T. Y. B. Sc. Electronic Science

Paper I: Semester IV

Advanced Communication systems

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Acknowledgment

Major problem faced by T. Y. B.Sc. students is availability of text books and most of them are reference books. Sometimes books may not be available for the students. Hence SPEED (Society for Promotion of excellence in electronics discipline) has taken a step forward to help these students. SPEED has contributed by online publishing reference notes for the "Advanced Communication Systems" for T. Y. B.Sc. Electronic science students. These notes will really be helping them for understanding the subject as well as preparation for examination. We are thankful to Dr. S. N. Khan, Associate professor, Pemraj Sarda college,Ahmednagar for his major contribution in preparation of notes. Such contribution from staff members is really appreciable.

Dr. A. D. Shaligram

Chairman, SPEED

Paper I: Semester IV

EL-341: Advanced Communication Systems

Unit 1: Antenna & Propagation [14]

Antenna**:** Basic consideration, Evolution of Dipole antenna, Parameters of Antenna, Effect of ground on Antennas. Resonant Antenna- Radiation patterns & length considerations, Non-Resonant antenna, Directional high frequency antennas, UHF & Microwave antenna, Wide-band & special purpose antennas

Propagation of Waves**:** Ground (Surface waves), sky wave propagation, space waves, Tropospheric scatter propagation.

Unit 2: Modulation & Demodulation [12]

Balanced Modulator- Using diodes & FETs

SSBSC- Filter Method, Phase shift method (third method)

Synchronous Demodulation, Product Demodulator,

Phase modulation & demodulation using PLL, Indirect method of FM generation.

Unit 3: Transmitter & Receiver [12]

AM transmitters: Block diagram,

FM Transmitters: Using Frequency multiplication & mixing, Frequency stabilized reactance FM transmitter, FM achieved through phase modulation

TV transmitter (monochrome/colour) Mobile receiver block diagram (800MHz), Doppler RADAR, Speed Gun, Low noise amplifier block diagram

Unit 4: Digital Communication [10]

Pulse modulation, Pulse code modulation, Differential Pulse Code Modulation, Delta modulation, Adaptive delta modulation, Companding, TDM, FDM, Vocoders

Block diagram- Digital Communication System

Recommended Books:

1. Electronic Communication By Dennis Roddy & John Coolean, Pearson Education
2. Principles of Communication Systems By Taub Schilling, McGraw Hill.
3. Antenna Theory: Design & Analysis By Balanis, Wiley Eastern
4. Electronic Communication systems By Kennedy & Davis, Tata McGraw Hill

Chapt. 1 Antenna and Propagation.

1

Khan S.N. Sarda College Ahmednagar.

Antenna is a means for radiating or receiving radio waves.
(transmitting)

An antenna is a metallic object, generally a wire or group of wires, used to convert high frequency current into electromagnetic waves and vice versa. [i.e. Transmitting antenna and receiving antennas are reciprocal in function.] It is capable of radiating or receiving the electromagnetic waves.]

Antenna couples the transmitter output to the free space or received input to the receiver from free space.

Write short note on Evolution of Dipole. Describe

Evolution of Dipole:

Let us consider RF generator is connected to one end of transmission line and other side is left open. The forward and reverse travelling waves combines to form a standing wave forms pattern on the line.

At the open ends, all energy must be ideally reflected back. But practically a small portion of energy in electromagnetic form escape from open ends and gets radiated(*). To increase the amount of radiated power, the open circuit must be enlarged by spreading of two wires as shown in following figure (a).

Due to this arrangement, the coupling between the transmission line and free space is improved and as tips have gone away from each other, and amount of cancellation has reduced.

(*) The percentage of amount of energy radiated of total energy is very small. This is due to two reasons (a) mismatch between the transmission line and surrounding space (b) As two wires are close to each other, radiation from one tip will cancel the radiation from the other tip, because of opposite polarities.

The radiation efficiency will increases further, when the two wires are bent at 90^0 (right angles) to each other (fig. b). The electric and magnetic fields are now fully coupled to the surrounding space and maximum radiation ~~results~~ in space takes place.

This type of radiator is called a dipole. When the total length of the two wires is a half wavelength ($\lambda/2$), the antenna is called a half wave dipole.

This configuration has similar characteristics to its equivalent length transmission line ($1/4\,\lambda$). It results in high impedance (Hi Z) at the far ends reflected as low impedance (Lo Z) at the end connected to the transmission line (i.e. at centre). This causes the antenna to have a large current anti node at the center and large voltage anti nodes at the ends, resulting in maximum radiation

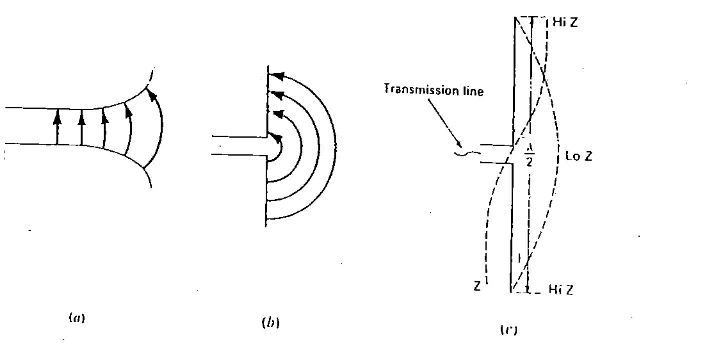
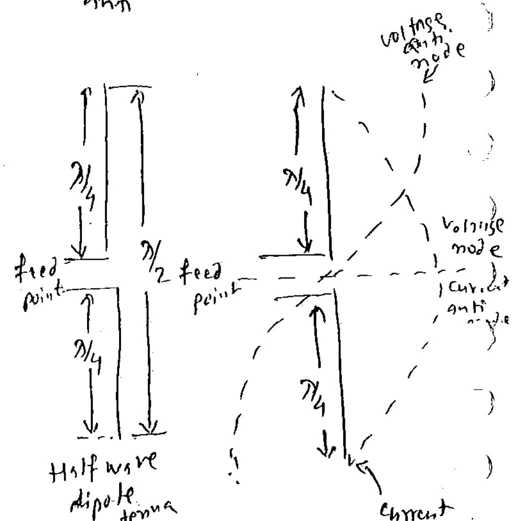


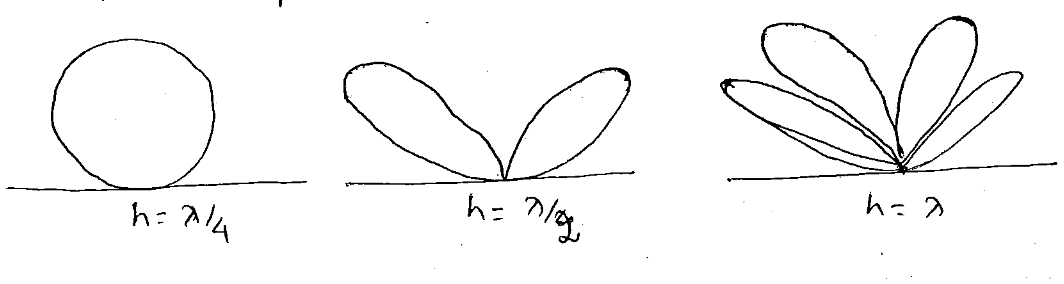
FIGURE 9-2 Evolution of the dipole. (*a*) Opened-out transmission line; (*b*) conductors in line; (*c*) half-wave dipole (center-fed).



voltage and current distribution in dipole antenna

The effects of ground on Antennas:

Following figure shows Radiation patterns of ungrounded half wave dipole located at varying heights above the ground.



(3)

With the help of diagram of ungrounded antenna & its image antenna Describe ungrounded Antennas.

