

## Learning Objectives

After completing this topic, the student will be able to understand

- The concept of communication system
- The basic idea of modulation
- The need for modulation
- The radio frequency spectrum
- The different types of modulation
- The analysis of AM and FM waves.
- The relation between modulated and unmodulated powers in AM signal
- The measurement of modulation index with CRO
- The advantages and disadvantages of FM over AM
- The pre-emphasis and de-emphasis in an FM system

## 1.1 Introduction

Communication is the process of sending information from one point to another point. Technically speaking, the term communication refers to the sending, receiving and processing of information by electronic means. In recent years, communication has become more widespread with the use of satellites and fiber optics today there is an increasing emphasis on the use of computer on communication typical examples of communication system are line telephony and telegraphy, radio telephony and telegraphy, radio broadcasting, point to point and mobile communication (civil or military), computer communication, radar, television broadcasting radio telemetry, radio aids to navigation, radio aids to aircraft landing, ship to shore communication, satellite communication etc.

## 1.2 Block diagram of communication system

### Information Source

Any communication system serves to communicate a message or information. This message originates in the information source. In general, there may be several messages in the form of words, group of words, code symbols etc. out of these messages only the desired message is to be selected and conveyed. The set of messages consists of various messages distinguishable from one another these messages may be in the form of words, groups of words

Next this electrical signal is processed to restrict its range of audio frequencies the signal or message in a radio transmitter is amplified in several stages of small signal amplifiers (voltage amplifiers) and large signal amplifiers ( power amplifiers ) and may be possibly encoded, to make it suitable for transmission and subsequent reception. Finally, the signal or information amplitude modulates the carrier (high frequency sine wave). The actual modulation system differs from system to system, thus the modulation may be done at high carrier level or low carrier level. However, a system may use amplitude modulation, frequency modulation, phase modulation or any variation or combination of these depending in the actual requirement.

### **The channel and the noise source.**

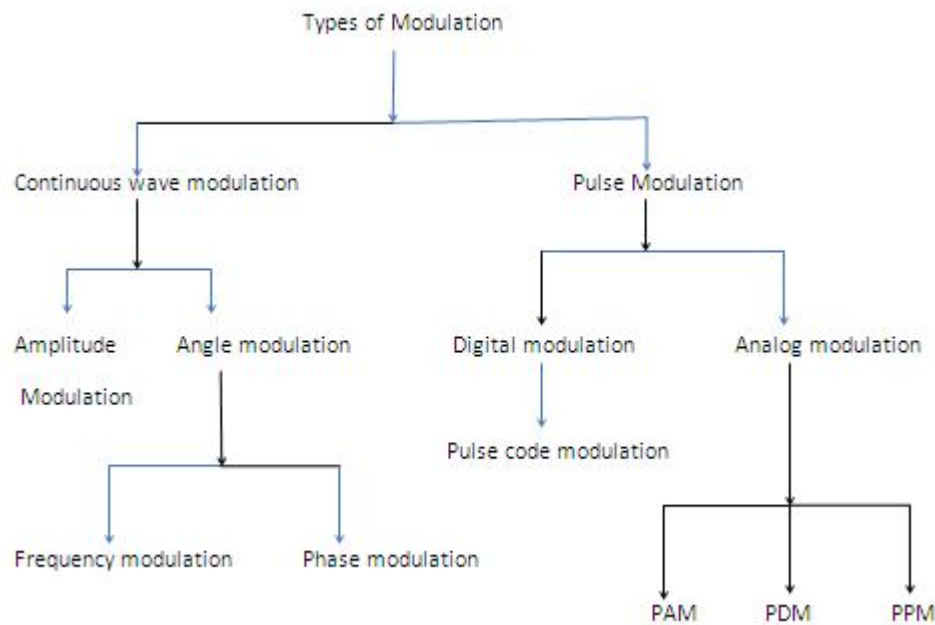
The channel is basically a medium which electrically connects the transmitter to the receiver. It may be a pair of wires, a coaxial cable, free space, optical fiber or even a laser beam. The properties of the channel can strongly influence the performance of a communication system. It may, however, be noted that the term channel is also often used to indicate the frequency range allocated to a given service or transmission. Thus a television channel occupies a band width of 7MHz while an amplitude modulation broadcast channel occupies a band width of 10KHz.

During the process of transmission and reception, the signal gets distorted due to (i) distortion in the system and (ii) noise introduced in the system. The noise introduced is an unwanted energy, usually of a random character and may be caused by various sources. The noise gets superimposed on the signal, with severe noise, the signal to noise power ratio becomes so poor that the signal becomes unintelligible and hence useless the noise in the channel or at the input to the receiver is most effective in deteriorating the signal to noise ratio and it must be minimized to the maximum possible extent

### **The Receiver**

Most of the radio receiver are of the super heterodyne type, the block diagram of which is shown in Fig 1.3.

The main function of the receiver is to reproduce the original message from the distorted signal available at the input of it in a simple superheterodyne broadcast receiver shown in Fig 1.3, the voltage induced in the receiving antenna is fed to RF amplifier. The amplified carrier voltage from the RF amplifier and the local oscillator voltage are heterodyned or mixed in the mixer stage resulting in intermediate frequency (I.F) voltage. This IF voltage is amplified in IF amplifier and then fed to the detector, at the output of which we get the original signal voltage. This audio voltage is amplified and fed to the loud speaker



**Fig. 1.4 Modulation types**

## 1.4 Amplitude Modulation

The message signal generated from the information source is known as baseband signal. This baseband may be a combination of two or more message signals. If the baseband signal is transmitted directly, then it is known as baseband transmission. This means that the baseband transmission does not use modulators and demodulators the baseband signal may be analog or digital. The baseband transmission is preferred at low frequencies and for short distances.

If the modulated signal is transmitted over the channel, it is known as band pass or simply pass band transmission whenever a modulating signal is impressed upon a carrier, the modulated signal is produced. This modulated signal has fixed band of frequencies around carrier frequency. In fact, the nature of such a signal is band pass type, therefore, modulated signals are known as pass band or band pass signals. The band pass transmission is generally used at high frequencies and for long distance communication such as radio and television.

In amplitude modulation, the amplitude of the carrier voltage is varied in accordance with the instantaneous value of modulation voltage the frequency and phase angle of the carrier voltage remains unaltered.

and demodulators the base band signal may be analog or digital. The base band transmission is preferred at low frequencies and for short distances.

If the modulated signal is transmitted over the channel, it is known as band pass or simply pass band transmission whenever a modulating signal is impressed upon a carrier, the modulated signal is produced. This modulated signal has fixed band of frequencies around carrier frequency. In fact, the nature of such a signal is band pass type, therefore, modulated signals are known as pass band or band pass signals. The band pass transmission is generally used at high frequencies and for long distance communication such as radio and television.

**Table 1.1.** Standard classification of frequency spectrum

Carrier Frequency	Class	Service
10.30 kHz	Very low frequency (VLF)	Long distance point to point communication.
30.300kHz	Low frequency (LF)	Long distance point to point communication, navigation
300.3000kHz	Medium frequency (MF)	Broad casting, ship to shore communication
3-30 MHz	High frequency (Hf)	National and international broad cast point to point telephone and telegraph communication, Aviation
30-300 MHz	Very high frequency (VHF)	Television, radar, FM broad cast, short distance communication.
300-300 MHz	Ultra high frequency (UHF)	Facsimile, television relay, Air navigation
3-30 GHz	Super high frequency (SHF)	Radar navigation radio relay.

## 1.6 Modulation index in AM signal

Let the modulating voltage be given by the expression

$$V_m = V_m \cos \omega_m t$$

Where  $\omega_m$  is the angular frequency of the signal and  $V_m$  is the amplitude  
let the carrier voltage be given by the expression

$$m = \frac{V_{c \text{ max}} - V_{c \text{ min}}}{V_{c \text{ max}} + V_{c \text{ min}}}$$

The total power in the modulated wave may be expressed as

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

$$= \frac{V_c^2}{R} + \frac{V_{LSB}^2}{R} + \frac{V_{USB}^2}{R}$$

Where all the three voltages are r.m.s. values and R is resistance in which the power is dissipated.

$$\begin{aligned} P_t &= \frac{(V_c/\sqrt{2})^2}{R} + \left(\frac{mV_c}{2}/\sqrt{2}\right)^2 \frac{1}{R} + \left(\frac{mV_c}{2}/\sqrt{2}\right)^2 \frac{1}{R} \\ &= \frac{V_c^2}{2R} + \frac{m^2 V_c^2}{8R} + \frac{m^2 V_c^2}{8R} \\ &= \frac{V_c^2}{2R} \left(1 + \frac{m^2}{4} + \frac{m^2}{4}\right) \\ &= \frac{V_c^2}{2R} \left(1 + \frac{m^2}{2}\right) \\ &= P_c \left(1 + \frac{m^2}{2}\right) \\ P_t &= P_c \left(1 + \frac{m^2}{2}\right) \end{aligned}$$

**Example 1.1.** A 1000 kHz carrier signal is amplitude modulated by a 10 kHz modulating signal. Determine the values of lower and upper side band frequencies.

**Solution.** Carrier frequency,  $f_c = 1000 \text{ kHz}$

Modulating signal,  $f_m = 10 \text{ kHz}$

Lower sideband frequency,  $f_{lsb} = f_c - f_m$

$$= 1000 - 10$$

$$= 990 \text{ kHz}$$

**Example 1.4**

A 5kw carrier  $\left(1 + \frac{0.7^2}{2}\right)$  modulated to a depth of 70% calculate the total power in the modulated wave.

**Solution** carrier power,  $P_c = 5 \text{ kw}$

Depth of modulation,  $m = 70\% = 0.7$

$$\begin{aligned} \text{Total power, } P_t &= P_c \left(1 + \frac{m^2}{2}\right) \\ &= 5 \left(1 + \frac{0.7^2}{2}\right) \\ &= 5 \left(1 + \frac{0.49}{2}\right) \\ &= 5(1+0.245) \\ &= 5(1.245) \\ &= 6.225 \text{ KW} \end{aligned}$$

**Example 1.5**

A broadcast radio transmitter radiates 10kw when the modulation percentage is 60. What is the unmodulated carrier power.

**Solution.** Total power,  $P_t = 10 \text{ Kw}$

Depth of modulation,  $m = 60\% = 0.6$

Unmodulated carrier power,  $P_c = ?$

$$\begin{aligned} \text{We know that, } P_t &= P_c \left(1 + \frac{m^2}{2}\right) \\ 10 &= P_c \left(1 + \frac{0.6^2}{2}\right) \\ &= P_c (1+0.18) \\ &= 1.18 P_c \\ P_c &= 8.475 \text{ KW} \end{aligned}$$

**1.8. Bandwidth and communication system**

Bandwidth is the difference between the upper and lower frequencies in a continuous set of frequencies. It is typically measured in hertz, and may sometimes refer to passband bandwidth, sometimes to baseband bandwidth, depending on context. Passband bandwidth is the difference between the upper and lower cutoff frequencies of, for example, a bandpass filter, a communication channel, or a signal spectrum. In case of a low-pass filter or baseband signal, the bandwidth is equal to its upper cutoff frequency.

Modification in phase according to low frequency will give phase modulation.

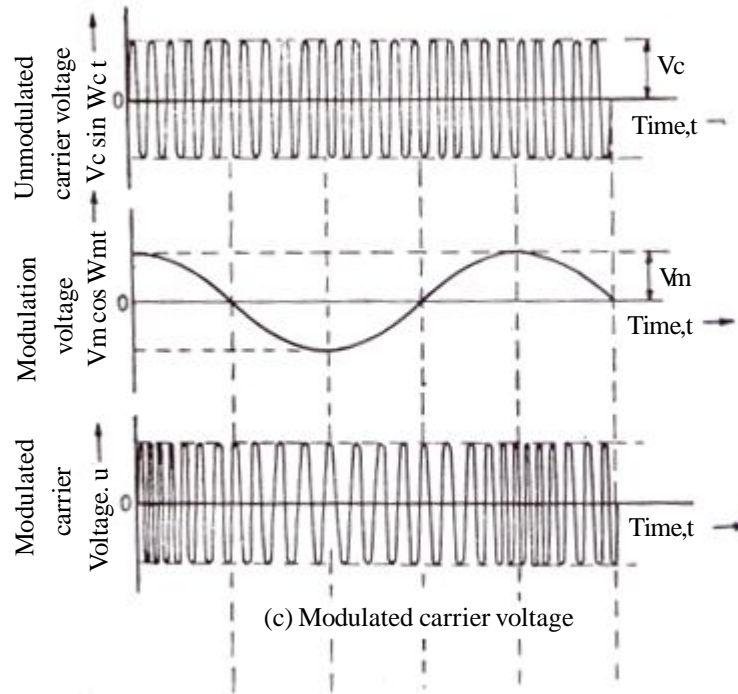


Fig. 1.6 Angle Modulation

### 1.11 Comparison between amplitude and angle modulation

1. While the amplitude modulation always has only two sidebands with or without the carrier depending on whether it is suppressed or transmitted carrier type, the frequency modulated spectrum contains an infinite number of sidebands together with the carrier, separated by the fundamental modulating frequency.

2. Amplitude modulation is a linear process where superposition principle holds. In contrast, the FM is a non-linear process. The spectrum of a frequency-modulated signal, modulated by two or more modulating signals, is not the same as that obtained by the mere summation of the spectra of the carriers frequency modulated individually by the components of the composite modulating signal.

3. Since each pair of the sidebands is preceded by the Bessel coefficients that are functions of the modulation index, hence they fluctuate on either side of zero and gradually diminish with increasing value of the modulation index. Higher the value of  $m_f$  larger will be the number of significant sidebands. The bandwidth requirement depends on the value of the modulation index and not on the

(ii) The sidebands at equal frequency intervals from  $f_c$  have equal amplitudes. Thus the side band distribution is symmetrical about the carrier frequency.

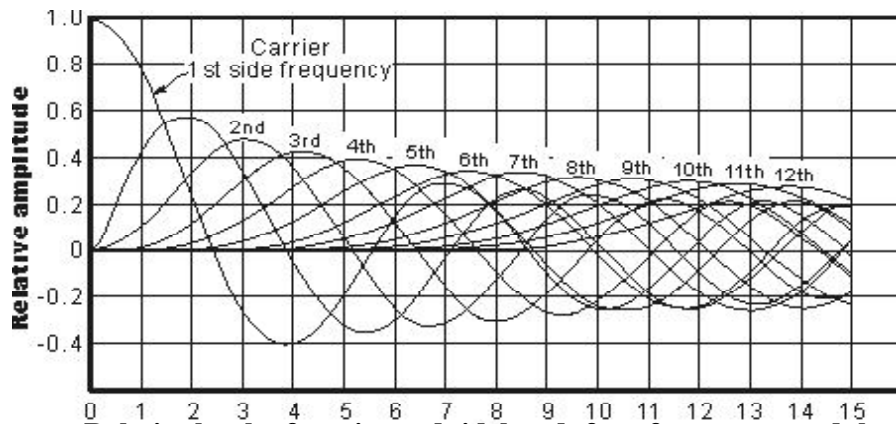
(iii) In FM, the total transmitted power remains constant

### 1.13 Calculations - bandwidth, side band frequencies

Any signal that is modulated produces sidebands. In the case of an amplitude modulated signal they are easy to determine, but for frequency modulation the situation is not quite as straightforward. They are dependent upon not only the deviation, but also the level of deviation, i.e. the modulation index  $M$ . The total spectrum is an infinite series of discrete spectral components expressed by a complex formula using Bessel functions of the first kind.

$$\begin{aligned} \text{Spectrum} = & V_c \{ J_0(M) \cos \omega_c t + \\ & J_1(M) [\cos (\omega_c + \omega_m)t - \cos (\omega_c - \omega_m)t] + \\ & J_2(M) [\cos (\omega_c + 2\omega_m)t - \cos (\omega_c - 2\omega_m)t] + \\ & J_3(M) [\cos (\omega_c + 3\omega_m)t - \cos (\omega_c - 3\omega_m)t] + \\ & \dots \dots \dots \} \end{aligned}$$

The total spectrum can be seen to consist of the carrier plus an infinite number of sidebands spreading out on either side of the carrier at integral multiples of the modulating frequency. The relative levels of the sidebands can be obtained by referring to a table of Bessel functions. It can be seen from the image below that the relative levels rise and fall according to the different values of modulation index.



