
- Education
- Business

- Industry
- Aerospace and oceanography
- Home
- Surveillance

5.8 CABLE TELEVISION (CABLE TV):

Cable television is a special form of CCTV in which TV signal on standard channels are provided to viewers through coaxial cables on payment of fixed monthly charges. When a common antennal system is used to deliver a strong signal through coaxial cable to every TV set connected to the system it is called Master Antenna TV (MATV). Fig.5.13. shows the block diagram of cable television system.

The cable television system is used for distributing high quality TV signals to a very large number of users. This system feeds increased TV programmed to subscribers who pay a fee for this service. The cable system may have many more active (VHF and UHF) channels than a receiver can directly select. This requires use of a special type active converter in the head end.

The main signal source of cable TV is from various satellites. High gain parabolic dish antennas are used for receiving satellite signals. The down-link signals from most communication satellites are in C-band (3700 MHz- 4200 MHz) frequency range.

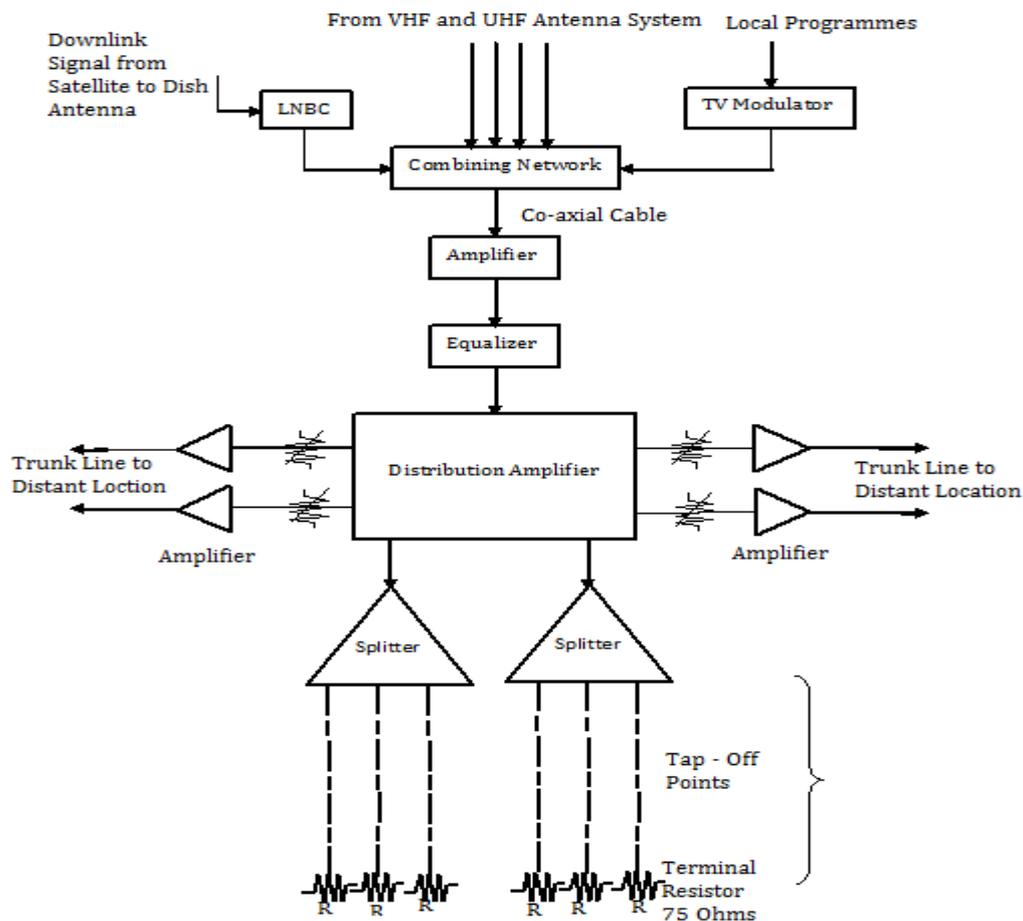


Fig.5.13. Block Diagram of Cable Television

The signals received by dish antennas are first converted into a lower frequency by using low noise block converter (LNBC). The VHF and UHF terrestrial broadcast signals are received by using convention antennas, mounted on high rise buildings. Events like local sports and cultural programmers can be distributed by using cable TV network. Programs like popular movies, play songs earlier recorded are also distributed through cable network.