Ungrounded Antennas:

When a radiation source (antenna) is placed near a reflecting surface (earth), the signal received at any distant point is the vector sum of the direct (sometimes called the incident) wave and the reflected wave. To explain this, let an image antenna is considred to exist below the earth's surface and it is a true mirror image of the actual antenna, as shown in the following figure.

When a wave is reflected its polarity is changed by 180°. If direct and reflected waves of equal magnitude and phase are received at exactly the same time, then the two signals will cancel out each other (the vector sum is equal to zero). In reality this condition is rarely obtained, but combination of this effect can cause reception to fade (if the signals are out of phase) or increases (if the signals are in phase i.e. voltage vector addition).

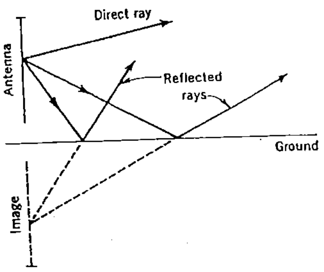


FIGURE 9-12 Ungrounded antenna and image.

Grounded Antennas:

If an antenna is grounded, the earth still acts as a mirror and becomes part of the radiating system.

The ungrounded antenna with its image forms a dipole array, but the bottom of the grounded antenna is joined to the top of the 4 images. Hence antenna acts as of double size antenna.

Thus in the following figure, a grounded quarter wave ($1/4\ \lambda$) vertical radiator (antenna) has effectively a quarter wavelength ($\frac{\lambda}{4}$) added, ~~by it~~ to it by its image. The voltage and current distribution of such a grounded vertical $\lambda/4$ antenna (commonly called the Marconi antenna) are

Describe grounded antenna. (4)

the same as those of the half wave dipole in space. (fig. 6)

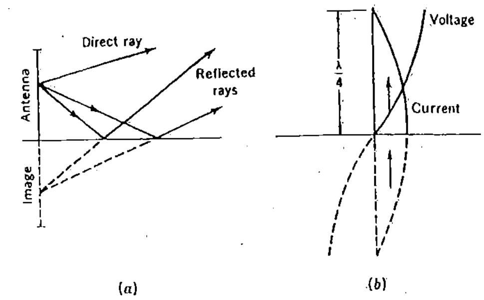
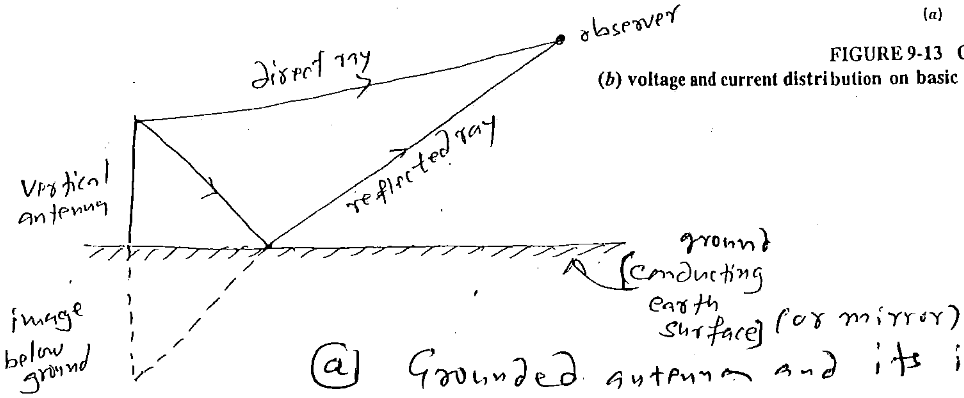
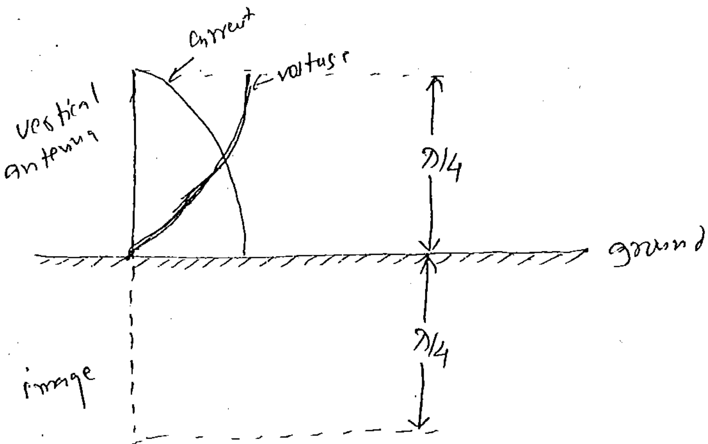
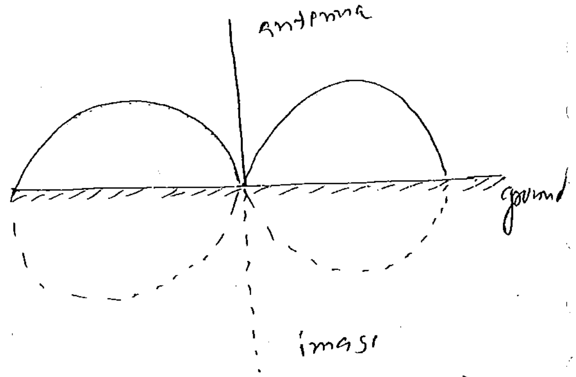


FIGURE 9-13 Grounded antennas. (*a*) Antenna and image; (*b*) voltage and current distribution on basic Marconi antenna.



(a) Grounded antenna and its image



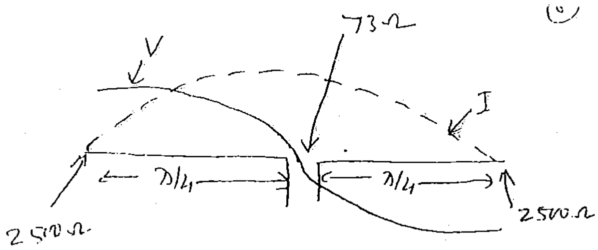


(b) Voltage and current distributions for grounded vertical $\lambda/4$ antennas. (marconi antenna)

(c) Radiation pattern of grounded $\lambda/4$ Vertical antenna.

The marconi antenna (vertical grounded $\lambda/4$ antenna) has one important advantage over the ungrounded antenna; to produce any given radiation pattern, it needs only half of the antenna.

Since ground plays important role in producing the required radiation patterns, the ground (soil) conductivity must be good. Where it is poor, an artifical ground is created by burying a mat of heavy conductors.



[The impedance varies along the length of antenna. It is highest where current is lowest and lowest, where the current is highest (at the centre. At the centre of half wave antenna the impedance is nearly 73Ω and is increases upto 2500Ω at either end.]

Resonant Antennas: Radiation Patterns and length considerations:

Draw the Radiation Patterns for different length of Resonant antennas.

The dipole antennas can be of different lengths, but the length will always be an integral multiple of λ/2. The radiation pattern depends on the length of the dipole antenna. Following figure shows the radiation pattern of dipole antennas of different lengths.