Relative levels of carrier and sidebands for a frequency modulated signal



spreading out over large amounts of the frequency spectrum. Usually it is necessary to limit the bandwidth of a signal so that it does not unduly interfere with stations either side.

As a frequency modulated signal has sidebands that extend out to infinity, it is normal accepted practice to determine the bandwidth as that which contains approximately 98% of the signal power.

A rule of thumb, often termed Carsons' Rule states that 98% of the signal power is contained within a bandwidth equal to the deviation frequency, plus the modulation frequency doubled, i.e.:

$$BT = 2 (f + fm)$$

Normally the bandwidth of a wideband FM signal is limited to the Carson's Rule limit - this reduces interference and does not introduce any undue distortion of the signal. In other words for a VHF FM broadcast station this must be  $(2 \times 75) + 15$  kHz, i.e. 175 kHz. In view of this a total of 200 kHz is usually allowed, enabling stations to have a small guard band and their centre frequencies on integral numbers of 100 kHz.

There are a few interesting points of summary relative to frequency modulation bandwidth:

- The bandwidth of a frequency modulated signal varies with both deviation and modulating frequency.
- Increasing modulating frequency reduces modulation index - it reduces the number of sidebands with significant amplitude and hence the bandwidth.
- Increasing modulating frequency increases the frequency separation between sidebands.
- The frequency modulation bandwidth increases with modulation frequency but it is not directly proportional to it.

### 1.14 Modulation index for FM signal

This ratio of frequency deviation to frequency of the modulating signal is useful because it also describes the ratio of amplitude to tone for the audio signal. These factors determine the number and spacing of the side frequencies of the transmitted signal. The modulation index formula is shown below:

$$\text{modulation index} = \frac{\Delta f}{f_m}$$

Where:

$\Delta f$  = frequency deviation

$f_m$  = modulating frequency

## 1.15 Merits and Demerits of FM over AM

### Merits of FM

- (i) The amplitude of the frequency modulated wave remains unaffected.
- (ii) In FM, there is a large decrease in noise and hence increase in S/N ratio
- (iii) In FM, frequency allocation allows for a guard band. This reduces adjacent channel interference.
- (iv) In FM, noise may be further reduced by increasing deviation
- (v) In the VHF bands where FM operates, there is less noise than in the HF or MF bands
- (vi) FM permits use of several independent transmitters on the same frequency with negligible interference demerits of FM

### Demerits

- (i) A much wider channel, typically 200 KHz is needed
- (ii) Transmitting and receiving equipment's are complex and costly.
- (iii) Reception using conventional method is restricted to line of sight

### Summary

1. Communication is the process of sending information from one point to another point. Typical examples are radio communication, television communication etc.

2. The constituent parts of a communication system are information source, transmitter channel and the Receiver.

3. Modulation is a process in which some characteristic such as amplitude, frequency or phase of a high frequency voltage called the carrier voltage in accordance with the instantaneous value of another low frequency voltage called the modulating voltage.

4. Modulation is necessary to

- (a) Reduce the transmitting antenna height
- (b) For effective radiation of signal and to operate several radio stations.

**Short Answer Type Questions**

1. Define the term modulation.
2. What is meant by amplitude modulation.
3. What is meant by the term frequency modulation ?
4. Define the term modulation index.
5. Write expression for the amplitude modulated voltage.
6. Give the mathematical expression for the frequency modulated voltage.
7. What is angle modulation ?
8. What is the need for modulation ?
9. What are based band and pass band signals in a communication system ?
10. What is the relation between total power (modulated) and carrier power (unmodulated) in an amplitude modulated signal ?
11. Define (a) Frequency deviation and (b) Modulation index in FM.
12. Define the terms frequency modulation.
13. Define the terms phase modulation.
14. Sketch the AM and FM waves.
15. Mention the values of AM and FM bandwidths.
16. Write the values of the maximum modulation signal frequencies in AM and FM.
17. What are significant sidebands in FM ?

**Long Answer Type Questions**

1. Prove that after amplitude modulation, the carrier power increases from  $P_c$  to  $P_c (1+m^2/2)$  where  $m$  is the modulation index.
2. Write the advantages and disadvantages of FM over AM.
3. Draw the block diagram of a communication system and explain it in detail.
4. Explain the procedure to calculate the value of modulation index of an AM wave with CRO.



## UNIT

## 2

**Transmitters & Basics of  
Digital Communication****Structure**

- 2.1 Introduction
- 2.2 Specifications of transmitter
- 2.3 Distinguish between high level and low level modulation
- 2.4 Block diagram of a high level AM transmitter
- 2.5 Block diagram of a low level AM transmitter
- 2.6 Digital Communication
- 2.7 Comparison between analog and digital communication
- 2.8 Pulse code modulation
- 2.9 Delta modulation
- 2.10 Basic types of digital modulation
- 2.11 Need for digital modulation
- 2.12 ASK, FSK and PSK

**Learning objective**

After completing this unit, the student will be able to explain

- The different types of transmitters
- The different between low level and high level modulation systems.

(d) VHF and UHF transmitters

(e) Microwave transmitters

### 2.3. High Level and low level modulation

In high power level modulation system, the carrier voltage is modulated at the highest power level. The required power level is obtained by using class C amplifiers.

In low power level modulation system, the carrier is modulated at low power level and the carrier power is subsequently raised to the desired level in class B Tuned power amplifiers. Transmitters using high power level are widely used at present.

### 2.4 Block diagram of high level AM transmitter

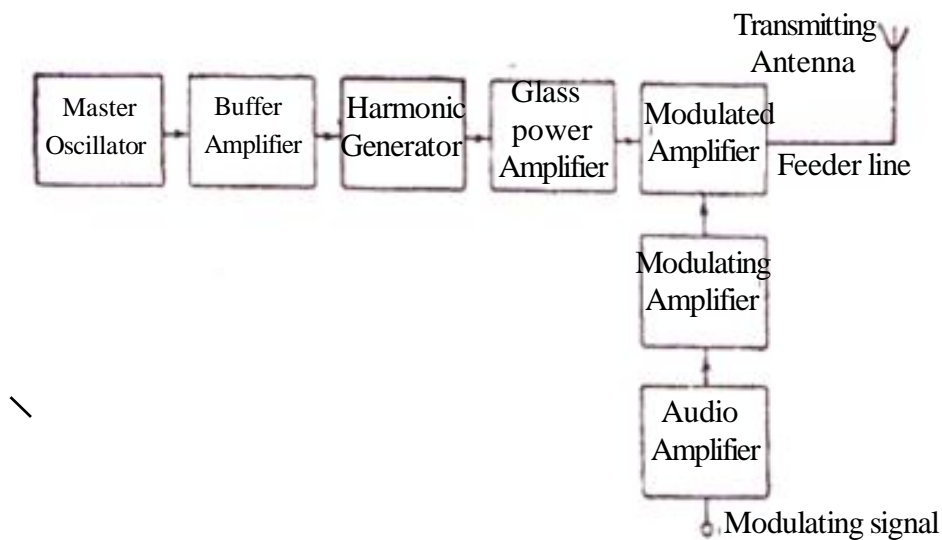


Fig 2.1 Block diagram of AM high level Transmitter

**1. Master oscillator :** It generates oscillations of desired frequency with high constancy of frequency. The generated frequency is required to remain constant within close limits, in spite of variation in the supply voltage, ambient temperature, temperature of components of load.

**2. Buffer amplifier or isolating Amplifier :** Buffer amplifier or isolating amplifier is placed between the master oscillator and the harmonic generators. This buffer amplifier does not draw input current and hence causes

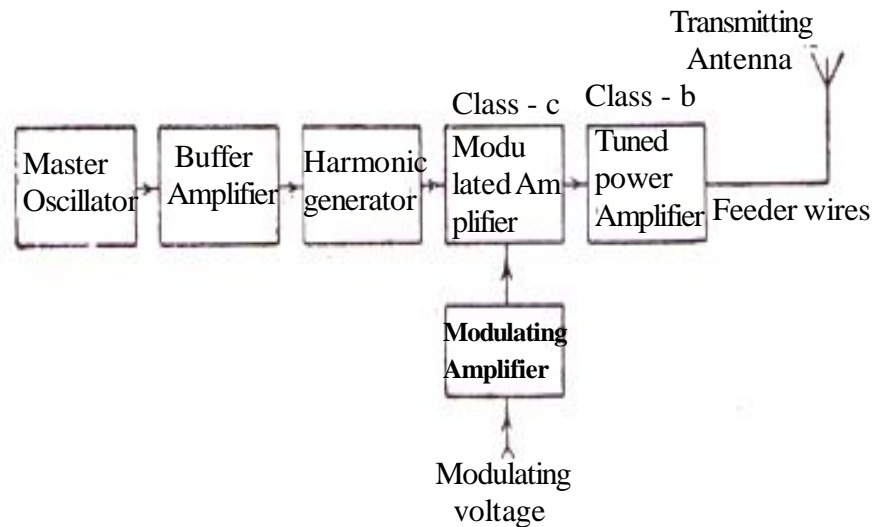


Fig 2.2 Block diagram of AM low level Transmitter

## 2.6 Concept of Digital Communication

In digital communication, the message signal to be transmitted is digital in nature. This means that digital communication involves the transmission of information in the digital form.

### 2.6.1 Basic model of Digital Communication system.

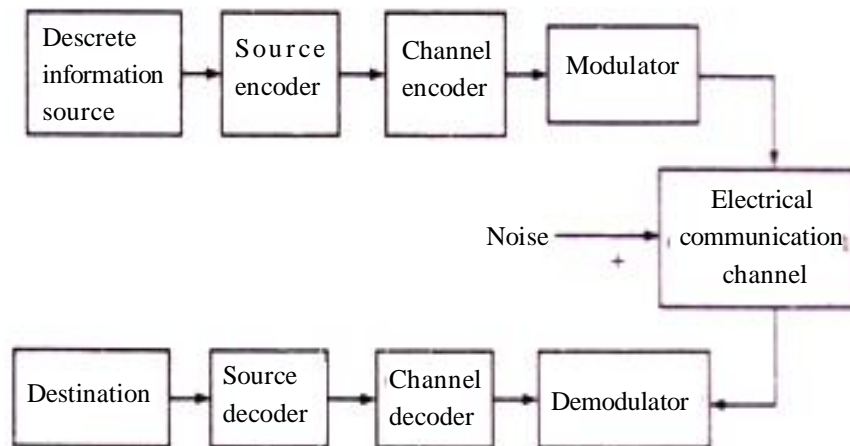


Fig. 2.3 Shows the model of digital communication system.

Sno.	Analog communication	Digital communication
5.	Needs lower band width	5.Higher channel band width requirement due to higher bit rates.
6.	Not suitable for transmission of secret information in defense applications.	6. Due to coding techniques, it is suited for military applications.
7.	Coding is not possible	7.Winding techniques can be used to detect and correct the errors.
8.	FDM is used for multiplexing.	8. TDM is used for multiplexing.

## 2.8 Pulse code modulation.

In pulse modulation continuous wave form is sampled at regular intervals and the information collected at the sampling times together with synchronizing pulses if any is transmitted. At the receiving end, the original wave form may be reconstituted with the received signal. Pulse modulation differs from AM and FM. In fact that the signal is not supplied continuously. But in spite of this, the signal at the receiver output may have negligible distortion.

Pulse modulation can be classified into two broad categories.

### (A) Analog pulse modulation

- i. Pulse amplitude modulation (PAM)
- ii. Pulse duration modulation (PDM)
- iii. Pulse position modulation (PPM)

PDM is also called pulse width modulation (PWM). PDM (or PWM) and PPM are sometimes lumped together under the general category of pulse time modulation (PTM).

### (B) Digital pulse modulation

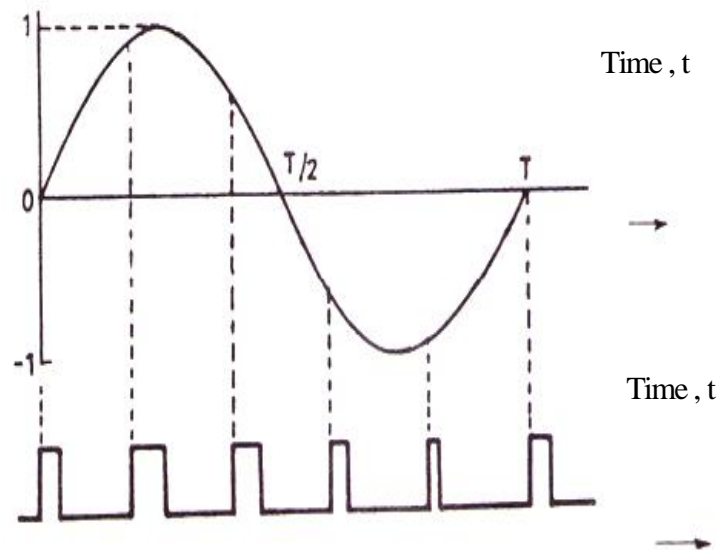
- i. Pulse code modulation (PCM)
- ii. Delta modulation

All these pulse modulations have sampling in common but they differ in the manner of indicating the sampled amplitude.



ii. Pulse position modulation

iii) Pulse frequency modulation out of these three the pulse frequency modulation is of little practical application and therefore PWM and PPM are discussed in detail.

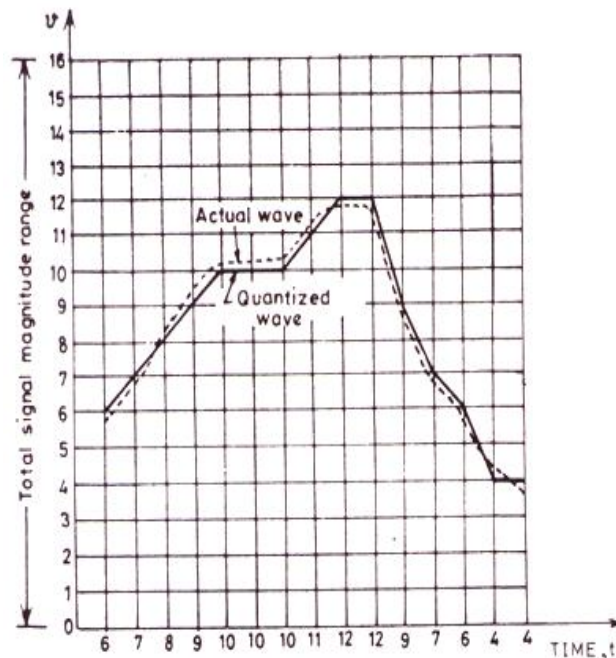


### Pulse width modulation (PWM)

In pulse width modulation, each pulse has fixed amplitude and starting time but the width of each pulse is kept proportional to the magnitude of the signal at that instant as shown in fig 2.2 the sequence of sampling pulses was shown in fig 2.5 occurring at intervals of  $0.2T$ , thus pulses occur at  $0, 0.2T, 0.4T, 0.6T, 0.8T$  and  $T$ . the corresponding values of signal at these time instants are approximately  $0, 0.8, 0.5, -0.5, -0.8$  and  $0$ . These can be represented by pulse widths of  $1, 1.8, 1.5, 0.5, 0.2$  and respectively. The width corresponding to zero value of signal was chosen in this system to be  $1.0$  res the signal value varies between the limits of  $+IV$  (width =  $2res$ ) and  $IV$  (Width= $0res$ ).