The signals from, various TV channels are applied the combining network. The signal received by the dish antenna is converted into low frequency signals by using LNB converter. Similarly the signal received by UHF antenna is converted into low frequency signals by using translator. The combining network combines all the signals, and is allotted to separate carrier frequency for each channel.

The outputs from combining network are fed to a number of trunk cables through a broadband distribution amplifier. The function of distribution amplifier is to raise the signal amplitude to higher level for overcoming the losses occurring in the distribution system. The output from the distribution amplifier is fed to splitter through co-axial trunk lines. The trunk cables carry signals to the utilization site (s) which may be several kilometers away. Feeder amplifiers are provided at several points along the line to overcome signal attenuation which occurs due to cable loss.

The signal attenuation which occurs due to cable loss. The signal distribution from splitters to tap-off points is done through multi core coaxial cables. A splitter is a resistive-inductive device which provides trunk line isolation and impedance matching. The subscriber taps can be either transformer coupled, capacitive coupled, or in the form of resistive pads.

They provide isolation between receivers on the same line thus preventing mutual interference. The taps are normally mounted on the wall. Wall taps may be obtained with 300 Ω , 75 Ω output or dual output. The improperly terminated lines will develop standing waves. To avoid standing waves each 75 Ω distribution cable is terminated with a 75 Ω resistor called terminator.

REVIEW QUESTION AND ANSWER

PART A

(TWO MARKS)

1. What is TV?

TELEVISION means to see from distance. Its aim is to extend the sense of sight beyond its normal limit along with the sound and scene being televised.

2. What is the Use of Camera Tube?

The camera tube converts light input to corresponding electrical variations for the video signal.

3. What is Scanning?

Scanning is a process, in which it converts the charge image produced inside the TV camera Tube into a variable electrical signal.

4. What is Horizontal Scanning? Give its Frequency Range.

The deflection of electron beam across the scene with a continuous uniform motion for the trace from left to right. Frequency range is 15625Hz.

5. What is Vertical Scanning? Give its Frequency Range.

The saw tooth current applied to the vertical deflection coils moves the electron beam from top to bottom of raster at a uniform speed, while the electron beam is being deflected horizontally.

6. What is Flicker?

The scanning rate of 25 frames per second in television picture is not rapid enough to allow the brightness of one picture of frame to blend smoothly into the next. This alternative bright & dark in a picture is known as flicker.

7. What is Interlaced Scanning?

The total number of lines divided into two fields called odd & even field. Each field is scanned alternatively known as interlaced scanning.

8. What is the Purpose of Blanking Pulse?

The purpose of blanking pulses is to make invisible the retraces required in scanning.

9. What is Horizontal & Vertical Blanking Pulse?

- Horizontal blanking pulses are used to blank out the horizontal retrace from right to left for each line.
- Vertical blanking pulses are used to blank out the vertical retrace from bottom to top for each field.

10. What is Synchronizing Pulses?

The synchronizing pulses are transmitted as part of complete picture signal for the receiver, but they occur during the blanking time when no picture information is transmitted.

11. What is Horizontal & Vertical Synchronizing Pulses?

- Horizontal synchronization at the end of each line determines the start of horizontal retrace.
- Vertical synchronization at the end of each field determines the start of vertical retrace.

12. Define Aspect Ratio.

The width to height ratio of the picture frame is called the aspect ratio.

13. What is the Average Brightness?

Average brightness means, the average value of brightness variation of horizontal lines in one frame.

14. What is Pedestal Height & DC Level?

- The distance between average brightness & blanking level is called **pedestal height**.
- The distance between zero level & the average brightness is called **DC level**.