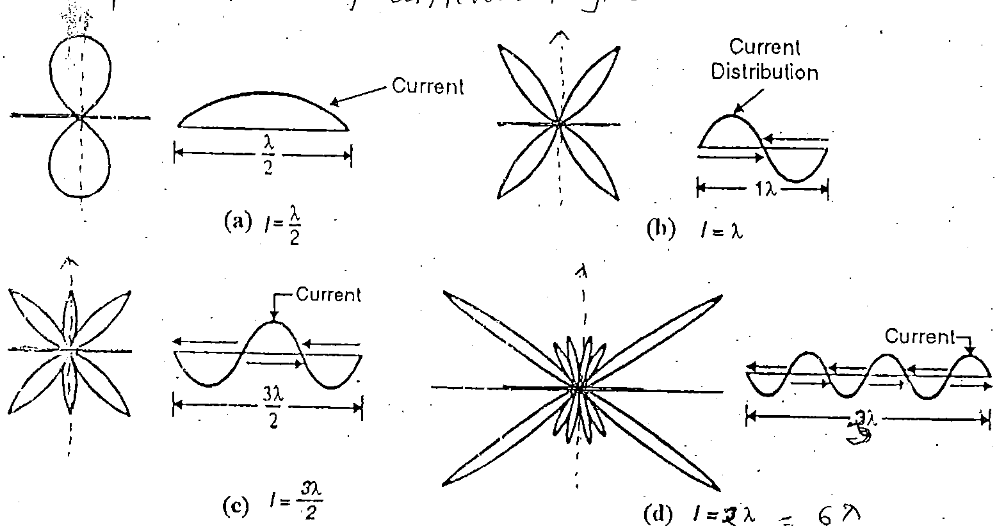


Fig. 10.4.2 : Radiation patterns of various resonant dipoles

Figure (a) shows standard figure eight pradiation patter for a half wave (length) dipole. Fig (b) shows for a full wave length. Figure (c) shows for 1½ wave length and fig (d) shows three wave lengths.

The half wave antenna has distributed capacitance and inductance and like a resonant circuit. It is The voltage and current will not be in phase. If RF voltmeter is connected end of antenna to ground, a large voltage will be measured. If the meter connected t at centre, meter shows zero volt.

The length of antenna can be calculated using expression ~~$L = \frac{c}{2f}$~~

$$L = \frac{Vel}{f} \times 0.95$$

Vel = speed of light = 3×10^8 m/sec, f = freq. in Hertz

V_f = velocity factor = 0.95 (also called end effect)

Explain marconi antenna with respect (a) antenna and its image, (b) voltage and current distribution (c) Radiation pattern.

The radiation pattern of a marconi antenna depends on its height and selection of pattern as shown in following figure. It is seen that horizontal directivity improves with height up to $5/8\lambda$. After that the pattern "lift off" the ground. It is shown in the figure, when length of marconi antenna (height) is λ.

Classification of Antennas:

Antennas are also classified into two types:

(i) Resonant antenna
(ii) ~~Nor~~ Non resonant antenna.

Draw the radiation patterns for $5/8\lambda$ & λ
& a marconi antenna $\lambda/4$ $\lambda/2$
its heigh $\lambda/4$,

What is resonant antenna?

Resonant antennas:

A resonant antenna is a transmission line the length of which is exactly equal to multiples of half wavelength ($\lambda/2$) and it is open at both ends.

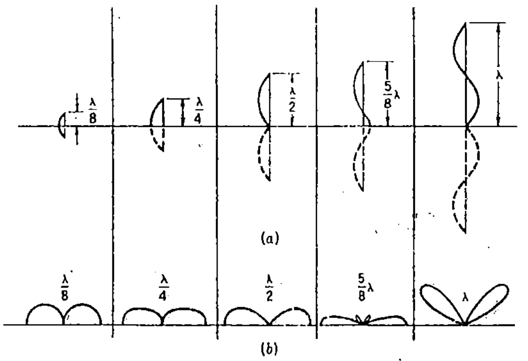
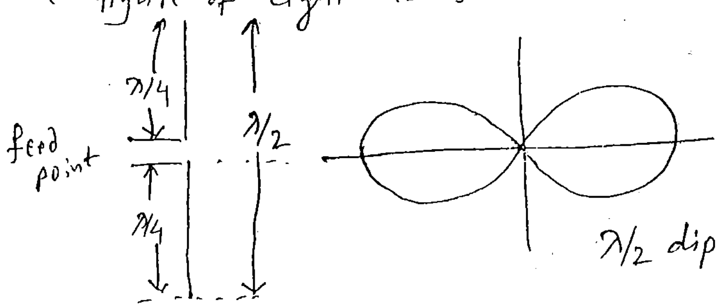


FIGURE 9-14 Characteristics of vertical grounded antennas. (*a*) Heights and current distributions; (*b*) radiation patterns.

· therefore the length of resonant antenna can be $\lambda/2$, λ, $3\lambda/2$ etc.

The radiation pattern of a $\lambda/2$ dipole resonant antenna is ~~ty~~ typically a figure of eight as shown in following figure.



$\lambda/2$ dipole resonant antenna

Write the difference between Resonant and Non resonant antennas.

(7)

Determine the length of an antenna operating at freq. of 500 KHz

$$\cancel{L = \frac{c}{f} = \frac{3\times10^8}{500\times10^3}}\ \ \cancel{\text{speed of light}}$$

$$L = \frac{Vel}{f} \times 0.95 = \frac{3\times10^8}{500\times10^3} \times 0.95 = 570m$$

Draw the diagram along with voltage and current distribution and Radiation pattern for non resonant antennas.

Nonresonant Antennas (Directional Antennas):

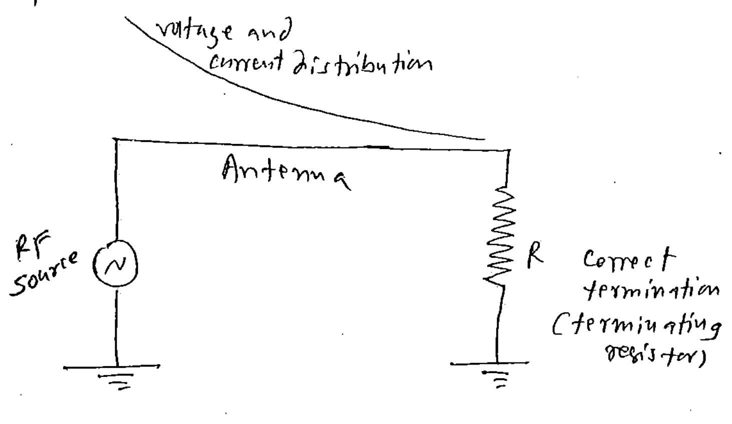
The length of these antennas is not equal to the exact multiple of $\lambda/2$

The non resonant antenna is the one, on which the standing wave is not present. The waves travel only in the forward direction. The standing waves are absent, because these antennas are terminated in correct impedance which avoids reflections (one End of antenna is grounded through resistor)

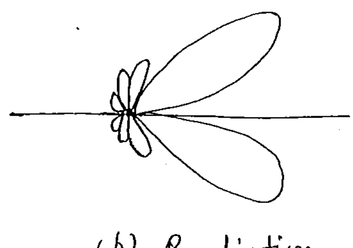
A nonresonant antenna like a properly terminated transmission line, produces no standing waves. They are suppressed by the use of a correct termination resistor and no power is reflected, ensuring that only forwarding traveling waves will exist.

In a correctly matched transmission line, all the transmitted power is dissipated in the terminating resistance.

When an antenna is terminated as in fig. (a) about two thirds of forward power is radiated and the remaining is dissipated in the antenna.



(a)



(b) Radiation Pattern

The ~~radiation~~ radiation pattern of non resonant antenna is shown fig. (b). It shows that the non resonant antenna is a unidirectional antenna.

What is difference between resonant and non resonant antennas.

Explain non-resonant antennas.

Demerits of non resonant antennas:
(1) Low gain (2) Low efficiency due to power dissipated in termination
(3) They occupy more space.

Parameters of antenna :-

Define (i) Directive gain
(ii) Directivity
(iii) power gain
Give relation between power gain and directive gain.

(A) Antenna gain:

(a) Directive gain (G_D) :-

It is defined as the ratio of power radiated by antenna in a particular direction to the power radiated by an isotropic antenna (standard reference antenna), provided that the input power is same for both antennas and power radiation of both antennas are measured at a same distance.

$$\text{Directive gain } G_D = \frac{\text{Power radiated by an antenna}}{\text{Power radiated by reference antenna (isotropic antenna)}}$$

(b) Directivity (D):-

The directive gain can be defined in any direction.
The maximum directive gain, can be obtained in only one direction, in which the radiation is maximum (along the larger lobe) is called Directivity.

$\therefore$ Directivity = maximum directive gain.