The zero magnitude of the signal is thus the average signal level and the corresponding pulse width is  $1res$ . evidently a negative pulse width is not possible. In a practical telephone system, pulse have recurrence rate of  $8000$  pulses per second. Then the time between the commencement of adjacent pulses is  $10^{-6}$

approximation may be made as close as desired but at best it remains an approximation.



In PCM, the total magnitude range of the signal is divided into a number of standard levels at equal intervals as shown in fig 2.7. These level are transmitted in a binary code, hence the actual number of these standard levels is a power of 2 such as 16,32,64 or 128. For the sake of simplicity, fog 2.7 shows only 1b levels but practical systems use as many as 128 levels. By the quantization process, the level actually sent a any sampling instant is the one nearest to the standard (ur quantum) level.

As shown in fig 2.7, the signal is continuously sampled and quantized, each sample magnitude being converted to the nearest standard amplitude, the quantized number is coded using binary code, converted into corresponding back to front binary number and then sent. If adequate quantizing levels are used, the result closely resembles the corresponding analog transmission.

PCM generation first involves in sampling and conversion to PAM. Then the PAM is quantized and encoded and supervisory signals are added. The signal is then sent directly via cable or is modulated and transmitted. PCM is highly immune to noise, Hence amplitude modulation may be used resulting in

The aim of **pulse modulation** methods is to transfer a narrowband analog signal, for example a phone call over a wideband baseband channel or, in some of the schemes, as a bit stream over another digital transmission system.

## 2.12 ASK, FSK, PSK

### Amplitude shift keying (ASK)

The ASK is also known as ON-OFF keying. We here use amplitude modulation of carrier. By pulses. ON corresponding to mark and OFF to space. However, this system suffers from the drawback that there is no indication for the space further the system being basically an AM system suffers from all drawbacks of AM. Hence this system is used only for manual Morse code CW operation but is never used for automatic telegraphy. These systems also suffer from noise.

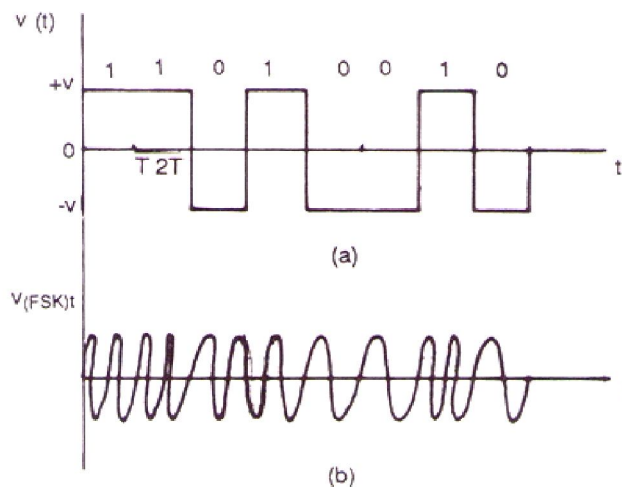


Fig 2.8 (a) Binary Signal (b) ASK Signal

Fig. 2.8 shows the wave forms of

- (a) Binary signal and
- (b) ASK signal,

It can be seen that the carrier signal is ON when  $v(t) = +v$  and it is OFF when  $v(t) = -v$ . ASK is rarely used for data transmission except in some cases.

### Frequency Shift Keying (FSK)

It is basically a frequency modulation system where in the nominal unmodulated carrier frequency corresponds to the mark condition and a lower

---

**Summary**

---

1. A radio transmitter is a system that can transmits or radiates the modulated carrier power
2. Transmitters are broadly classified as AM, FM and plus modulation transmitters based on the modulation techniques employed.
3. Depending on the service involved, transmitters are classified as radio transmitters, television transmitters, radio transmitters etc.
4. In high level modulation system, the carrier voltage is modulated at the highest power level and in low level modulation system the carrier is modulated at low power level.
5. In digital communication system, the message signal is transmitted in digital form.
6. In digital communication system, noise immunity is excellent.
7. Pulse modulation is broadly divided into analog pulse modulation and digital pulse modulation
8. Analog pulse modulation is divided into PAM, PDM, and PPM.
9. Digital pulse modulation is divided into PCM and delta modulation
10. Pulse communications find extensive application in telegraphy and telemetry.

---

**Short Type Answer Questions**

---

1. Give the classification of radio transmitters according to the type of service involved.
2. Classify the radio transmitters according to the carrier frequency
3. Classify the transmitters based on the type of modulation
4. What is low level modulation ?
5. What is high level modulation ?
6. What is pulse modulation ?
7. What is pulse amplitude modulation ?
8. What is pulse width modulation ?
9. Give the principle of pulse code modulation



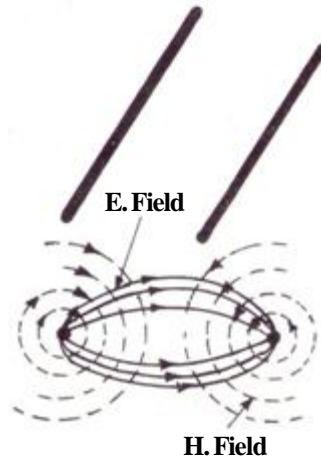
## UNIT

**3****Transmission lines, Wave Propagation & Antenna****Structure**

- 3.1 Introduction to transmission lines
- 3.2 Losses in Transmission Lines
- 3.3 Transmission-Line Matching
- 3.4 Nature and behavior of Radio Waves
- 3.5 Effects of the Environment on Electro Magnetic Waves
- 3.6 Ground wave propagation of EM waves
- 3.7 Space wave propagation of EM waves
- 3.8 Sky Wave Propagation
- 3.9 Impedance of free space
- 3.10 Introduction to Antenna systems
- 3.11 Radiation Pattern of Isotropic and halfwave dipoles
- 3.12 Radiation Resistance
- 3.13 Antenna parameters
- 3.14 Principles of operation of antenna systems
- 3.15 Concept of grounding.
- 3.16 Principle of Marconi antenna

**Basically there are four types of transmission lines**

1. Parallel wire type, a common form of transmission line also known as open wire line because of its construction.



**Fig 3.1 Electromagnetic fields in H - field open wire lines**

They are commonly employed as telephone lines, telegraphy line and power lines. Short runs of these lines are also used as antenna feeders and impedance matching purpose. These lines are easy to construct and are cheaper. Since insulation between line conductors is normally air, the dielectric loss is extremely small. Open wire lines are balanced with respect to earth. However, there is significant energy loss due to radiation. As a result of which these lines become unsuitable for frequencies above 100 MHz.

Electric energy propagating through these lines set up electric field between line conductors. These fields are at right angles to each other and to the direction of propagation and are shown in fig 3.1 this type of energy transmission is commonly known as transverse electromagnetic mode of propagation.

2. Coaxial type employs different types of construction. In this conductor is a hollow tube, the second conductor being located inside and coaxial with the tube. The dielectric may be solid or gaseous.

In order to avoid surface radiation losses taken place in open wire lines at frequencies beyond 100MHz, a closed field configuration is employed in coaxial cable by surrounding the inner conductor with an outer cylindrical hollow conductor.

- (a) TE wave (Transverse electric wave).
- (b) TM wave (Transverse magnetic wave).
- (c) TEM wave (Transverse electromagnetic wave).

The TEM mode is one the most commonly excited made in coaxial cables. It cannot be propagated in a waveguide, each component of the wave contains a factor give by  $e^{(j\omega t - P_{m,n} Z)}$  where  $w = 2\delta$ , and  $P_{m,n}$  is the propagation coefficient determining the phase and amplitude of the wave components.

For each mode of transmission, a lower limit exists where the complex quantity  $P_{m,n}$  becomes purely real and is equal to the attenuation constant  $\alpha_{m,n}$ . The amplitude of the component then decreases exponentially and the frequency goes below cut-off and the wave does not propagate when  $P_{m,n}$  is imaginary, the phase of the wave then varies with distance and wave then propagates without any attenuation. In practice,  $P_{m,n}$  can never be purely imaginary and hence some attenuation always occurs due to energy loss in transmission line.

Wave guides are widely used in communication network for the transmission of electromagnetic waves.

**2. Optical fibers :** Optical fibers are increasingly replacing wire transmission lines in communication system. Such optical fibers lines offer several important advantages over wire lines superior transmission quality, higher information carrying capacity, light weight and smaller size, reduced cost and higher security.

Because of optical fiber's low attenuation signal can be transmitted for a considerable distance. A signal repeater less fiber optical link can replace a coaxial trunk and its 35 to 40 R.F. amplifiers and deliver more channels. The optical fibre is immune to electromagnetic interference (EMI) and radio frequency interference. Fibre,s dielectric nature ensures a clean, clear signal not plagued by cross talk or degraded by severe weather. Optical fiber has tremendous band width or information carrying capacity and fibre – optical network and is not bit rate dependent. The system band width can be increased by transmitting at more than one wavelength using multiple light sources (laser operating at different wavelength). This is called wave division multiplexing which increases a fiber useful information carrying capacity.

Next, material used in fibres is silica glass, or silicon oxide, which is one of the most abundant materials on earth resulting in much lower material costs than wires lines. With the much higher information capacities, multiple channel routes using fibres can be compressed into much smaller cables, greatly reducing congestion in over crowded ducts.



a vacuum and slower in all other media. The velocity of light in a medium is given by.  $V = \frac{V_c}{\sqrt{K}}$

Where  $V$  = velocity in the medium

$V_c$  = velocity of light in a Vacuum

$K$  = dielectric constant of the medium ( 1 for a vacuum and very nearly 1 for air)

The velocity factor of a dielectric substance, and thus of a cable, is the velocity reduction ratio and is there fore given by  $V_f = \frac{1}{\sqrt{K}}$

The dielectric constants of materials commonly used in transmission lines range from about 1.2 to 2.8, giving corresponding velocity factor from 0.9 to 0.6. note also that since  $v = f\lambda$  and  $f$  is constant, the wave length  $\lambda$  is also reduced by a ratio equal to the velocity factor. This is of particular importance in stub calculations. If a section of 300 twin lead has a velocity factor of 0.82, the speed of energy transferred is 18 percent slower than in a vacuum.

### 3.3 Transmission-Line Matching

The transmitter output is usually connected to the antenna via a transmission line, which is typically coaxial cable. In other applications, the transmission line may be a twisted pair or some other medium.

A cable becomes a transmission line when it has a length greater than  $\lambda/8$  at the operating frequency where:

$$\lambda = 300/f_{\text{MHz}}$$

For example, the wavelength of a 150-MHz frequency is:

$$\lambda = 300/f_{\text{MHz}} = 300/150 = 2 \text{ meters}$$

A connecting cable is a transmission line if it's longer than  $\lambda/8 = 0.25$  meters. All transmission lines have a characteristic impedance ( $Z_0$ ) that's a function of the line's inductance and capacitance:

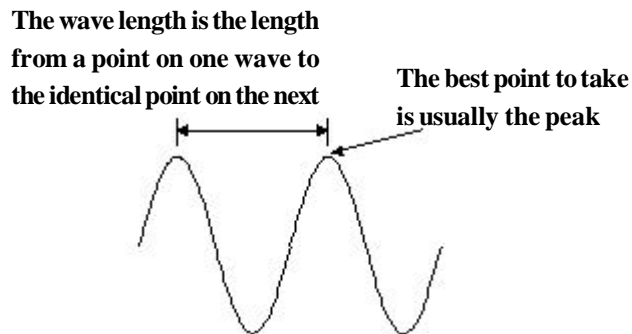
$$Z_0 = \sqrt{L/C}$$

To achieve maximum power transfer over a transmission line, the line impedance must also match the source and load impedances as shown in Fig 3.3. If the impedances aren't matched, maximum power will not be delivered. In addition, standing waves will develop along the line. This means the load doesn't absorb all of the power sent down the line.

The electric field results from the voltage changes occurring in the RF antenna which is radiating the signal, and the magnetic changes result from the current flow. It is also found that the lines of force in the electric field run along the same axis as the RF antenna, but spreading out as they move away from it.

This electric field is measured in terms of the change of potential over a given distance, e.g. volts per meter, and this is known as the field strength. Similarly when an RF antenna receives a signal the magnetic changes cause a current flow, and the electric field changes cause the voltage changes on the antenna.

There are a number of properties of a wave. The first is its wavelength. This is the distance between a point on one wave to the identical point on the next. One of the most obvious points to choose is the peak as this can be easily identified although any point is acceptable.

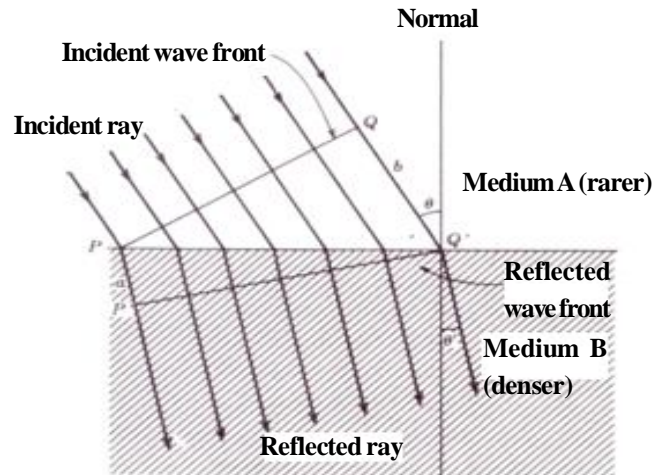


**Fig 3.5 Wave length of an electromagnetic wave**

### **3.5 Effects of the Environment on Electro Magnetic Waves**

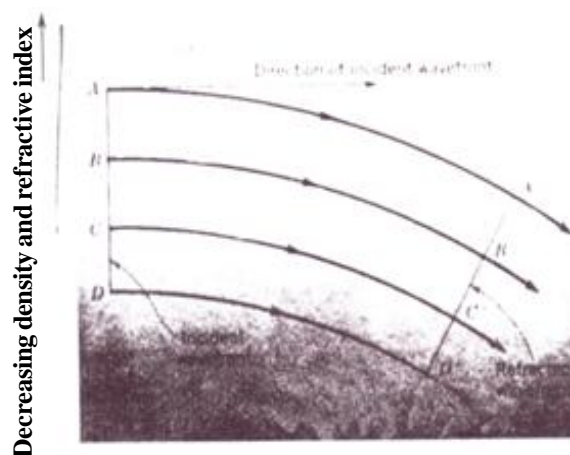
When propagation near the earth is examined, several factors which did not exist in free space must be considered. They will be refracted by the ground, mountains and buildings. They will be refracted as they pass through layers of the atmosphere which have differing densities or differing degrees of ionization. Also, electromagnetic waves may be diffracted around tall, massive objects. They may even interfere with each other, when two waves from the same source meet after having traveled by different paths. Waves may also be absorbed by different media, but it was more convenient of consider this topic in the preceding section.

### 3.5.2 Refraction



**Fig 3.7 Refraction at a plane, sharply defined boundary**

As with light, refraction takes place when electromagnetic waves pass from one propagating medium to a medium having a different density. This situation causes the wave front to acquire a new direction in the second medium and is brought about by a change in wave velocity. The simplest case of refraction, concerning two media with a plane, sharply defined boundary between them, is shown in Fig 3.7.