15. What are the TV Systems Used in Broadcasting?

- FCC (Federal Communication Commission)
- NTSC (National Television Committee)
- CCIR (International Television Consultative Committee)
- PAL (Phase Alteration with Line)
- SECAM (Sequential Couleures A Memory)

16. What is Video Processing Unit?

The video signal input of 1 Vpp is processed in a stabilizing amplifier & sync regenerator to minimize the effects of hum & noise, sync compression & other distortions, to correct it to the standard forms.

17. What is Audio Processing Unit?

The audio signal is given a pre-emphasis by a high pass network with the time constant of 50µs. The signal is amplified to make the signal at the standard level.

18. What is Diplexer?

It combines both the outputs of visual & aural transmitters. It maintains sufficient isolation between the visual & aural inputs.

19. What is Sound Trap Circuit?

The video amplifier usually has a trap circuit, tuned to the inter carrier sound frequency of 5.5MHz to keep the signal out of picture signal. It removes the sound signal from the carrier signal.

20. Classify FM Sound Detectors.

- Foster seeley discriminator
- Slope detector & Ratio detector
- Quadrature detector
- Differential detector
- Phase locked loop detector

21. What is the Use of Color Camera?

The colour camera is used to develop signal voltages proportional to the intensity of each primary color light.

22. What is the Luminance Signal?

The luminance signal processing network recovers the luminance signal from the composite video signal.

23. What are the Features of LED Display?

- Lower power consumption
- More balanced color saturation

- Thinner & better contrast
- Brighter display & more reliable

24. What are the Advantages of Plasma Displays?

- Wider viewing angles & Less visible motion blur
- Fast response time & Capable of producing superior contrast ratio.

25. What are the Disadvantages of Plasma Displays?

- Consume more Power
- Does not work at high altitudes
- More expensive plasma compatible sensor must be used.

26. What is Handy Cam?

Handy cam is small model of camcorder, handle with single hand. It is the combination of camera cum recorder.

27. What are the Features of Handycam?

- Longer recording
- Offer 16 bit sound with higher resolution
- Easy to edit & archive.

28. What is CCTV?

CCTV is used in large number of applications like education, industry, machine, traffic control, business & so on.

29. What is Cable TV?

The cable television system is used for distributing high quality TV signals to a very large number of users. This system feed increased TV programs to subscribers who pay a fee for this service.

30. What is Burst Blanking Circuit?

The circuit prevents the signal flow to the chrominance band pass amplifier during color burst intervals.

31. What is the use of Chrominance Band Pass Amplifier?

The chrominance band pass amplifier selects the chrominance signal & rejects other unwanted components of the signal.

32. What is the Use of Burst Amplifier?

The burst gate amplifier separates the color burst signal from the chrominance signal & amplifies it to the required level. The frequency of this signal is 4.43MHz.

PART B

(THREE MARKS)

1. What is scanning? And its types.
2. Explain horizontal scanning.

3. Explain vertical scanning.
4. Explain Flicker .
5. Explain interlaced scanning.
6. Explain blanking Pulse.
7. Explain synchronization pulse with Horizontal and vertical.
8. Draw composite Video Signal.
9. Explain TV broadcasting Standards.
10. Explain LCD display Unit.
11. Explain LED display Unit.
12. Explain handycam.
13. Draw Cable Television.

PART C

(TEN MARKS)

1. What is scanning? Explain a) Horizontal scanning 2) Vertical scanning 3) Flicker 4) interlaced scanning.
2. Explain blanking Pulse and synchronization pulse with Horizontal and vertical.
3. Draw and explain composite Video Signal.
4. Explain TV broadcasting Standards
5. Draw and explain Block diagram of TV Transmitter.
6. Draw and explain Block diagram of TV Receiver.
7. Explain the Principle of color Television.
8. Draw and explain Color CCD Camera.
9. Draw and explain Block diagram of PAL Color TV Receiver.
10. Explain 1. LCD display Unit 2. LED display Unit
11. Draw and explain Plasma Display with advantages and disadvantages.
12. Explain handycam
13. Draw and explain Closed Circuit Television (CCTV).
14. Draw and explain Cable Television.