(c) Power gain (Ap):

Power gain is a comparison of the output power of antenna in particular direction to that of an isotropic antenna.

It is defined as ratio of power fed to an isotropic antenna to the power fed to a directional antenna, to develop the same field strength at the same distance, in the direction of maximum radiation.

$$\text{Power gain } A_p = \frac{\text{Power fed to the isotropic antenna}}{\text{Power fed to the directional antenna}}$$

comment.

Nonresonant antennas have higher directive gain than resonant Antennas.

The relation between the power gain and directive gain is given as

$$\boxed{A_p = \eta \cdot D}$$

A_p = Power gain
D = Directivity (maximum directive gain)
η = Antenna efficiency

When efficiency $\eta = 1$ for isotropic antenna (ideal loss less antenna) therefore power gain is equal to Directivity. in case of isotropic antenna.

(B) Antenna Resistance:

There are two resistances associated with the antenna.

(i) Radiation resistance (R_{rad})

(ii) Resistance due to the actual losses in the antenna (R_{loss})

Radiation resistance :- It is a (hypothetical) fictitious resistance, if replaced by an equivalent resistor would dissipate exactly the same amount of power that the antenna would radiate. Radiation resistance is an ac resistance.

Radiation resistance is defined as the power radiated by the antenna to the square of the current at the input of the antenna. (at the feed point).

$$R_{rad} = \frac{P_t}{I^2} = \frac{\text{Radiated power by the antenna}}{\text{square of current at the feed point}}$$

(C) Antenna losses and efficiency:

The Power input to antenna = Power radiated by antenna + Power loss by antenna

$$P_{in} = P_{rad} + P_d$$

$$I^2 \cdot R_{in} = I^2 \cdot R_{rad} + I^2 R_{loss}$$

Explain Antenna resistances.
obtaine the expression for η in terms antenna resistance.

Antenna efficiency is defined as the ratio of power radiated by antenna to the power fed to it.

$$\eta = \frac{P_r}{P_i} = \frac{\text{radiated Power}}{\text{input Power to antenna}} = \frac{\text{Radiated Power}}{\text{Power radiated + Power loss}}$$

$$= \frac{I^2 R_{rad}}{I^2 R_{rad} + I^2 R_{loss}}$$

$$\boxed{\eta = \frac{R_{rad}}{R_{rad} + R_{loss}}}$$

comment "The radiation resistance of antenna should be large" because the greater is the R_{rad}, greater is power radiated by the antenna. Hence efficiency (η) also increases.

R_{loss} = Resistance of antenna due to which, some power dissipated as heat.

The resonant antennas have lengths, which are multiples of half wave length ($\lambda/2$). Therefore Low and medium frequency antennas are least efficient because making them of resonant length (atleast $\lambda/2$) is practically impossible. These antennas have efficiencies of only 75 to 95 %

Antennas at higher frequencies can easily achive efficiency up to 100% and easily constructed.

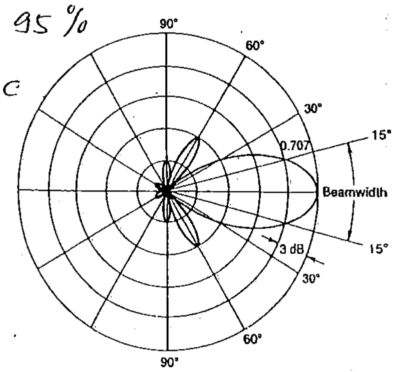


FIGURE 9-9 Beamwidth.

(D) Beamwidth and Bandwidth:

(i) Bandwidth:

The bandwidth of antenna is defined as the frequency range, over which, its operation is satisfactory (i.e. antenna radiate effectively)

It is the frequency difference between the half power points.

When the antenna power drops to 1/2 (or 3 dB) the upper and lower extremities of these frequencies have been reached then antenna no longer performs satisfactorily.

Define Bandwidth and Beam width.

The bandwidth of an antenna refers to the range of frequencies over which the antenna can operate correctly.

The bandwidth is equal to the difference between the frequencies at which the received power falls to half of the maximum, in the direction of maximum radiation.

Beamwidth:

The beamwidth of an antenna is the angular separation in degree between the two the half power points (3dB) on the main radiation lobe of radiation pattern.

In the figure, the beam angle is 30°, which is the sum of the two angles created at the points, where the field strength (radiated power) drops to 0.707 of the maximum voltage at the center of the lobe. (These points are known as half power points or 3dB points.)

"Beam width is a measure of directivity of an antenna" comment.

If smaller is the Beamwidth, more will be the directive lobe and vice versa.

Ex. 10.3.1 : To produce a power density of 1 mW/m² in a given direction at a distance of 2 km an antenna radiates a total of 180W. An isotropic antenna would have to radiate 2400 W to produce the same power density at that distance. What in dB is the directive gain of the practical antenna ?

Soln. :

$$\text{Directive gain, (D)} = \frac{\text{Power radiated by the isotropic antenna}}{\text{Power radiated by the practical antenna}}$$

$$\text{and directive gain in dB} = 10 \log D = 10 \log_{10}\left[\frac{2400}{180}\right] = 10\log_{10}(13.33)$$

$$= 11.25 \text{ dB.}$$

10×1.1248

$= 11.24 dB$

Ex. 10.3.2 : A practical antenna has a directive gain of 5 dB radiates 1200 W power. How much power an isotropic antenna should radiate in order to have the same power density at the same distance.

Soln. :

$$\text{Directive gain in dB} = 10 \log_{10}\left[\frac{\text{Power radiated by isotropic antenna}}{\text{Power radiated by practical antenna}}\right]$$

$$\text{Substituting the values,} \quad 5 = 10 \log\left[\frac{P_i}{1200}\right]$$

$$\therefore \quad P_i = 3794.7 \text{ Watt.}$$

$0.5 = \frac{P_i}{1200}$

$\text{Antilog}(.5) = \frac{P_i}{1200}$

$P_i = 3.162 \times 1200$

$P_i = 3794.7$ watt

3.162×1200

Antilog ~~0.5~~ $= 10 \times \frac{P_i}{1200}$

$\frac{1.6094 \times 1200}{10}$

$P_i = 3794.7$ watt

3794.7 watts.

$\frac{180}{2400} : 0.075$

$10\log_{10}[.075] = 10\log_{10}\left[\frac{75}{1000}\right]$

$10\log_{10}\left[\frac{1000}{75}\right]^{-1}$

Ex. 10.3.3 : Calculate the directivity for the antennas having following specifications :

(a) Power gain $10^3 : 1$, efficiency 90% (b) Power gain 45 dB, efficiency 90%.
find directive gain in dB

Soln. :

(a) The relation between the power gain, directivity and efficiency is,

$$A_p = \eta D \quad \text{...(1)}$$

Given that $A_p = 10^3 = 1000$ and $\eta = 0.9$

$$\therefore \quad D = \frac{1000}{0.9} = 1111.1 \quad \text{...(2)}$$

A_p in dB $= 10 \log_{10} A_p$

(b) Given that $A_p = 45$ dB

$\therefore$ Power gain in the form of ratio is given by $A_p = 31622.7$

$45 = 10 \log_{10} A_p$

$4.5 = \log_{10} A_p$

$\text{Antilog}(4.5) = \text{Antilog}(\log_{10} A_p)$

$\text{Antilog}(4.5) = A_p$

$\cancel{1.5040} \; 31622.7 = A_p$

$A_p = \eta D$

$D = \frac{A_p}{\eta} = \frac{\cancel{1.5040}\,31622.7}{.9} = 35136.3$

directivity in dB $= 10 \log_{10}(D) = 10 \log_{10}(35136.3) = 10 \times 4.5457 = 45.45$ dB

Ex. 10.4.1 : Calculate the length of a half wave dipole antenna designed to operate at a frequency of 100 MHz. Also find length of marconi antenna.