**Fig 3.8 Refraction in a medium having linearly decreasing density**

When the boundary between the two media is curved. Refraction still takes place, again following the optical laws. If the change in density is gradual, the situation is more complex, but refraction still takes place. Just as fig 3.7

large object between them. This is particularly noticeable on some long wave broadcast transmissions. For example the BBC long wave transmitter on 198 kHz is audible in the Scottish glens where other transmissions could not be heard. As a result the long wave transmissions can be heard in many more places than transmissions on VHF FM.

To understand how this happens it is necessary to look at Huygen's Principle. This states that each point on a spherical wave front can be considered as a source of a secondary wave front. Even though there will be a shadow zone immediately behind the obstacle, the signal will diffract around the obstacle and start to fill the void. It is found that diffraction is more pronounced when the obstacle becomes sharper and more like a "knife edge". For a radio signal a mountain ridge may provide a sufficiently sharp edge. A more rounded hill will not produce such a marked effect. It is also found that low frequency signals diffract more markedly than higher frequency ones. It is for this reason that signals on the long wave band are able to terrain where signals at VHF and higher would not. Provide coverage even in highly or mountainous areas

### 3.6 Ground wave propagation of EM waves

The ground wave is the radio wave which results because of the presence of the ground. The ground wave may further be classified into two categories:

- (a) Surface wave
- (b) Space wave (tropospheric waves)

The surface wave is that part of the radio wave which travels along the surface of earth. It is vertically polarized. i.e., the electric vector of the electromagnetic wave is vertical. The wave is supported by the lower edge of the ground. Such propagation takes place when the transmitting and receiving antennas are close to the surface on earth. Surface wave propagation is of importance only for medium wave and long wave signals. All medium wave signals received during day time use surface wave propagation.

### 3.7 Space way propagation of EM waves

1. The space wave, on the other hand, is that part of the radio wave which travels from the transmitting antenna to the receiving antenna through the space, i.e., the earth's troposphere. The region of earth's atmosphere extends from the earth's surface up to about 15 kilometers. The space wave is constituted by two components namely (i) the direct wave and (ii) the ground reflected wave as shown in fig. 3.3 this diagram neglects the curvature of earth and the curvature of the radio waves produce by the variations of the refractive index of the earth's atmosphere with the height. In 3.3, T represent the transmitting antenna

reach the receiving antenna as a result of reflection or refraction produced by sizeable variations in the electrical characteristics of the troposphere and by diffraction around the curvature of earth.

### **The surface wave propagation**

The space wave is that part of the radio wave which travels along the surface of earth. It is vertically polarized. All medium wave signals received during day time use surface wave propagation. When the transmitting and receiving antennas are close to the surface of the earth, the space wave is negligibly small and the surface wave alone is of importance, thus the day time propagation of broadcast (medium wave) signals takes place only through surface wave.

### **The surface wave propagation may be studied under the following two heads**

- (i) Short distance surface wave propagation.
- (ii) Long distance surface wave propagation

Short distance (or plane earth) surface propagation. If the distance of the receiving antenna from the transmitting antenna is short, the curvature of the earth's surface may be neglected and the earth may be assumed to be plane in the intervening distance. If the heights  $h_t$  and  $h_r$  of the transmitting and receiving antenna expressed in wave lengths are low, then the surface wave alone may be considered. The field strength for surface wave propagation for a flat earth is given by the sommerfeld equation.

Where  $E_0$  is the unit distance field strength at the surface of earth, neglecting earth's losses,  $d$  is the distance from the transmitting antenna expressed in the same unit as used in  $E_0$  and  $A$  is a factor which takes into account the losses caused by the earth.

Unit distance field strength  $E_0$  depends upon (a) power radiated from the transmitting antenna and (b) the directivity of the transmitting antenna in the vertical and horizontal planes. With an antenna which is non directional in azimuth and which produces a field varying proportional to the cosine of the angle of elevation (as is the case with short vertical antenna), the field strength at a distance of 1 kilometer is given by,

$$E_0 = 300\sqrt{P} \text{ milli – volts meter}$$

$$\text{And at one mile, } E_0 = 186.4 \text{ milli – volts/meter}$$

Where  $P$  is the effective radiated power in kilo-watts

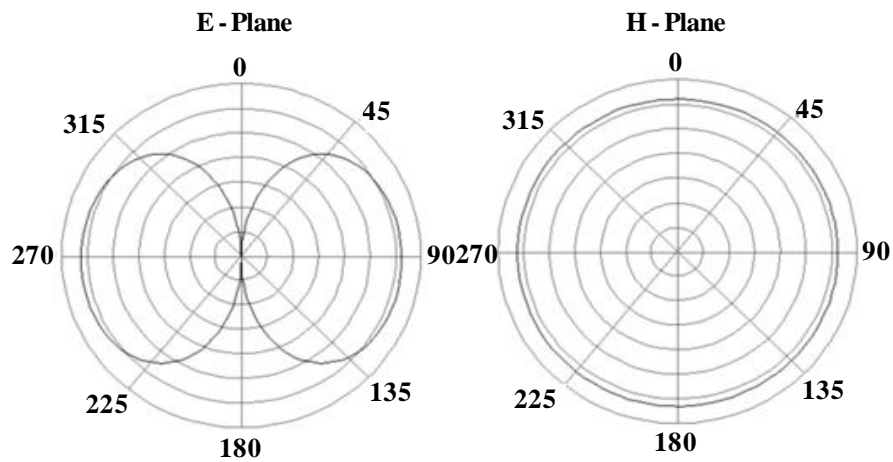


Fig 3.10 Typical 3D Radiation pattern of a depole antenna

### 3.11.1 An isotropic radiator

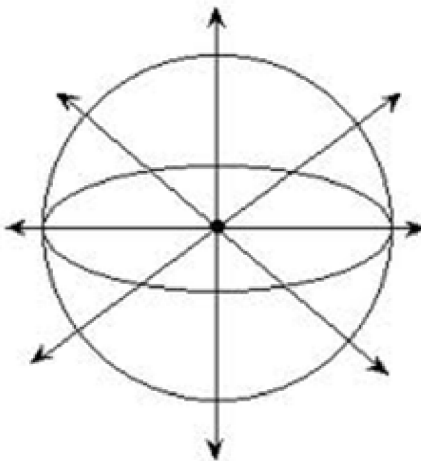


Fig. 3.11 Isotropic radiator

Is a theoretical point source of electromagnetic or sound waves which radiates the same intensity of radiation in all directions. It has no preferred direction of radiation. It radiates uniformly in all directions over a sphere centred on the source. Isotropic radiators are used as reference radiators with which other sources are compared.

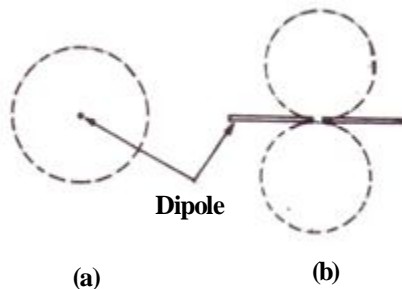
### 3.11.2 Half wave dipole

The most common form of dipole has an electrical length of half a wavelength. As a result this antenna is called a half wave dipole. As before the

### 3.13 Antenna parameters

#### (a) Directivity

The directivity of an antenna is the relative ability of an antenna to receive in different direction in space. It is usually plotted in the form of a polar diagram or polar pattern in the plane of interest. Directivity is often expressed by parameter like beam width, front-to-back ratio and auxiliary lobes in various planes. It can receive or transmit equally well in all direction in that plane. Its directivity pattern in the plane passing through the dipole length is a figure of eight pattern, i.e., a cosine function with respect to angle with the normal to the dipole, as shown in Fig 9.3(b). the pattern is thus a doughnut shaped pattern in a three dimensional plot. The pattern alters at frequencies. The pattern alters at frequencies off – resonance due to change in the current distribution at these frequencies. The pattern can be made more directive by addition of reflectors and direction at appropriate distances from the dipole.



**Fig 3.14 14 Directivity of a dipole antenna : (a) Circular pattern in perpendicular plane, (b) Figure-of-eight pattern in the dipole plane**

**(b) Beam width or Lobe width :** The directivity of an antenna is expressed simply by the beam width of the main lobe in the directive pattern as the angle within which the signal received is within 3 dB or half of the maximum power.

#### (c) Front to back ratio

This is the ratio of the amount of signal received from the front to the received from the back. This indicates the directivity of an antenna.

#### (d) Directive Gain

This is defined as the ratio of power received by an antenna in a particular direction (usually in the direction of the main lobe) to the power that would be received by an isotropic antenna i.e., an antenna receiving equally in all directions. Though such an isotropic antenna does not exist, its properties are easy to visualize and calculate.



The half-wave antenna is equivalent to a resonant circuit and its determines the band width. Large diameter conductors have a lower resistance but much lower reactance (due to equivalent paralleling of inductances or capacitances of smaller conductors connected in parallel to form a larger conductor). Hence fat dipoles employing larger tubes for their conductors or fat dipoles made out of flat strips (as shown in Fig 111) have a lower Q and hence a wider frequency response.

### 3.14 Principles of operation of antenna's system

The interaction of ground with antenna impedance and radiation characteristics has been touched on previously. Now is the time to go into a more detailed discussion of the interaction.

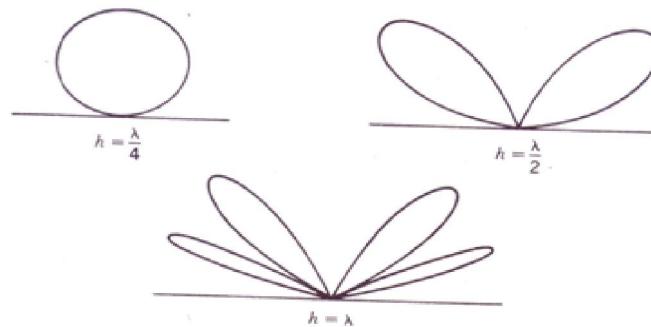


Fig 3.16 Radiation patterns of an ungrounded half-wave dipole located at varying heights above the ground

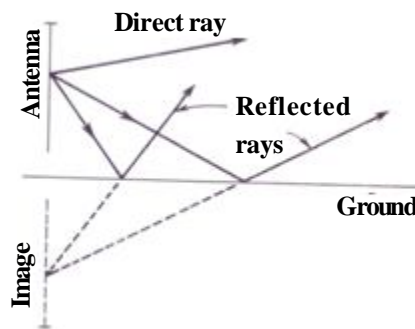


Fig 3.17 Ungrounded antenna and image

#### Ungrounded Antennas

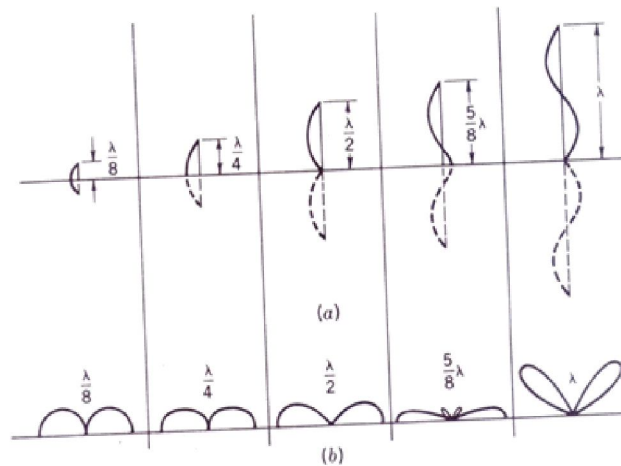
when a radiation source is placed near a reflecting surface, the signal received at any distant point is the vector sum of the direct wave and the reflected



The effect is caused by cancellation of the wave in the horizontal direction because of opposing currents in the various parts of an antenna at this effective height.

### Grounding Systems

The earth has generally been assumed to be perfect conductor so far. This is often not the case. For this reason the best ground system for a vertical grounded radiator is a network of buried wires directly under the antenna. This network consists of a large number of "radials" extending from the base of the tower, like spokes on a wheel, and placed between 15 and 30 cm below the ground. Each radial wire has a length which should be at least  $\lambda/4$ , and preferably up to 120 such wires may be used to good advantage, and the whole assembly is then known as a ground screen. A conductor.



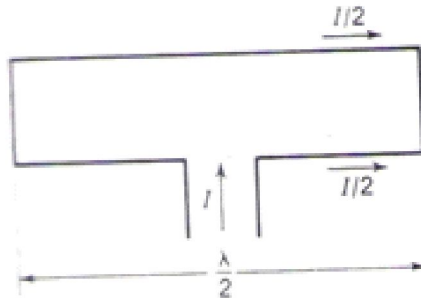
**Fig 3.19 Characteristics of vertical grounded antennas. (a) Heights and current distribution (b) Radiation patterns**

Joined all the radials. At a distance of about half the radial length, is often employed. The far end of each radial is grounded, i.e., attached to a metal stake which is driven deeply into the subsoil (especially if this is a better conductor than the topsoil, as in sandy locations).

A good screen will greatly improve the field strength and distance of Marconi antennas, especially those used for medium-frequency broadcasting. The improvement is most pronounced for short antennas (under  $\lambda$  is height). And/or with soil of poor conductivity. Even an antenna between  $\lambda/4$  and  $\lambda/2$ , on soil with good conductivity. Will have its radiation pattern improved noticeably

When a ground screen is not practical, a counterpoise is used. A counterpoise consists of a system of radials, supported aboveground and

then the current in each arm is  $I/2$ . If this had been a straight dipole, the total would have flowed in the first (and only) arm now with the same power applied, only half the current flows in the first arm and thus the input impedance is four times that of the straight dipole. Hence  $R_r = 4 \times 72 = 288$  for a half-wave folded dipole with equal diameter arms.



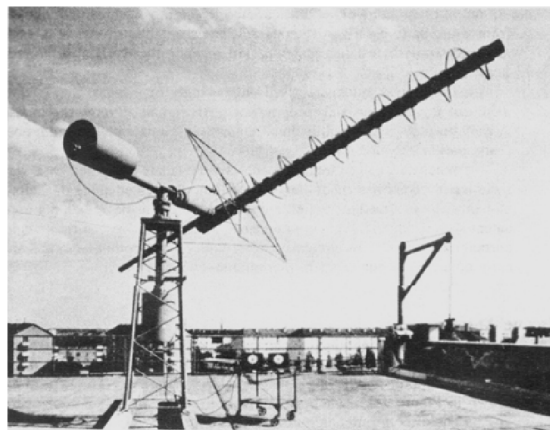
**Fig 3.21 Folder Dipole**

If elements of unequal diameters are used, transformation ratio from 1.5 to 25 may be practicable, and if greater ratios are required, more arms can be used. Although the folded dipole has the same radiation pattern as the ordinary dipole, it has many advantages: its higher input impedance and its greater bandwidth as well as ease and cost of construction and impedance matching.

### 3.18 Working and application of loop, helical antenna.

#### Helical Antenna

A helical antenna, illustrated in figure 3.22 is a broadband VHF and UHF antenna which is used when it is desired to provide circular polarization characteristics, mainly for reasons as described.



**Fig 3.22 Helical Antenna**

is then necessary to rotate the loop until the received signal is minimum. The plane of the loop is now perpendicular to the direction of the radiation. Since the loop is bidirectional, two bearings are required to determine the precise direction. If the distance between them is large enough. The distance of the source of this transmission may be found by calculation.

There are a large number of variations on the theme of the loop, far too many to consider here. They include the Alford loop, cloverleaf, Adcock antenna, and the Bellini-Tosi antenna.

Loops are sometimes provided with several turns and also with ferrite cores, these, being magnetic, increase the effective diameter of the loop. Such antennas are commonly built into portable broadcast receivers. The antenna configuration explains why, if a receiver tuned to any station is rotated, a definite null will be noticed.

### 3.19 Radiation pattern and applications of Yagi / Uda antenna

A Yagi-Uda antenna is familiar as the commonest kind of terrestrial TV antenna to be found on the rooftops of houses. It is usually used at frequencies between about 30MHz and 3GHz, or a wavelength range of 10 metres to 10 cm. (There are some obsessive amateur radio enthusiasts who construct Yagi-Uda antennas for the 80 metre wavelength band. This is rather impractical as spacing them from the ground by more than half a wavelength is difficult.) The rod lengths in a Yagi-Uda are about a half wavelength each, and the spacings of the elements are about 1/3 of a wavelength. This puts the overall sizes of Yagi-Udas in the ranges.



Fig 3.24 Yagi Uda Antenna

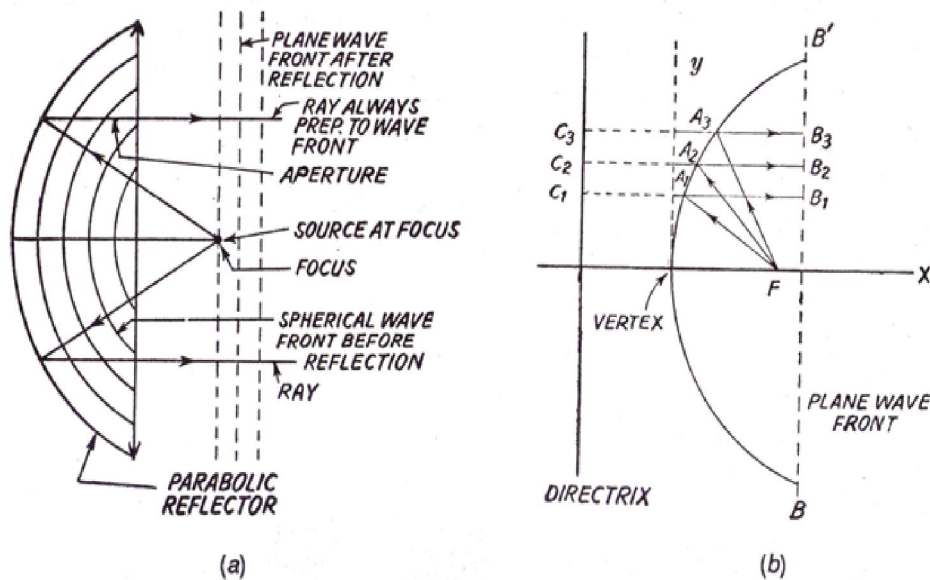
narrow beam. The same directivity objectives to built antennas with apertures of many wave lengths.

**Focusing by a Parabolic Reflector.** A parabolic reflector antenna is shown in fig 3.24 The parabola reflects the waves origination from a source at the focus into a parallel beam and consequently these spherical waves originated by the source at the source at the focus of the reflector are converted into a plane wave of uniform phase across the mouth or aperture or the parabola this can be expressed as follows.

$$FA_3 = A_3C_3$$

$$FA_2 = A_2C_2 \text{ etc.}$$

$$C_3B_3 = C_3A_3 + A_3B_3 = FA_3 + A_3B_3 = \text{Path } FA_3B_3$$



**Fig 3.24 Parabolic reflector focusing action**

Parabolic curve is defined such that the distance from a fixed point F. called focus to any point. Say  $A_3$ , on the curve is always equal to the perpendicular distance between that point  $A_3$  on the curve and fixed line called directrix, i.e.

$$\text{Also } C_2B_2 = C_2A_2 + A_2B_2 = FA_2 + A_2B_2 = \text{Path } FA_2B_2$$

$$C_3B_3 = B_3C_3 \text{ So that path } FA_3B_3$$

Or in other words, the length of this path is equal to the distance between directrix and plane  $BB'$  irrespective of the angle at which the ray is incident on

## Summary

1. A **transmission line** is conductive method of guiding electrical energy from one place to another. In communications, these lines are used as a link between an antenna and transmitter or a receiver

2. Basically there are four types of transmission lines.

(a) Parallel wire type (b) Coaxial type (c) Wave guides (d) Optical fibres

3. There are three ways in which energy, applied to a transmission line, may become dissipated before reaching the load: radiation, conductor heating and dielectric heating.

4. To achieve maximum power transfer over a transmission line, the line impedance must also match the source and load impedances. If the impedances aren't matched, maximum power will not be delivered. In addition, standing waves will develop along the line. This means the load doesn't absorb all of the power sent down the line.

5. Electromagnetic waves are the same type of radiation as light, ultra-violet and infra red rays, differing from them in their wavelength and frequency. Electromagnetic waves have both electric and magnetic components that are inseparable. The planes of these fields are at right angles to one another and to the direction of motion of the wave. They travel with the velocity of light i.e., 300000 Kilometres/second

6. The **ground wave** is the radio wave which results because of the presence of the ground. The ground wave may further be classified into two categories: a) Surface wave b) Space wave (tropospheric waves)

7. The **surface wave** is that part of the radio wave which travels along the surface of earth. It is vertically polarized. All medium wave signals received during day time use surface wave propagation

8. The **space wave** is that part of the radio wave which travels from the transmitting antenna to the receiving antenna through the space, i.e., the earth's troposphere. The region of earth's atmosphere extends from the earth's surface up to about 15 kilometers. The space wave propagation is normally used for frequencies above 30MHz i.e., for television, FM broadcast etc.

9. Long distance radio communication is possible through the sky wave propagation

10. The **impedance of free space**,  $Z_0$ , is a physical constant relating the magnitudes of the electric and magnetic fields of electromagnetic radiation

3. Explain the importance of impedance matching with respect to transmission lines
4. What is refraction?
5. What are the losses that occur in transmission lines?
6. What is meant by diffraction of radio waves?
7. Sketch the radiation patterns of isotropic and half wave dipoles
8. Give the value of impedance of free space
9. Define radiation resistance
10. What is a folded dipole and give its applications?
11. Name the elements of a Yagi-Uda antenna
12. Define the term directivity of an antenna
13. What is beam width or lobe width?
14. Define front to back ratio of an antenna system
15. Define directive gain and antenna gain
16. Give the value of dipole antenna impedance
17. Define the term Impedance with respect to an antenna system
18. Write the principle involved in parabolic reflectors
19. Describe the principle of Marconi or Monopole antenna
20. What is the use of reflector and director in Yagi-Antenna system?

---

### Long Answer Type Questions

---

1. Explain about ground wave propagation of electromagnetic waves
2. Describe the space wave propagation of electromagnetic waves
3. Describe the sky wave propagation of electromagnetic waves
4. Describe about reflection, refraction and diffraction of electromagnetic waves
5. Define Antenna gain, Directivity, Beam width and Front to back ratio of an antenna system
6. Describe the operation and applications of a folded dipole



## UNIT

# 4

### Radio Receivers

#### Structure

- 4.0 Introduction
- 4.1 Basic function of a radio receiver
- 4.2 Receiver Characteristics
- 4.3 Comparison between AM and FM receiver
- 4.4 Working of super heterodyning in radio receiver
- 4.5 Block diagram and working of super heterodyning receiver
- 4.6 Need for AVC (AGC)
- 4.7 Demodulation in FM receivers
- 4.8 Block diagram of TRF receiver working
- 4.9 IC radio receiver block diagram and working
- 4.10 Common faults and servicing of radio receiver

#### Learning Objectives

After completing this unit, the student will be able to understand

- The basic function of radio receiver
- The classification of radio receivers



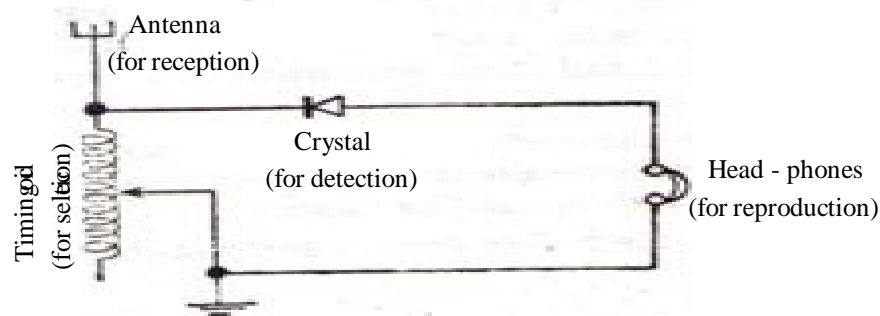
- i. Amplitude Modulation (AM) broadcast receivers
- ii. Frequency Modulation (FM) broadcast receivers
- iii. T.V.Receivers
- iv. Communication receivers
- v. Radar receivers
- vi. Code receivers

**(b) Depending upon the technique of operation**

- i. Straight receivers.
- ii. Super heterodyne receivers.

**Straight Receivers.**

In a straight receiver, the receiver operation is in a straight forward manner without frequency conversion. The Fig 4.1 shows the simple circuit of a crystal receiver in which the signal is picked up by the antenna, a tuning coil selects the desired signal, a crystal diode is used for the detection of the audio modulating signal and finally the loudspeaker reproduces the original programme.

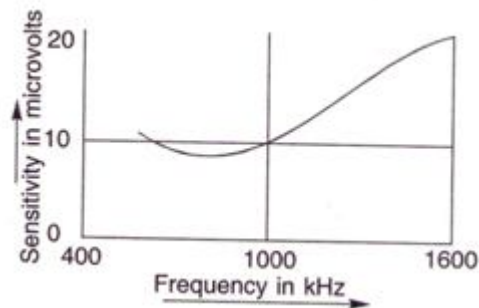


**Fig. 4.1 Circuit diagram of a Simple Crystal Receiver**

The crystal receiver has drawbacks of poor sensitivity and poor selectivity.

**Tuned Radio Frequency (TRF) Receiver**

The TRF receiver is also a straight receiver in which the incoming signal is first amplified in one or more tuned RF amplifier stages. This increases the magnitude of the signal and hence improves the sensitivity of the receiver. The amplified signal is then fed to the detector to re obtain the original modulation



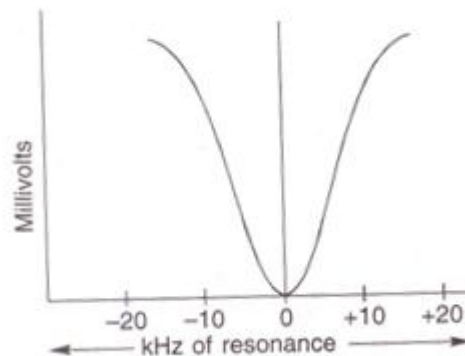
**Fig 4.3 Typical Sensitivity Curve of a Receiver**

For measuring sensitivity, the input RF signal must be modulated 30% at an AF of 400Hz. Moreover, the RF signal voltage is applied through an artificial antenna called the dummy antenna.

The sensitivity of a receiver is expressed in microvolts. The smaller the input in microvolts, the greater is the sensitivity of a receiver. A high grade broad cast receiver will have a sensitivity of less than  $10\text{ }\mu\text{v}$ . The sensitivity curve of a receiver is shown in Fig 4.3

### (ii) Selectivity

Selectivity of a receiver is its ability to select a desired signal frequency without any objectionable interference from other neighboring stations. It is a measure of the extent to which the receiver can reject all other neighboring stations and accept only the desired station.



**Fig 4.4. Typical Selectivity Curve of a Receiver**

The selectivity is generally expressed in the form of a curve as shown in fig 4.4. The sharper the selectivity curve, the more selective a receiver is.

Here again, the composite-signal is amplified and sent to the detector.

(2) In the AM and the FM detector, the information is removed from the carrier, and sent to the amplifier and converted to sound by the loud-speaker.

b. AM and FM differences.

(1) One difference between the two radio functions are the frequencies received and the detection systems.

(2) Another difference is that the FM radio requires an AFC to stabilize the local oscillator, thereby keeping the desired broadcast station tuned in.

AM Receiver	FM Receiver
1. Noisy reception.	1. Noiseless reception.
2. RF amplifiers used have a band width of 10 KHz.	2. RF amplifiers used have a band width of 150 KHz.
3. AM receivers are cheap and simple in design.	3. FM receivers are complex and costly.
4. Typical value of IF is 455 KHz.	4. Typical value of IF is 12 MHz.
5. AM receivers do not employ limiter stage.	5. FM receivers have an additional stage of limiter which removes amplitude variations.
6. In AM receivers, local oscillator frequency is kept higher than the income signal frequency by an amount of IF i.e; $f_o = f_s + f_i$ .	6. In FM, oscillator frequency is kept smaller than the incoming signal frequency by an amount of IF i.e; $f_o = f_s - f_i$
7. In AM, a signal transistor may be used for both mixer and oscillator stages.	7. Separate transistors are used for the both local oscillator and frequency mixer.
8. AF amplifiers used can accommodate a bandwidth of 5 KHz.	8. AF amplifiers used can accommodate a band width of 15 KHz.
9. A single loudspeaker may be used for the reproduction of audio modulating signal	9. Often two or more loudspeakers are used to cover 15KHz coverage of modulating signals.

### 4.5 Block diagram of AM Super Heterodyne Radio Receiver

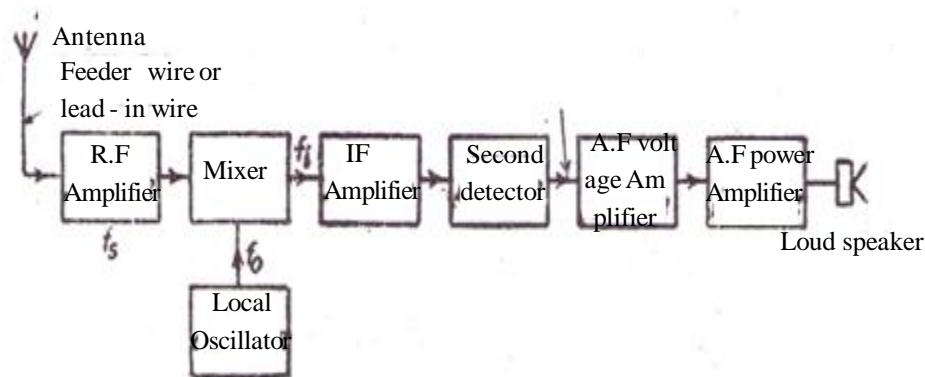


Fig 4.7 Block diagram of an AM Super heterodyne Radio Receiver

**Antenna or Aerial :** It intercepts the electromagnetic waves. The voltage so picked up is fed to the input of the RF amplifier stage

**RF Amplifier :** This stage is generally a small signal tuned voltage amplifier tuned to the desired carrier frequency the main functions of RF amplifier are

1. To increase the signal/ noise ratio by amplifying the input signal voltage and
2. To provide discrimination or selectivity against image frequency signal and intermediate frequency signal

**Frequency Converter :** This consists of local oscillator and frequency mixer. To the frequency mixer are fed both the local oscillator voltage as well as signal voltage. The mixer, being a nonlinear device, produces at its output the various intermodulation terms. The difference frequency voltage picked up by the tuned circuit in the output circuit of the mixer. This difference frequency is called the intermediate frequency and is standardized at 455 kHz in AM receivers.

**IF Amplifier stage :** It consists of two or more stages of fixed frequency voltage amplifier having a 3dB bandwidth of 10 kHz for AM broadcast. This IF amplifier provides most of the receiver amplification and selectivity.

**Second detector :** Output from the last IF amplifier is given to a linear diode detector to recover the original modulation frequency voltage.