Soln. :

The wavelength, $\lambda = \frac{c}{f}$ where c = speed of light.

$$\therefore \quad \lambda = \frac{3 \times 10^8 \text{ m/s}}{100 \times 10^6 \text{ Hz}} \qquad \therefore \quad \lambda = 3 \text{ meters.}$$

$\therefore$ Length of the half wave dipole antenna is $\lambda/2 = 1.5$ meters.

length of marconi antenna $= \lambda/4 = \frac{3}{4} = 0.75$ meters

Ex. 10.3.5 : An antenna radiates 1200 W when it is drawing 10A current at its feedpoints. Calculate the radiation resistance of the antenna. Another antenna has a radiation resistance of 20 Ω. Which of the two antennas will you select and why ?

Soln. :

$$\text{Radiation resistance, } R_r = \frac{\text{Power radiated}}{\text{Square of current drawn}} = \frac{1200}{(10)^2}$$

$$R_r = 12\ \Omega \quad \text{...(1)}$$

Thus radiation resistance of the first antenna is 12 Ω. Now consider the second antenna with $R_r = 20\ \Omega$. Let us calculate the current drawn by this antenna to radiate the same amount of power.

$$\therefore \quad P = (I_2)^2 R_r$$

$(I_2)^2 = \frac{P}{R_{rad}}$

$$\therefore \quad (I_2)^2 = \frac{1200}{20} = 60$$

$$\therefore \quad I_2 = 7.74 \text{ Amp.} \quad \text{...(2)}$$

Thus the second antenna draws only 7.74 amp. current to radiate the same power hence it should be selected.

Calculate directivity of an antenna if ~~pow~~ power gain = 75 dB and efficiency 93%.

Ex. 10.4.2 : An antenna has a radiation resistance of 72 Ω, a loss resistance of 8 Ω and a power gain of 16. What is its efficiency and directivity ?

Soln. :

Given : $R_r = 72\,\Omega$, $R_d = 8\,\Omega$, $A_p = 16$

(i) Antenna efficiency, $\eta = \dfrac{R_r}{(R_r + R_d)} = \dfrac{72}{(72+8)} = 0.9$ or 90%. $= \dfrac{R_{rad}}{R_{rad} + R_{loss}}$

(ii) Power gain, $A_p = \eta D$

$$\therefore \quad D = \frac{A_p}{\eta} = \frac{16}{0.9} = 17.7$$

$\therefore$ Directivity is 17.7. Directivity D in dB $= 10 \log_{10} (17.7)$

$= 10 \times 1.247 = 12.47$ dB

Ex. 10.3.1 : To produce a power density of 1 m W/m² in a given direction at a distance of 2 km an antenna radiates a total of 180W. An Isotropic antenna would have to radiate 2400 W to produce the same power density at that distance. What in dB is the directive gain of the practical antenna ? Directive gain D

Soln. :

$$\text{Directive gain, (D)} = \frac{\text{Power radiated by the isotropic antenna}}{\text{Power radiated by the practical antenna}}$$

$$\text{and directive gain in dB} = 10 \log D = 10 \log_{10}\left[\frac{2400}{180}\right]$$

$$= 11.25 \text{ dB.}$$

Ex. 10.3.2 : A practical antenna has a directive gain of 5 dB radiates 1200 W power. How much power an isotropic antenna should radiate in order to have the same power density at the same distance.

Soln. :

$$\text{Directive gain in dB} = 10 \log_{10}\left[\frac{\text{Power radiated by isotropic antenna}}{\text{Power radiated by practical antenna}}\right]$$

Substituting the values, $5 = 10 \log\left[\dfrac{P_i}{1200}\right]$ $\quad 0.5 = \log_{10} P_i - \log_{10} 1200$

$\therefore \quad P_i = 3794.7$ Watt. $\quad .5 = \log_{10} P_i - 3.079$

$3.579 = \log_{10} P_i$

Antilog

Ex. 10.3.3 : Calculate the directivity for the antennas having following specifications :
(a) Power gain $10^3 : 1$, efficiency 90% (b) Power gain 45 dB, efficiency 90%. $P_i = 37$

Soln. :

(a) The relation between the power gain, directivity and efficiency is,

$$A_p = \eta D$$

Given that $A_p = 10^3 = 1000$ and $\eta = 0.9$ 45

$$\therefore \quad D = \frac{1000}{0.9} = 1111.1$$

age 175

(b) Given that $A_p = 45$ dB

$\therefore$ Power gain in the form of ratio is given by $A_p = 31622.7$

An experimental antenna has gain of 3dB above a reference antenna. How much power would the reference antenna have to radiate to provide the same signal picked up when 5W is radiated by experimental antenna?

$$\text{directional gain in dB} = 10 \log_{10}\left[\frac{\text{Power radiated by reference antenna}}{\text{Power radiated by experimental antenna}}\right]$$

$$3\,dB = 10 \log_{10}\left[\frac{\text{Power radiated by ref. antenna}}{5}\right]$$

$$0.3 = \log_{10}\left[\frac{\quad}{5}\right]$$

∴ power required to be radiated by reference antenna to have same signal strength.

$$\text{Power radiated by ref. antenna} = 5 \text{ Antilog}[0.3] = 5 \times 1.9952$$

$$= 9.976W \simeq 10W$$

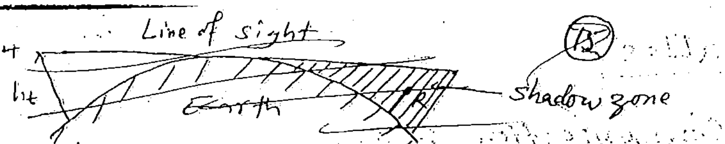
An antenna having gain of 3dB over a reference antenna is radiating 1500W. How much power must the reference antenna radiate in order to be equally as effective in most preferred direction?

$$\text{Gain in dB} = 10 \log_{10}\left[\frac{\text{Power radiated by refrence antenna}}{\text{Power radiated by given antenna}}\right]$$

$$3 = 10 \log_{10}\left[\frac{\quad}{1500}\right]$$

$$= 1500 \text{ Antilog}_{10}[0.3] = 1500 \times 1.9952$$

$$= 2992.89W \simeq 3000W$$



15

~~The shaded area is known as the shadow zone and it is region~~

Problem :— $1000 = [5]^2 R_{radiation}$

$\therefore$ Radiation resistance of the antenna $R_{radiation} = 40\,\Omega$

Ex. 8.14 : *How much current does an antenna draw when radiating 500 W if it has a radiation resistance of 300 Ω ?*

Sol. :

$$\text{Power radiated} = [I]^2 R_{radiation}$$

$$\therefore \quad 500 = [I]^2 [300]$$

$$\therefore \quad \text{Current } I = 1.29 \text{ A}$$

Ex. 8.15 : *An antenna having a radiation resistance of 50 Ω draws 8 A. How much power is it radiating ?*

Sol. :

$$\text{Power radiated} = [I]^2 R_{radiation}$$

$$= [8]^2[50]$$

$$= 3200 \text{ W}$$

Ex. 8.16 : *Calculate the radiation resistance of an antenna which is fed 12 A when radiating 10 kW.*

Sol. :

$$\text{Power radiated} = [I]^2 R_{radiation}$$

$$\therefore \quad 10 \text{ kW} = [12]^2 R_{radiation}$$

$$\text{Radiation resistance,} \quad R_{rad} = 69.44\Omega$$

$R_{rad} = \frac{10 \times 10^3}{144}$

Ex. 8.17 : *Calculate the length of a half-wave dipole antenna cut for a frequency of 75 MHz.*

Sol. :

$$\text{Frequency} \times \text{Wavelength} = c = 3 \times 10^8$$

$$[75 \times 10^6][\lambda] = 3 \times 10^8$$

$$\therefore \quad \lambda = \frac{300 \times 10^8}{75 \times 10^6} = 4 \text{ m.} = 4\text{m}$$

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

$$\text{Length of half wave dipole} = \frac{\lambda}{2} = 2 \text{ m}$$

Calculate the radiation resistance of an antenna which radiates 1000 W, when drawing 5A.