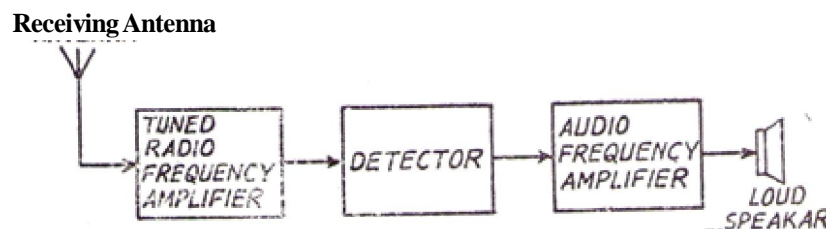
**AF amplifier :** Audio frequency output from the second detector is given to the input of the AF amplifier for providing further amplification of the

Since point C is the true electrical centre of the secondary the RF voltages  $V_{1c}$  and  $V_{2c}$  AS SHOWN IN Fig 4. are equal in magnitude and opposite in phase. The RF voltage  $V_{10}$  applied to diode  $D_1$  is equal to  $V_1 + V_{1c} = V_1 + V_{1c}$ . similarly RF voltage  $V_{20}$  applied to diode  $D_2$  is equal to  $V_1 + V_{2c} + V_i + V_{2c}$ . voltages  $V_{1c}$  and  $V_{2c}$  are not in phase with  $V_i$  and hence these have to be added vectorially to  $V_i$  to produce to voltages  $V_{10}$  and  $V_{20}$ .

RF voltage  $V_{10}$  and  $V_{20}$  are separately rectified in linear detector diodes  $D_1$  and  $D_2$  respectively to produce output voltage  $V_{30}$  and  $V_{40}$  across the resistors  $R_3$  and  $R_4$ . The RF components of the rectified currents are bypassed through shut capacitors  $C_2$  and  $C_4$  leaving only the modulation frequency component to flow through resistor  $R_3$  and  $R_4$ . The output voltages  $V_{30}$  and  $V_{40}$  then represent the amplitude variation of RF voltages  $V_{10}$  and  $V_{20}$ . The final output voltage,  $V_o$  is then equal to the arithmetic difference  $|V_{30}| - |V_{40}|$  this resultant rectified output voltage  $V_o$  will then vary with the instantaneous frequency of the applied signal in the same way as the difference  $|V_{10}| - |V_{20}|$  of RF voltage applied to the diodes. Where  $K$  is a constant of proportionality.

#### 4.8 Working of TRF receiver

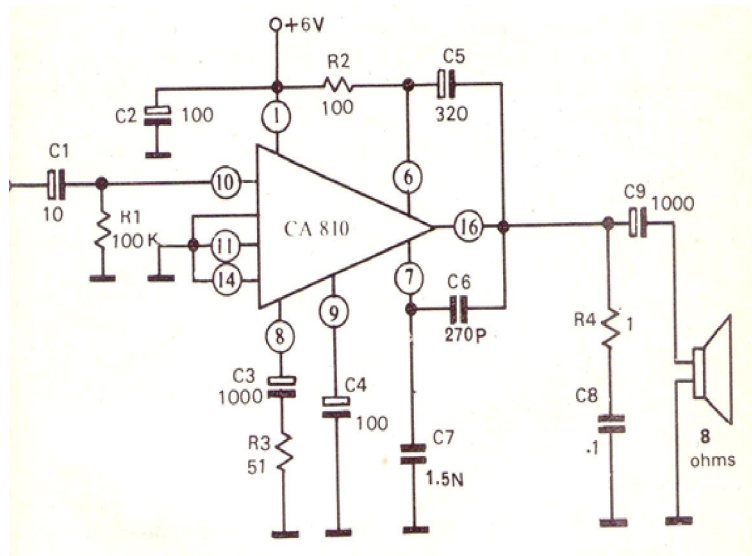
The TRF receiver is also a straight receiver in which the incoming signal is first amplified in one or more tuned RF amplifier stages. This increases the magnitude of the signal and hence improves the sensitivity of the receiver. The amplified signal is then fed to the detector to re obtain the original modulation frequency signal. The modulation frequency is further amplified in one or more stages of audio frequency amplifiers before being fed to the loud speaker. Basic block diagram of a TRF receiver was shown in



#### Disadvantages

- (i) Instability
- (ii) The selectivity of TRF receiver is poor
- (iii) TRF receiver suffers from the drawback of band width variation over the tuning range.
- (iv) Lower fidelity

### Audio amplifier circuit using CA810



**4.20 Typical Audio amplifier circuit using CA810**

A typical audio amplifier circuit using CA810 was shown in Fig 4.20. CA810 forms the complete circuit of an audio amplifier. The circuit has been designed for operation at 6 volts supply and is suitable for use in transistorized radio receivers. The circuit can deliver .6w output in 8Ω load. CA810 is widely used in AM and AM/ FM radio receivers

A complete super heterodyne AM radio receiver with monolithic IC LM 1820 is shown in Fig 4.21. We have to make tuning circuit arrangement externally for coupling the in-built RF amplifier, mixer and IF amplifier stages. It is available in 14 pin version. The linear IC AM radios find extensive applications in high quality AM radio receivers.

### other ICs used in radio receiver

**The TAA300 IC :** It consists of an input stage, a driver stage and an output stage.

**The BEL 700 IC :** The monolithic integrated circuit BEL 700, in dual – in-line plastic package is designed for use in AM and AM/FM radio receivers. This versatile IC circuit contains all the blocks that are required for an AM receiver except for the mixer/oscillator and the detector stages. Similarly, for FM receiver, the tuner and the ratio detector are to be provided externally.

The integrated circuit requires few external connections for required amplification. The transformers shown in dotted boxes are double tuned IF transformers and are usually available as a complete assembly for all standard intermediate frequencies

#### 4.10 Common faults and servicing of radio receivers

**(a) Signal substitution testing :** In testing by signal substitution a signal frequency and proper voltage is applied at some selected points in the radio starting from the loudspeaker terminals and moving back. On applying the signal at any point if normal sound is heard the stages following that point are likely to be all right. If no sound is heard from any point, or the sound is low or distorted the stage following the point is likely to be defective

Testing by signal substitution requires a signal generator for giving signals at various points. If a signal generator is not available a MW oscillator or even a signal injector can be used. This low sound or distorted sound.

**(b) Disturbing :** In this method various points in a radio are touched by the point of a screw driver. This will create a disturbance at that point and will give crackles at the loudspeaker. If no sound is heard on touching a point the stages following the point are likely to be defective.

This method is similar to signal substitution testing and can be used in case of low sound or no sound

**(c) Signal tracing :** In signal tracing, signal is given to the aerial terminal of the radio from a signal generator and signals available at various points are checked with a signal tracer, moving progressively from convertor base to the loudspeaker terminals. The defective stage, in this case, is likely to be before the point at which sound is not heard, it becomes low or distorted.

This technique requires a signal tracer besides a signal generator, this is useful in case of no sound owing to the fact that an exact instrument, a signal tracer is required this is not much used in our country.

**(d) Stage shorting :** In this method selected points in a radio are shorted to the chassis using a condenser of suitable value. In case of oscillations, once a point is shorted, the oscillations will not be heard if these are being produced in a stage ahead of the point of shorting. This method is suitable in case of oscillations. Noise and whistling etc.

Symptom	Defective stage / components
10. The radio operates for a few minutes after switching in and then stops.	10. Weak cells. Partly weak cells and high current drain from battery (see under fault7)

## Summary

1. The basic functions of a radio receiver are

- (a) Reception
- (b) Selection
- (c) Detection or demodulation and
- (d) Reproduction

2. In a straight radio receiver, the signal is processed in a straight forward manner without using frequency conversion. e.g, crystal receivers and TRF receivers.

3. TRF receivers suffer from the draw- backs of poor selectivity, lower fidelity and poor sensitivity.

4. The important characteristics of a radio receiver are

- (a) Sensitivity
- (b) Selectivity
- (c) Fidelity

5. Sensitivity of a radio receiver is its ability to respond to weak signals

6. Selectivity of a radio receiver is its ability to select a desired signal without and objectionable interference from other neighboring stations.

7. Fidelity is the ability of a radio receiver to reproduce faithfully all the audio frequencies with which the carrier is modulated

8. When a receiver picks up the same short wave station at nearly points on the receiver dial the double spotting will be decreased with the increase of the image frequency rejection.

9. In a super heterodyne radio receiver, the incoming signal frequency is mixed with the local oscillator signal too produce the intermediate frequency. Thus frequency conversion take place so as to obtain good image signal rejection



**Long Answer Type Questions**

1. Define sensitivity, selectivity and fidelity of a radio receiver.
2. Draw the block diagram of AM super heterodyne radio receiver and explain the function of each block.
3. Draw the circuit diagram of an RF amplifier and describe its working.
4. Draw the circuit diagram of frequency convertors and explain its working.
5. Draw the circuits diagram of an IF amplifier and explain its working
6. Draw the circuit diagram of detector stage and explain its working
7. Draw the block diagram of FM super heterodyne radio receiver and explain its working.
8. Draw the circuits diagram of typical limiter stage used in FM receivers and explain its working
9. Compare AM and FM receivers.
10. Draw the circuit diagram of a foster-seeley discriminator and explain in its working.
11. Give the procedure fw RF and IF alignment in a radio receiver
12. Explain in brief about
  - (a) Signal tracing and
  - (b) Signal substitution methods of radio servicing.
13. Write any three common faults in radio receiver and mention the reasons for them.

- The Reflection, Refraction, Diffraction and absorption of sound waves
- The concept of Hi-Fi system
- Multi way speaker system
- Two Crossover network
- The difference between mono and stereo systems
- The various stereo amplifier controls
- The basic requirements of a PA system and its block diagram
- Typical PA system installation plan for college sports
- The basic principle of Disc recording and reproduction systems
- The basic block diagram of a tape recorder and the tape transport mechanism in it
- The advantages and disadvantages of magnetic recording over disc recording

## **5.1 Introduction**

A public address system is a system for amplifying speech so that it can be comfortably heard by larger gatherings and at larger distances. It is used in sports meets, public meetings, auditoriums, concerts and functions. It is also used to convey information at railway stations, airports, hospitals, factories etc.

### **5.1.1 Speech**

Speech is one form of communication. The following organs help to produce sound a) lungs b) vocal chord in the larynx and cavities formed by the larynx, and the nose which acts as a resonator. By the expansion and contraction of lungs, air gushes out through the narrow slit of the vocal cord which is made open by a membranous reed under tension of vocal cord and vibrates with the desired frequency. This vibration is communicated to the cavities of mouth and nose which function as resonators.

### **5.1.2 Music**

A musical sound has the following properties.

- (a) Periodicity
- (b) Regularity of frequency
- (c) Regularity of shape

### 5.2.2 Refraction of sound wave

Refraction is a general property of the wave motion. As sound travels in wave, it is expected that when it falls on the surface of separation of two media, where is a difference in density and hence difference in velocity of propagation of sound, there will be change in the direction of propagation in the two media, except when the incidence is normal to the surface of separation. Like light, it is also totally internally reflected, when travels from denser to rarer medium making an angle of incidence greater than the critical angle

### 5.2.3 Absorption of sound

All materials absorb sound to some extent. Hard inflexible substances with shining surfaces may absorb very little sound.

The term absorption coefficient is used to represent absorbent properties of materials. If the absorption coefficient of a material is represented by  $\alpha$  the

Absorption coefficient  $\alpha$  = amount of sound energy absorbed / total incident sound energy

A sound absorber is a material which reduces air particle movement by friction. There is a range of commercially available sound absorbers – often in the form of some sort of tiles which can be fitted to the ceilings or walls. Some of the absorbers are porous absorbers of fibrous materials. Other absorbent materials used are perforated/porous boards, felts, asbestos, cloth, maps and pictures on the walls. The floor should be covered with matting's, carpets etc. to provide adequate absorption to reduce the reverberation time to the required value.

## 5.3 Hi-Fi Stereo

### 5.3.1 High – Fidelity or Hi-Fi system

when sound is reproduced with a high degree of similarity to the original sound or live sound, it can be said to be of high fidelity or simply HI-Fi. The basic requirements needed for a system to consider it as Hi-Fi are

- (a) Signal to noise ratio should be better than 50dB
- (b) Frequency response should be flat within  $\pm 1$ dB over the frequency range of 40 Hz to 15 KHz
- (c) Stereophonic effect should be provided

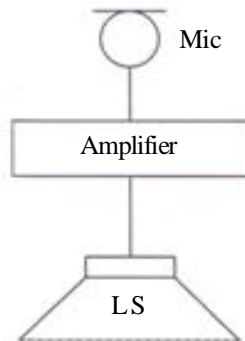


Fig 5.2 Monophonic sound system

### Difference between Hi-Fi and stereo

High-Fidelity system pertains to the basic quality of reproduction of sound. Stereo specifies a system of two channel reproduction to produce the three dimensional space affect in reproduction of sound. Stereo system is not a substitute for Hi-Fi. In the same way, a stereo system will not be a Hi-Fi system unless both the channels used for stereo reproduction are individually Hi-Fi channels. Thus, Hi-Fi added to a stereo system makes it the most natural and satisfying system of sound reproduction.

### Multy way speaker system

A single loudspeaker cannot have flat frequency response for the whole audio frequency range from 16 Hz to 20 KHz, and not even for the practical Hi-Fi range from 40Hz to 15KHz. Therefore the frequency spectrum is divided into at least 2 and 3 parts. Separate speakers are designed for each part, so that each speaker has to cover only a small range of frequency. The speakers which cover low frequencies from 16Hz to 1000 Hz are called **woofers**. The speakers which cover higher audio frequencies are called **tweeters**. Many a time, a third speaker, called squawker is used for mid-frequency coverage from 5000 Hz onwards.

Woofers and tweeters can either be separate speakers mounted in a common enclosure, or there can be a dual cone loudspeaker.

### Cross-over network

A cross-over network divides the incoming signal into separate frequency ranges for each speaker. In the absence of crossover networks, the speakers will suffer overheating and the output will be distorted when full power at frequencies outside their range is fed to them. The overall efficiency will be much reduced in the absence of crossover networks.

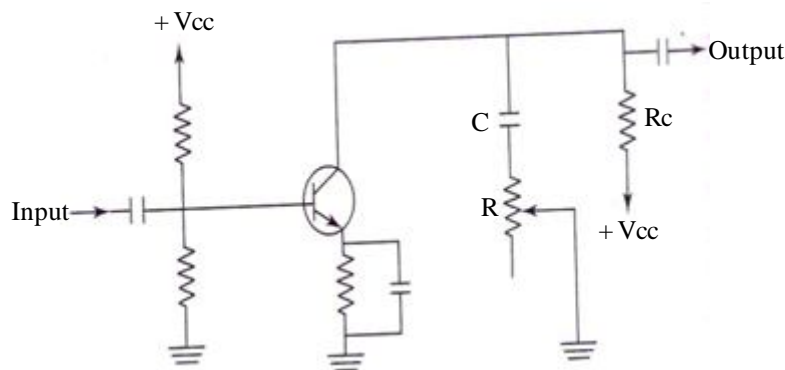
to the listener. The circuit used to compensate for such variations is called Balance control. A balance control is a simple potentiometer.

**(c) Loudness control :** The control which provides desired big boosting at bass and a little boosting at treble is called loudness control. It boosts audio by + 12dB at 50Hz and + 3dB at 10 kHz. The loudness control should be used only when sound level is low. Some times, the volume control is so designed that it raises the level of audio in accordance with ear's logarithmic response. Such a volume control also acts as loudness control and is known as contour control.

**(d) Tone controls :** To cater to the individual's taste and also to offset the effect of noise present in the signal, provision of bass and treble controls is made. The combined control is called tone control.

## 5.5 Simple circuit showing Basss and Treble control

**Treble control** A simple treble control is shown in Fig 5.5



**Fig 5.5 Basic Treble Control**

The potentiometer R in series with the capacitor C forms the treble control. When the slider is at the lower end, maximum signal develops across RC (load). As the slider is moved upwards, less and less resistance of the potentiometer comes in series and hence, there is more and more cut in the high-frequency signals. Cut is maximum when the slider is at the top end, short-circuiting the potentiometer completely. This position is called treble cut. The other position where treble cut is minimum is called treble boost. The capacitor alone will have low reactance for high frequencies.

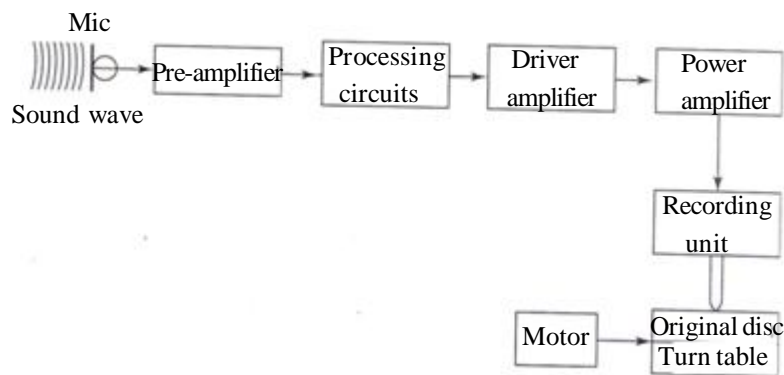
### 5.5.1 Bass control

Bass would be cut if capacitive reactance in series of signal increases. Lower the capacitance, greater will be the reactance ( $X_C = 1/2\pi fC$ ). Hence for

When the needle of the record player tracks the recorded grooves on the disc, it vibrates in accordance with lateral variations of grooves. These vibrations move a magnet placed in the magnetic field permanent magnet. Due to vibrations of the magnet, the flux density through a coil, placed in that magnetic field, changes and hence an emf is induced in the coil depending on the rate of change of flux. The electrical signal is amplified by transistor amplifiers and is fed to a loudspeaker to get original sound.

### 5.6.1 Block diagram of a disc recording system

Audio signals in the output of a microphone are processed and amplified to drive a disc-recording unit to cut grooves in the lacquer compound of the disc. Fig 5.8 depicts the block diagram of a disc-recording system.



**Fig 5.8 Block Diagram of Disc – Recording System**

#### Function of Each Block

**Microphone :** A high – grade microphone like condenser microphone or a specially designed cardioid microphone is used to convert the sound waves into electrical variations called audio signals.

**Pre-amplifier :** It amplifies the weak output of the microphone. It is a low noise high-gain amplifier to get high signal to noise ratio.

**Processing circuits :** The amplified signals are processed to de-emphasis low frequency signal and emphasis high frequency signal to eliminate chance of over – modulation for low frequencies and improve signal- to noise ratio for high frequencies

**Drive amplifier :** The processed signals are further amplified for voltage amplification so that high output voltage is available to drive the next stage to give adequate power.

**Function of each block**

**Motor :** It is a synchronous motor and rotates the turn table with steady speed its body vibrations are mechanically filtered to ensure elimination of rumble noise speed steadiness reduces wow flutter noise.

**Turn table :** It carries the play back disc over itself, and hence rotates it. **Disc** playback disc also called record consists of a mixture of shellac rags carbon black, etc it has the recorded grooves on its surface. When playback needle moves radially over the rotating disc it tracks the spiral grooves.

**Stylus :** It is the play back needle whose function is to track the grooves of the record disc and transmit the resultant vibrations to the cartridge transducer

**Cartridge :** It converts vibration received from the stylus into electrical signals of the same frequency (called audio signals). The magnetic cartridge gives signal of the order of 10mv for grooves of 25  $\mu\text{m}$  radius.

**Pre- amplifier :** It amplifies the weak output of the cartridge

Equalizer it equalizes the signal by emphasizing low notes and de-emphasizing high notes. Volume control, bass and treble controls are also generally located after the pre amplifier

**Driver Amplifier :** It further amplifies the signal to give sufficient input to the power amplifier to drive it for optimum power.

**Power Amplifier :** It gives power amplification to the signal. It also has a matching transformer to match source- impedance with the impedance of the load (i.e loudspeaker)

**Loud speaker :** It converts audio power into sound

---

**5.8 Principle of Magnetic Recording and Reproduction**

---

Magnetic recording is storage of the sound pressure variations in the form of elementary magnets (formed in a magnetic material) whose length and strength depend on audio frequency currents.

Magnetic recording is based on the principle that certain materials (like iron oxide) when brought in a magnetic field, get magnetized and retain that magnetism permanently until altered the various steps involved in magnetic recording are described below. Fig 5.10 shows the basic concept of magnetic recording on Tape.

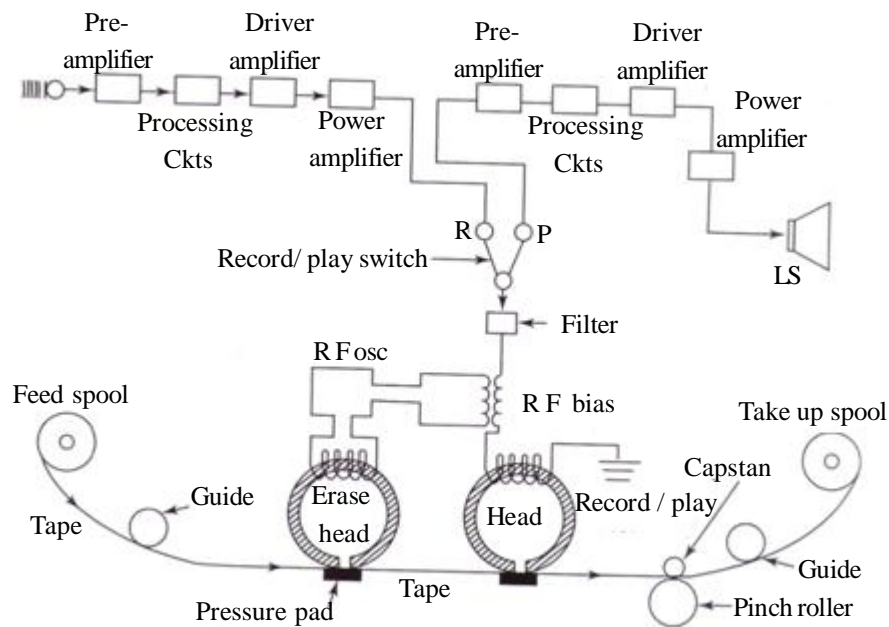


Fig 5.11 Block diagram of a tape recorder

### Function of each Block

**Microphone :** Sound waves strike the diaphragm of the microphone which converts the sound-pressure variations into electrical signals, called audio signals.

**Pre-amplifier :** It amplifies the weak output of the microphone. Its noise figure is low .

**Processing Circuits :** These circuits in the record section control the gain and level of recording and also provide de-emphasis and pre-emphasis for low frequency and high frequency audio signals, respectively.

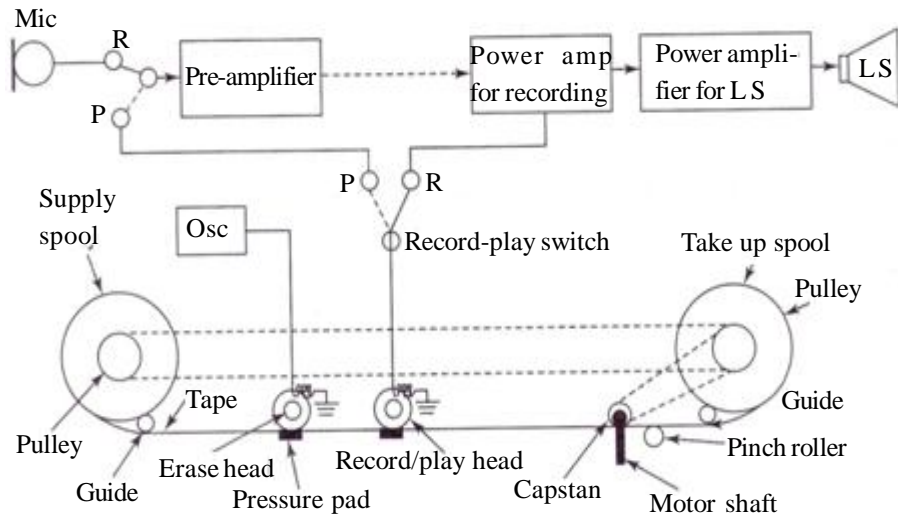
**Driver Amplifier :** It gives further voltage amplification to the signal so as to reduce the internal resistance of the power of the power amplifier and hence, to drive it to give power amplification.

**Power Amplifier :** It amplifies the power of the audio signal so as to drive the record head.

**Filter:** It is trap circuit which does not allow the bias oscillator's signal to go to the amplifier as it will unnecessarily get overloaded..



voltage or load. This steadiness results in reduction of wow and flutter distortion. In good tape machines, wow and flutter are not more than 0.2% speed of motor for fast rewind and fast forward can be changed with the help of speed gears.



**Fig 5.12 Transport mechanism for cassette recorder**

Motor rotations are transferred to the capstan flywheel assembly by means of a rubber belt. It prevents motor vibrations from reaching the capstan and thus reduces the rumble noise.

**Capstan and press (or pinch) Roller :** Capstan is a spindle machined accurately, and pulls the tape past the heads. The tape is pressed against the capstan by means of a rubber-covered pinch roller.

**Flywheel :** It is a heavy wheel made of metal and is fitted to the capstan shaft. This damps minor variations in the speed. It should be free from any tendency to vibrate, because any vibration here will cause a rumbling problem.

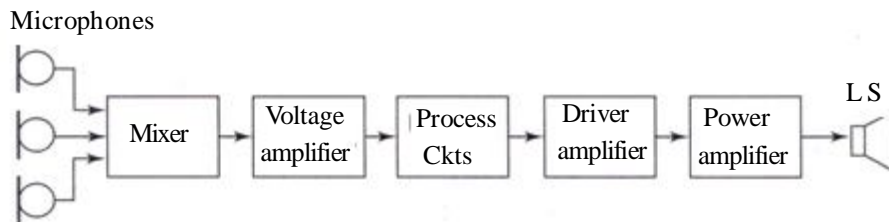
**Tape-guides :** These provide the desired tension in the tape and keep it in the correct position. The angle round which the tape should turn at any point in the transport should not be excessive. All the bearings over which the tape passes must be of high quality.

**Spools :** There are two spools. While in use, one spool feeds the tape to the other spool, and hence, the spools are known as

- i. Feeds spool or supply spool, and
- ii. Take – up spool.

### 5.11 Basic PA system

It is an electro acoustic system in which sound is first converted into electrical signals by a microphone. The electrical audio signals are amplified, processed and fed to another transducer called the loudspeaker. The loudspeaker converts the audio signals into sound waves. The block diagram of a basic PA system is shown in Fig 5.13



**Fig. 5.13 Basic Block Diagram of PA system**

**Microphone :** it picks up sound waves and converts them into electrical signals called audio signals. Generally, amplifiers have provision of two or more microphones and in addition, an auxiliary input for tape/ record player

**Mixer :** The output of microphones is fed to a mixer stage. The function of the mixer stage is to effectively isolate different channels from each other before feeding to the main amplifier. It can be either a built in unit or a pluggable unit.

**Voltage amplifier :** It further amplifies the output of the mixer

**Processing Circuits:** These circuits have master gain control and tone controls (bass/ treble controls)

**Driver Amplifier :** It gives voltage amplification to the signal. Thus, It drives the power amplifier to give more power

**Power Amplifier :** It gives desired power amplification to the signal. Generally, class B push- pull type of circuits are employed for this purpose. The output is given to the loud speaker through an impedance matching transformer to obtain maximum sound output

**Loudspeaker :** It converts audio signals into sound waves.

### 5.12 Typical PA system plan for college sports

The annual event of college sports requires coverage to the teachers and guests on the entrance side, to the spectator students on the opposite side

by three subjective characteristics (a) Pitch (b) Loudness (c) Quality.

3. An unwanted sound present in the environment, or coming out of the loudspeaker in an audio system is called 'noise'. Noise consists of pressure variations of random nature, without any regularity of frequency, shape, and amplitude, and also without continuity. It produces unpleasant sensation in our ear due to abruptness and sharpness. It has no definite regularities.

4. Like light waves, sound is also reflected and diffracted. Reflection causes reverberation. Diffraction causes bending around obstacles.

5. Refraction is a general property of the wave motion. As sound travels in wave, it is expected that when it falls on the surface of separation of two media, where is a difference in density and hence difference in velocity of propagation of sound, there will be change in the direction of propagation in the two media, except when the incidence is normal to the surface of separation.

6. Stereo effect is produced by the fact that listening is done by both the ears. The sound reaching the two ears from any point travels slightly different distances and is also of slightly different intensity. This difference of distance (or phase) and intensity of sound is computed by the brain to give the feeling of depth and direction which is the stereo effect. Simulating the stereo effect in actual practice will, therefore, require two separate recording or reproducing channels complete with two microphones, two amplifiers and two loudspeakers.

7. A single loudspeaker cannot have flat frequency response for the whole audio frequency range from 16 Hz to 20 KHz, and not even for the practical Hi-Fi range from 40Hz to 15KHz. Therefore the frequency spectrum is divided into at least 2 and 3 parts. Separate speakers are designed for each part, so that each speaker has to cover only a small range of frequency. The speakers which cover low frequencies from 16Hz to 1000 Hz are called **woofers**. The speakers which cover higher audio frequencies are called **tweeters**. Many a time, a third speaker, called **squawker** is used for mid-frequency coverage from 5000 Hz onwards.

8. A cross-over network divides the incoming signal into separate frequency ranges for each speaker. In the absence of crossover networks, the speakers will suffer overheating and the output will be distorted when full power at frequencies outside their range is fed to them. The overall efficiency will be much reduced in the absence of crossover networks.

9. All materials absorb sound to some extent. Hard inflexible substances with shining surfaces may absorb very little sound. The term absorption coefficient is used to represent absorbent properties of materials. If the absorption

4. What are the basic requirements of a Hi-Fi system ?
5. What is a Public Address system ?
6. Distinguish between Hi-Fi and Stereo systems.
7. Write the need of a crossover network in an acoustic system.
8. Draw the block diagram of a PA system.
9. Draw the circuit diagram of a two way cross over network for speakers.
10. Mention the various controls of a stereo amplifier
11. What is the difference between a woofer and a tweeter?
12. Give the basic principle of Disc reproduction system
13. Write the basic principle of reproduction of magnetic recording of sound.
14. Write the main components of tape transport mechanism used in tape recorders.

---

### Long Answer Type Questions

---

1. What is a PA system? Give its main requirements.
2. Distinguish between monophonic and stereophonic sound systems with a schematic diagram.
3. Explain in detail about Bass and Treble controls used in a stereo amplifier with a typical circuit diagram.
4. Draw the basic block diagram of a PA system and explain the function of each block.
5. Draw the block diagram of a disc recording system and explain the function of each block.
6. Draw the block diagram of a disc reproduction system and explain the function of each block.
7. Explain in detail about magnetic recording and reproduction with a schematic diagram.
8. Draw the block diagram of a tape recorder and explain the function of each block.