Power radiated = $I^2 \, R_{radiation}$

$1000 = [5]^2 \cdot R_{rad}$

$R_{rad} = \frac{1000}{25} = 40\,\Omega$

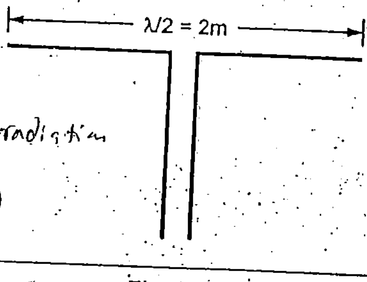


Fig. 8.44

navigation aids

(18)

Ex. 8.18 *Determine the optimum length of a Marconi antenna which is to be cut for a frequency of 65 MHz.*

Sol. :

$$\text{Frequency} \times \text{Wavelength} = c = 3 \times 10^8$$

$$\therefore \quad [65 \text{ MHz}] [\lambda] = 3 \times 10^8$$

$$\therefore \quad \lambda = \frac{300 \times 10^8}{65 \times 10^6} = 4.615 \text{ m}$$

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

$\lambda = \frac{3 \times 10^8}{65 \times 10^6} = 4.615$ metres

∴ length of marconi antenna $= \therefore \frac{\lambda}{4} = 1.1538$ m

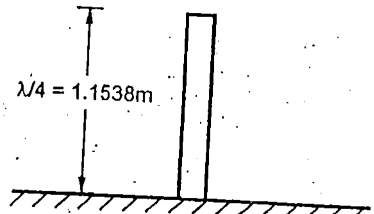


Fig. 8.45

Ex. 8.19 : *Calculate the required length of a half-wave dipole antenna cut for 90 MHz.*

Sol. :

$$\text{Frequency} \times \text{Wavelength} = c = 3 \times 10^8$$

$$\therefore \quad [90 \text{ MHz}] [\lambda] = 3 \times 10^8$$

$$\therefore \quad \lambda = \frac{300}{90} = 3.333 \text{ m}$$

$\lambda = \frac{3 \times 10^8}{90 \times 10^6} = 3.33$ m

$$\therefore \quad \frac{\lambda}{2} = 1.667 \text{ m}$$

Ex. 8.20 : *A Marconi antenna is 2.8 m in length. For what frequency can this antenna be used effectively ?*

Sol. :

$$\text{Length of Marconi antenna} = \frac{\lambda}{4}$$

$$\therefore \quad 2.8 = \frac{\lambda}{4}$$

$$\therefore \quad \text{Wavelength} = [2.8] [4] = 11.2 \text{ m}$$

$$\text{Frequency} \times \text{Wavelength} = c = 3 \times 10^8$$

$\lambda = \frac{c}{\lambda}$

$$\therefore \quad \text{Frequency} = \frac{3 \times 10^8}{11.2} \text{ Hz} = 26.7857 \text{ MHz} = \frac{c}{\lambda}$$

$= 0.267857 \times 10^8$ Hz

$= 26.7857 \times 10^6$ Hz

$= 26.7857$ MHz

(17)

Polarization : Polarization is defined as the direction of electric field (E- field) in electromagnetic wave radiated from the transmitting antenna.

(E)
6.3.1 Polarisation :

An electromagnetic wave transmitted from an antenna may be vertically or horizontally polarised. In the former case, the E vector is vertical and requires a vertical antenna to transmitt it. Alternatively, if the E vector is horizontal, the wave is horizontally polarised and requires a horizontal antenna to transmitt it. Figure (6.4) shows how

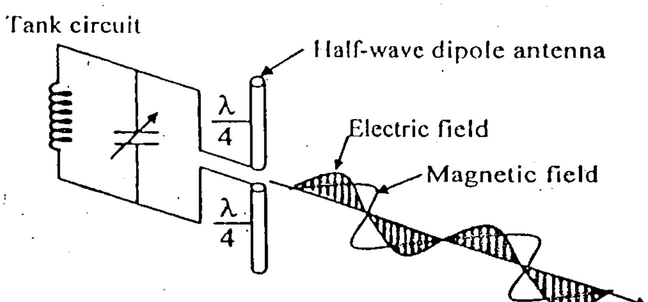


Fig. 6.4

a vertical half-wave dipole antenna transmitts vertically polarised electromagnetic wave. The wave polarisation is primarily a function of antenna orientation. However the plane of polarisation of an electromagnetic wave may actually be turned somewhat from its original direction during its travel through space. In order to receive maximum signal, the polarisation of the receiving antenna must be the same as that of the transmitting antenna which is the same as that of the transmitted wave. Consider figure (6.5 a). Here both transmitting

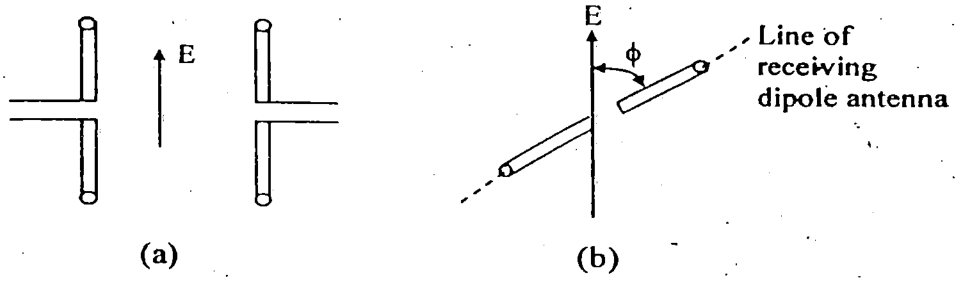


Figure 6.5

and receiving antennas are vertical and receiving antenna is parallel to the electric vector. This will give maximum reception. In figure (6.5 b) the line of receiving antenna makes an angle ϕ with the electric field vector. In this case only the component of electric field in the direction of line of receiving antenna E cos ϕ will induce signal in it. Hence less signal is received and polarisation loss factor is

$$\text{plf} = \cos\phi$$

Electromagnetic waves are usually vertically polarised, though other types of polarisation may be used for some applications. For example TV transmission for DD1 channel in Pune is horizontal. Sometimes

18

6.15 Antenna Polarization :

>>> [Asked in Exam : Dec. 2007 !!!]

Polarisation refers to the physical orientation of the radiated electromagnetic wave in free space. Polarisation of an electromagnetic wave can be defined by the direction in which the Electric Vector E is aligned during the wave propagation for atleast one full cycle.

An electromagnetic wave is shown in Fig. 6.53 which shows that the electric vector $\bar{E}$ and magnetic vector H are mutually perpendicular and they in turn are mutually perpendicular to the direction of wave propagation.

An electromagnetic wave is said to be linearly polarised if they all have the same alignment in space.

Define polarization in
state its type

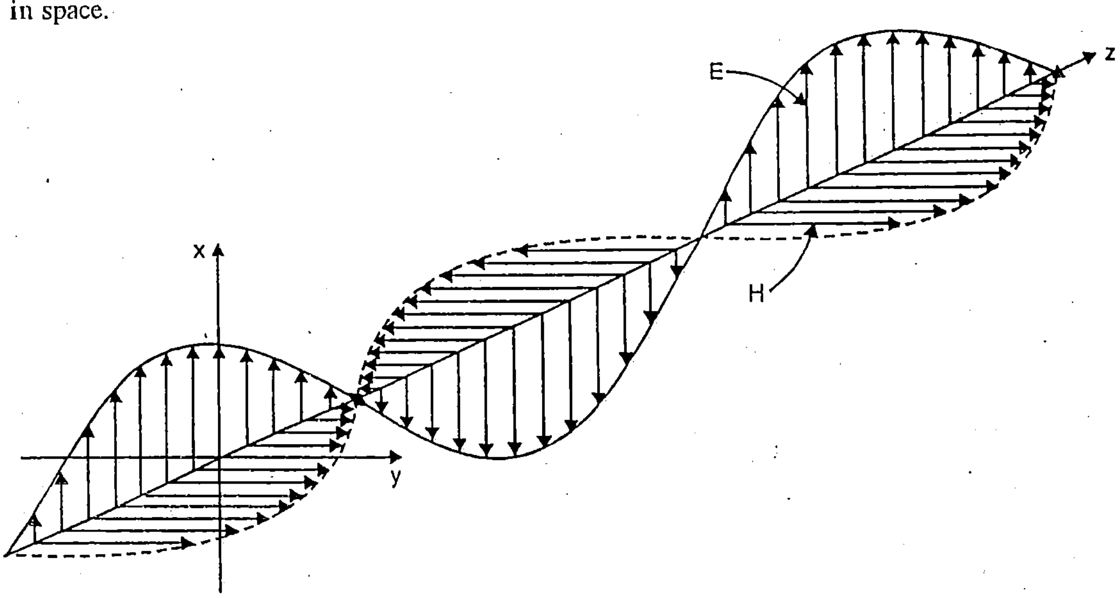


Fig. 6.53 : Electromagnetic wave in free space

In the Fig. 6.53 the wave is linearly polarised or vertically polarised since all the electric field vectors are vertical. The wave is said to be vertically polarised because the electric field vector E is lying in the vertical plane. Similarly, the wave is said to be horizontally polarised if the electric field vector E lies in the horizontal plane.

It is observed that the direction and polarisation of an antenna is alike because, if an antenna is vertical, it will radiate vertically polarised waves and a horizontal antenna will radiate horizontally polarised waves. The initial polarisation of electromagnetic waves is determined by the orientation of antenna itself in the space.

At LF, VLF, MF and HF where ground wave propagation is used vertical polarisation is preferred for transmission and reception because the vertical electric field vector do not get short circuited with ground. For television broadcasting in the VHF range horizontal polarisation is preferred to avoid man-made noise which has vertical polarisation.

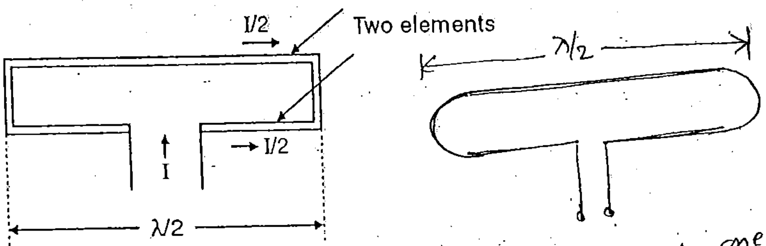
comment

Besides linear polarisation, antenna may also radiate circularly or elliptically polarised waves. Circularly polarised waves are produced when two linearly polarised waves are produced simultaneously in the same direction from the same antenna but with a phase difference of 90° as shown in Fig. 6.54.

At LF, VLF, MF and HF vertical polarisation is preferred for transmission ~~and reception~~ comment

What is folded dipole antenna?
State advantages of folded dipole antenna.

(19)



Folded dipole Antenna: Fig. 10.7.1 Folded dipole antenna

A folded dipole antenna is made from two half wave antennas joined at the ends, one of them open at centre where transmission line is connected.

The radiation pattern of the folded dipole is same as that of the straight dipole i.e. figure of eight (8).

But the input impedance of the folded dipole is four times higher than that of the straight dipole.

Typically the input impedance of $\lambda / 2$ folded dipole of Fig. 10.7.1. is 288 Ω.

The bandwidth of folded dipole, is higher than that of the straight dipole.

0.7.1 Advantages of folded dipole :

- Higher input resistance.
- Higher bandwidth.
- Ease of construction.
- Cost efficient.
- Impedance matching.

Note that the folded dipole antenna can be used as a transmitting as well as a receiving antenna.

Yagi Antenna:

It is generally used in TV receivers. It consists of a folded dipole with one reflector and two or more directors. The reflector is placed behind the folded dipole and is larger than folded dipole. The back lobe of Radiation Pattern is reduced by the reflector and thus improve the unidirectional radiation pattern.

The director elements are placed in front of dipole to increase radiation in forward direction (to get narrow beam of Radiation pattern). Thus using large number of such elements, a very narrow beam of Radiation can be obtained. In other words, such elements are used to improve the directivity in forward direction, therefore they are called as Directors.

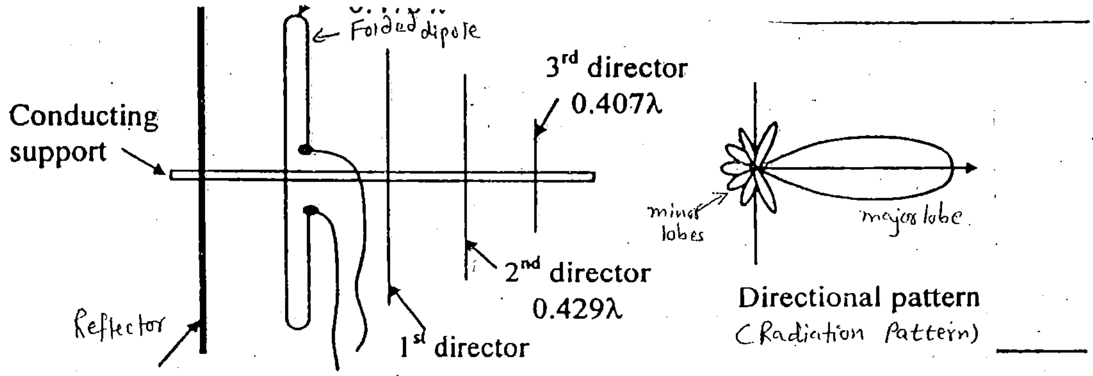
Write the role of Reflector and directors in Yagi antenna.

- When used as a transmitting antenna, the source is connected to the dipole called driven element. The parasitic elements (reflectors and directors) are not connected electrically anywhere.

(20)

10.8.1 Radiation Pattern :

- The radiation pattern of the Yagi antenna is shown in Fig. 10.8.2. which shows that the Yagi antenna is a directional antenna.



- The radiation pattern of the dipole is a figure of eight. But it is modified as shown in the Fig. 10.8.2 by the reflectors and directors.
- By adjusting the distance between the adjacent directors it is possible to reduce the back lobe of the radiation pattern. This will improve the front-to-back ratio of the yagi antenna.
- Generally only one reflector and many directors are used. The lengths of directors will reduce progressively.

10.8.2 Advantages of Yagi Antenna :

Some of the important advantages of Yagi antenna are as follows :

1. It is a directional antenna.
2. It has a moderate gain of about 7dB.
3. It is very compact.
4. Large bandwidth.
5. Can be used at high frequencies.
6. Adjustable front-to-back ratio.

Explain construction of yagi antenna. list its advantages and disadvantages.

10.8.3 Disadvantages :

- The gain is not very high.
- Needs a large number of elements to be used.

10.8.4 Applications :

- Yagi antenna is used as HF transmitting antenna.
- It is also used at higher frequencies at VHF as TV receiving antenna.
- A stack of Yagi antennas can be used as a super gain antenna.

10.8.5 Equivalent circuit of Yagi-antenna :

- The optical equivalent of a Yagi-antenna, is shown in Fig. 10.8.3.
- The reflector acts as a mirror, dipole acts as a source and the directors act as a lens.

Propagation of waves

In Radio communication system, there are no direct connections. The system is wireless. The Radio frequency (RF) signal generated by transmitter and is sent into the antenna of transmitter, which converts the RF electrical signal into electromagnetic signal, that can propagate or travel over a long distance. The antenna of Receiver pick up the electromagnetic signal and converts it into electrical signal.