## UNIT

## 6

**Modern Disc  
Recorders/Players****Structure**

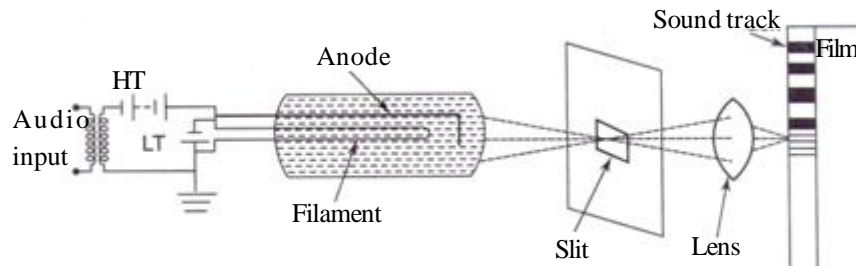
- 6.1 Types of optical recording of sound
- 6.2 Methods of optical recording of sound on film
- 6.3 Method of optical recording of sound
- 6.4 Advantages and disadvantages of CD's
- 6.5 Optical Pick-up system
- 6.6. Working of Compact Disc Player
- 6.7. MPEG Standards
- 6.8 Comparison of VCD and DVD
- 6.9 DVD Player
- 6.10. Dolby's Method of noise reduction
- 6.11. Blu-ray disc technology

**Learning Objectives**

After completing this unit, the student will be able to understand

- Different types of optical recording of sound
- The method of optical recording of sound on films
- The method of optical recording of sound on compact discs

series with the audio voltage as shown in Fig 6.1 the filament of the lamp is connected to a low dc voltage (called IT).

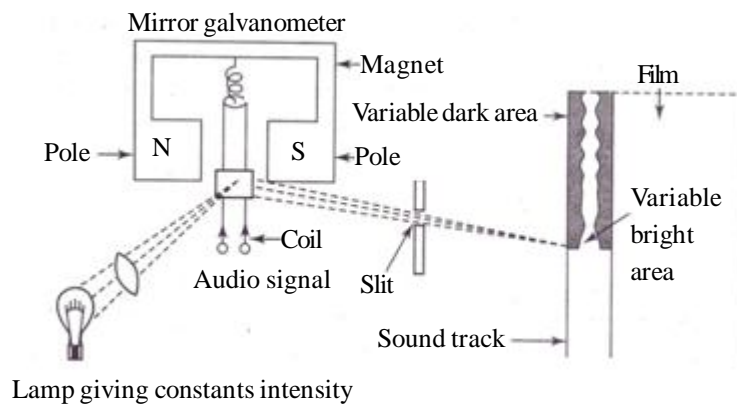


**Fig 6.1 Variable- density method of optical recording**

The intensity of light coming out from the lamp varies in accordance with the audio signal. This varying light passes through a slit and a focusing lens. The focused light falls on a moving photographic film where the image is recorded in the form of bars of varying density on the film.

### 6.2.2 Variable Area Method

In this method, light of constant intensity falls on a slit. The area of the slit opened for this light varies in accordance with the variations of sound pressure. Hence, the light falls on the variable area on the soundtrack edge of the film. Thus, the area which is bright to light varies. The area of the slit is made variable with the help of a mirror or galvanometer as illustrated in Fig 6.2



**Fig 6.2 Variable area method of optical recording**

Sound is first converted into electrical (audio) signals by a microphone. The audio signals are amplified and reach the coil of a mirror galvanometer. The current-carrying coil is placed in a magnetic field and hence, deflects in

## 6.4 Advantages and Disadvantages of CD's

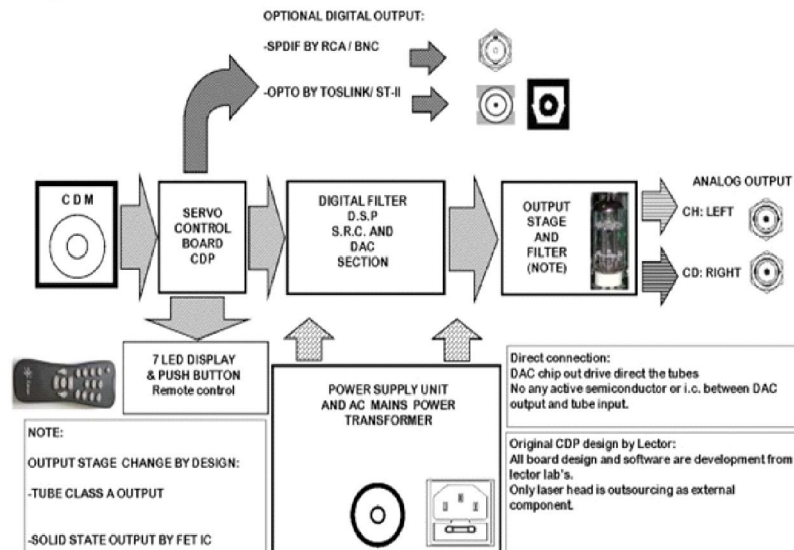
### 6.4.1 Advantages of compact discs

1. As it is covered by plastic or transparent lacquer, the tracks and recordings remain safe and are not affected by dust, grease and scratches. A compact disc is immune to surface contamination.
2. Signal-to-noise ratio is high, as high as 90 dB, an improvement of 30 dB over a high-fidelity gramophone disc.
3. Dynamic range is high, as high as 90dB, an improvement of 35 dB over a high-fidelity gramophone disc.
4. Channel separation is high, as high as 80 dB, an improvement of 50 dB over gramophone discs.
5. Wow does not exist
6. Flutter does not exist
7. Total distortion is low
8. Frequency response is excellent and covers complete audio range from 20Hz to 20KHz within only + 0.5 dB.
9. Size is quite small.
10. Drop-outs up to 2.5nm of disc (4000 bits) are not felt due to error-correction codes, and distortion due to scratches are also automatically corrected.
11. As the audio signals are converted into binary digits, the system has all the advantages of digital system over analog ones, for example,
  - (i) Pules can be regenerated and hence, any noise introduced is automatically eliminated, and
  - (ii) No equalization is require as pules are free from noise.

### 6.4.2 Disadvantages

1. Earlier, the cost of a compact disc was more than the high-fidelity analogue disc. Recently, the cost has been substantially reduced due to popularity of the new video-audio discs.
2. Compared to tapes, it has another disadvantage; the recording cannot be erased, and hence fresh recording cannot be done on the same disc without involving a complex and costly process.





**Fig. 6.5 Block Diagram of a compact disc player**

Much of a CD Player will comprise various servo systems that will enable the laser beam to be accurately focused on to the surface of the disc and simultaneously track the laser beam across the surface of the disc, whilst it is rotated at the correct speed. All CD players are required to operate in a specific manner, the sequence of which is controlled by some form of system control.

Most of the electronic operations take place within LSI circuits, which despite their complexity are usually reliable. Mechanical operations are relatively simple and are limited to motors to drive the disc, optical assembly and the loading and the unloading mechanism, and a pair of coils which will enable the lens within the optical assembly to move vertically and laterally.

The optical assembly contains a laser diode to provide a laser beam that is reflected from the surface of the CD through certain components within the block onto a photo diode array which will produce a range of signals that are in fact the effect of the reflected light variations from the 'Pit' or 'bump' information on the playing surface of the CD, which in turn becomes the current variations from the from the photo diodes contained in the within the photo diode array.

These current variations in due course produce a range of signals that will represent the music information or data from the disc, together with focus, tracking and spindle or disc speed information that will enable the disc to be played in the correct manner. Linked to the optical assembly is the automatic

## 6.8 Comparison of VCD and DVD

Parameter	VCD	DVD
1. Material used	Plastic	Plastic
2. Immunity to scratches, grease, etc.	Immune	Immune
3. Diameter	120mm	120mm
4. Thickness	1.2mm	1.2mm
5. Pitch for tracks	1.6µm	0.74µm
6. Minimum pit length	0.834µm	0.4µm
7. Laser's wavelength	780nm	635 or 650 nm
8. Technology	Pits and flats formed by laser beam (of infra-red)	Pits and flats formed by laser beam (of infra-red)
9. Capacity	700MB	4.7 GB TO 17 GB
10. Resolution		
NTSC	350 x 240	720 X 480
PAL	352 x 288	720 X 576
11. Video compression	MPEG-1	MPEG-2
12. Video bit rate	1150 kb/s	9000 kb/s
13. Duration of video recording	About 80 minutes	8 hours using both sides with 2 layers on each side
14. Compatibility	Good	Very good
15. Computer usage	Low	High
16. Quality	good	Very good

**Table 6.2. Comparison of VCD and DVD**

environment so that the board does not become contaminated by dust. All the primary components of the electronic circuit should be made out of silicon.

### Working of DVD Player

The pits and bumps in the DVD are hit by the laser from the optical mechanism of the DVD player. This laser will be reflected differently according to the change of pits and bumps. Though the laser hits a single spot, the DVD moves in a circular motion so that the entire area is covered. Mirrors are also used to change the spot.

These reflected laser beams are then collected by a light sensor (eg. photo-detector) which converts the different signals into a binary code. In short, the optical system helps in converting the data from the DVD into a digital code.

The binary signal is then sent to a Digital to Analog converter which will be setup in the PCB. Thus the corresponding analog signal of the DVD is obtained. The PCB also has amplifiers which amplify the signal and then sends it to the graphic and audio systems of the computer/TV. Thus, the corresponding audio/video signal is obtained.

### 6.10 Dolby's Method of noise reduction

In normal pre-emphasis, it is presumed that weak intensity is present only in high frequencies. This is not always the case. All weak signals, irrespective of frequencies need to be emphasized. This difficulty was solved by Dolby as explained below.

When the strength of signal falls below a pre-determined level the circuits boost the strength before recording. All signals which are 40 dB or higher pass through the Dolby system directly without any change. The lower level signals pass through the boosting stages which boost these signals by 10-15 dB. Boosting is done before recording. A signal in the absence of boosting is shown in fig 6.5

(a). After boosting, the recording noise remains unchanged but the signal is boosted as shown in Fig 6.7

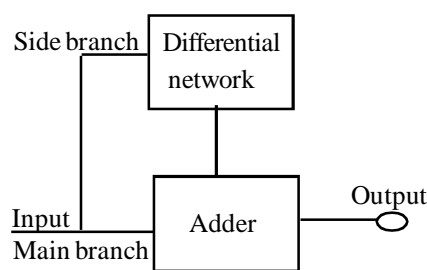
(b) During Playback, signal and noise both are reduced as shown in Fig. 6.7

(c) Indicates that signal-to-noise ratio is finally improved

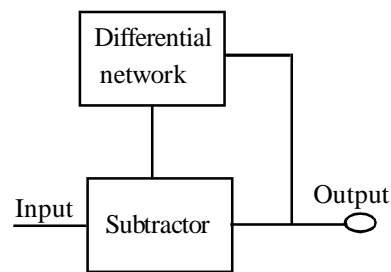
a low-pass filter which causes improvement in signal-to-noise ratio with respect to hum and rumble. The 80 Hz to 2999 Hz signal goes to a band-pass filter which deals with the mid-band noise. Most of the sound energy for music is concentrated in the band.

The 3000 Hz and 9000 Hz high pass filters improve signal-to-noise ratio with respect to hiss and modulation noise. The output of the four separate units is added. All this is done in a side branch and this branch is known as the differential network. The output of the differential network goes to the adder of the main branch as shown in Fig. 6.8 the output of the adder of the Dolby processed signal.

In playback. The differential network separates out the boosted signals in the side branch and subtracts them from the input signal as shown in Fig 6.9



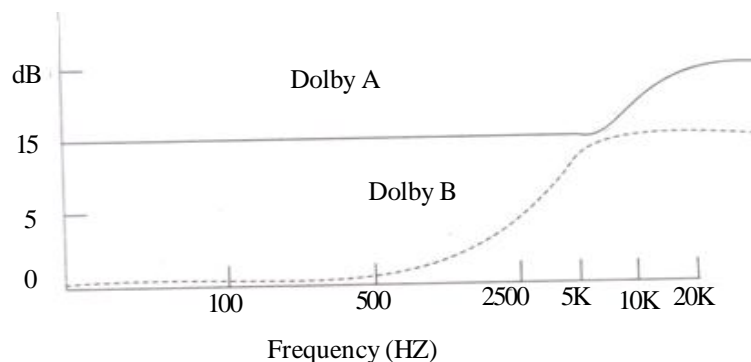
**Fig 6.8 Coding of Dolby signal**



**Fig 6.9 Decoding of Dolby signal**

The decoded output signal consists of the original signal with noise suppressed as shown in Fig

The dolby A system gives a improvement of 10dB in the signal-to-noise ratio by up to about 5 KHz, increasing it as the frequency increases until it becomes 15 db at 15KHz. It then remains at 15dB for frequencies higher than 15kHz this is shown by the curve A (full-line curve) in Fig.6.10



**Fig 6.10 S/N ratio improvement in Dolby systems**

extra capacity combined with the use of advanced video and audio codecs will offer consumers an unprecedented HD experience.

While current optical disc technologies such as DVD, DVD±R, DVD±RW, and DVD-RAM rely on a red laser to read and write data, the new format uses a blue-violet laser instead, hence the name Blu-ray. Despite the different type of lasers used, Blu-ray products can easily be made backwards compatible with CDs and DVDs through the use of a BD/DVD/CD compatible optical pickup unit. The benefit of using a blue-violet laser (405nm) is that it has a shorter wavelength than a red laser (650nm), which makes it possible to focus the laser spot with even greater precision.

This allows data to be packed more tightly and stored in less space, so it's possible to fit more data on the disc even though it's the same size as a CD/DVD. This together with the change of numerical aperture to 0.85 is what enables Blu-ray Discs to hold 25GB/50GB. Recent development by Pioneer has pushed the storage capacity to 500 GB on a single disc by using 20 layers.

## Summary

### 1. Optical recording of sound is of two types

#### 1. Recording on Photographic Films

#### 2. Recording in compact Discs

2. There are two methods for varying the intensity of light in accordance with the sound pressure variations – 1. Variable – density method, and 2. Variable-area method. These two methods are discussed below.

3. In the variable density method, sound is picked up by a microphone, and converted into electrical signals which are amplified. Audio output of the amplifier is fed to the anode of a special type of vacuum tube, called an AEO lamp. The lamp contains a little quantity of helium gas. The anode gets high dc voltage (called HT) in series with the audio voltage the filament of the lamp is connected to a low dc voltage (called IT). The intensity of light coming out from the lamp varies in accordance with the audio signal. This varying light passes through a slit and a focusing lens. The focused light falls on a moving photographic film where the image is recorded in the form of bars of varying density on the film.

4. In the variable area method, light of constant intensity falls on a slit. The area of the slit opened for this light varies in accordance with the variations of sound pressure. Hence, the light falls on the variable area on the soundtrack

**Short Answer Type Questions**

1. List the types of optical recording of sound.
2. Explain briefly about the optical pickup system.
3. What is a compact disc ?
4. What are advantages of CDs ?
5. What are the disadvantages of CDs ?
6. What is digital dolby system ?
7. Explain the working principle of DVD player.
8. Draw the block diagram of DVD player.
9. Distinguish between MP3 format and Audio CD format.
10. Write any four differences between VCD and DVD
11. What is the advantage of Blu-Ray format compared to conventional DVD format ?
12. Abbreviate the following terms (a) CD (b) VCD (c) DVD (d) MPEG

**Long Answer Type Questions**

1. Explain how sound can be recorded on a film.
2. Explain the recording of sound on a disc by a laser beam.
3. What are advantages and disadvantages of Compact Discs?
4. Explain in detail about the Dolby system of noise reduction.
5. Explain the working of Compact Disc player with a block diagram.
6. Draw the block diagram of DVD player and explain its operation.
7. Differentiate between CD and DVD.
8. Give an idea of Blu-Ray Disc technology.