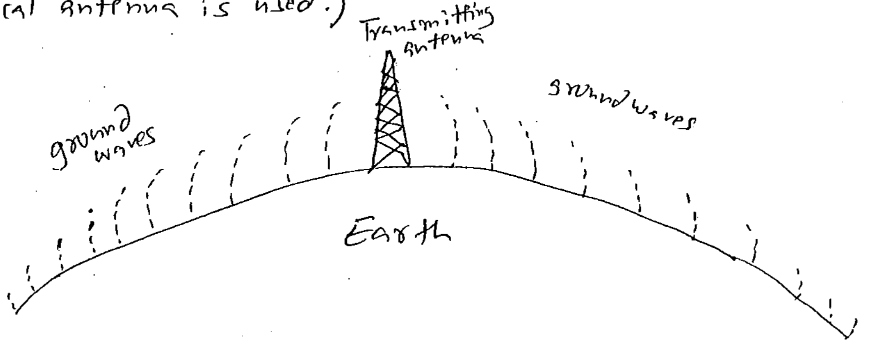
The Propagation of Radio waves from the transmitter to the Receiver (in space) can be of three types (three basic paths for Radio wave Propagation).

(i) Along the surface of the earth (Ground wave Propagation)
(ii) Space wave Propagation - From transmitter to receiver in a straight line.
(iii). Sky wave Propagation. - Upto the layer called ionosphere and back.

Ground wave Propagation (surface wave propagation):-

The electromagnetic waves propagate (travels) along the surface of (or closed to the) earth. The ground waves follows the curvature of earth and therefore travel at a distance beyon the horizon.

The ground waves must be vertically polarized (i.e. electric field component vertical), to prevent short circuiting of electric field component (Since the earth would short out the electric field, if ground waves horizontally polarised i.e in horizontal polarization, ground losses are more. Thus vertical antenna is used.)



②

Describe Ground wave propagation?

The ground wave propagation is the strongest at low and medium frequency range i.e. for radio signals in the range of 30 KHz to 3 mHz.
At higher frequencies beyond 3 mHz, the earth begins to attenate the radio signals (Attenuation increases rapidly with increasing frequency.)

Attenuation of the Ground waves:

The ground waves get attenuated due to the following reasons:-

(i) While passing over the surface of the earth, the ground waves induces some current into it. They thus lose some energy due to ~~propagation~~ absorption.

(ii) Due to diffraction the wave fronts will gradually tilt over as shown in the following figure. The angle of tilt (θ) goes on increasing as the ground waves propagates (Progress) over the surface of the earth. The increasing tilt causes greater short circuiting of the electric field component of the wave and hence reduction of field strength.
Eventually at some distance, the waves "lies down and dies".

(iii) The tilt angle (θ) increases with increasing in frequency, hence puts a limitation on the range of transmission.
It shows that the maximum range of such a transmitter depends on its frequency as well as its power. Thus in the VLF band, the range of transmission can be improved by increasing ~~th~~ the transmitting power. But solution will not work near the top of of the MF range, since propagation is limited by tilt. This is the reason, why the ground wave propagation cannot be practically used above 3 mHz.

Explain attenuation mechanism of ground waves for frequencies greater than 3 mHz.

at the end of process

Why ground wave progation is not used for MF and large frequency RF signals

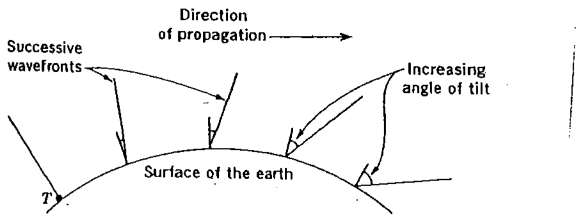


FIGURE 8-12 Ground-wave propagation.

Field strength at a distance due to Ground waves:

The radiation from antenna by means of ground waves, gives rise to a field strength at ~~a distance~~ a distance d from the antenna. The field strength in volts/meter is given by

write the expression for field strength of transmissing antenna and signal received by receiving antenna for ground wave propagation.

$$E = \frac{120\pi h_t \cdot I}{\lambda d}$$

E = field strength in volt/meter.

$120\pi = Z_0$ = characteristic impedance of free space

d = Distance from the transmitting antenna.

h_t = effective height of transmitting antenna.

I = antenna current.

λ = wave length

Now if a receiving antenna is placed at this point, the signal received by the antenna ~~is~~ in volts is given by

$$V = \frac{120\pi h_t h_r I}{\lambda d}$$

Where h_r = Effective height of receiving antenna.

The effective heights of the antennas are not same as their actual heights.

VLF propagation:-

When propagation is over a good conductor like seawater, and if frequencies below 100 KHz, The sea water is a better conductor than earth surface. Therefore the power loss due to surface ~~ab~~sorption in the surface is very small. The attenuation due to the atmosphere is also low. (hence ground waves can be used for a long distance communication below 100 KHz).

Thus the range of propagation is dependent mainly on the angle of tilt. The degree of tilt depends on the distance from the antenna in wavelength.

Therefore the ground wave can travel longer distances at very low frequencies (VLF), because of large wavelengths of ~~LF~~ VLF signals. The ~~range~~ waves in VLF range are able to travel long distances, around the globe, if sufficient power is transmitted.

The strength of low frequency signals changes very gradually, so the rapid fading does not occur.

slowly.

Thus the transmission at very low frequencies (VLF) is a very reliable means of communication over long distances, by ground wave propagation.

Applications of Ground wave propagation:

(i) In the AM radio broadcasting, operating in MW band.

(ii) The most frequent users of long distance ~~LF~~ VLF transmissions are ship communications such as radio navigation and marine mobile communications.

(iii) The VLF transmission is also used for time and frequency transmission. The transmission is continuously hourly transmission of stable radio frequencies, standard time intervals, time announcements, standard audio frequencies etc.

Write the Applications of ground wave Propagation

Write advantages/disadvantages of ground wave propagation.

Disadvantages of Ground wave Propagation:-

(1) limited range for higher operating frequencies.

(2) At low operating frequencies, very tall antennas should be used. [because antenna height should be atleast $\lambda/4$]

(3) High transmission power is necessary to cover the sufficient range. The power in excess of 1 MW is a common thing.

Advantages of Ground wave Propagation:

(1) Atmospheric conditions do not affect ground wave propagation too much.

(2) If transmitted power is large enough, then ground wave propagation can be used to communicate between any two points of world (i.e. it can be used for long distance communication)

Sky wave Propagation - The Ionosphere :-

The sky wave propagation is useful for long distance communication. In this case, the transmitted signal travels towards the ionised region in the upper atmosphere called ionosphere. The bending and refraction of the transmitted signal takes place due to ionosphere and come back towards earth.

The ionosphere and its layers.

The ionosphere is the upper portion of atmosphere, which absorbs large amount of radiation from the sun, which heats the upper portion of atmosphere and produces ionization in the form of free electrons, positive ions and negative ions. This layer of ions is known as ionosphere.

Along with the ultraviolet radiation, the α, β and γ radiation from the sun and the cosmic rays are also responsible for the ionization and hence formation of ionosphere.

The ionosphere is generally divided into four main layers as D, E, F1 and F2. The last two layers combine at night to form one single layer.

The D layer is the lowest layer of the ionosphere, extends from 50 km upto 90 km above the surface of earth. The ionization in this layer is least, because it is farthest from the sun. This layer has ability to refract ~~signals~~ VLF and LF waves. and ~~absorbs~~ high frequency signals passes through it.

The D-layer only present in day time and disappears during night time.

Write short note on ionosphere

The E layer exists between 90 km and 145 km above the earth. This layer also disappears during night, because of recombination of ions into molecules.

This layer can refract MF waves and some HF waves in day time.

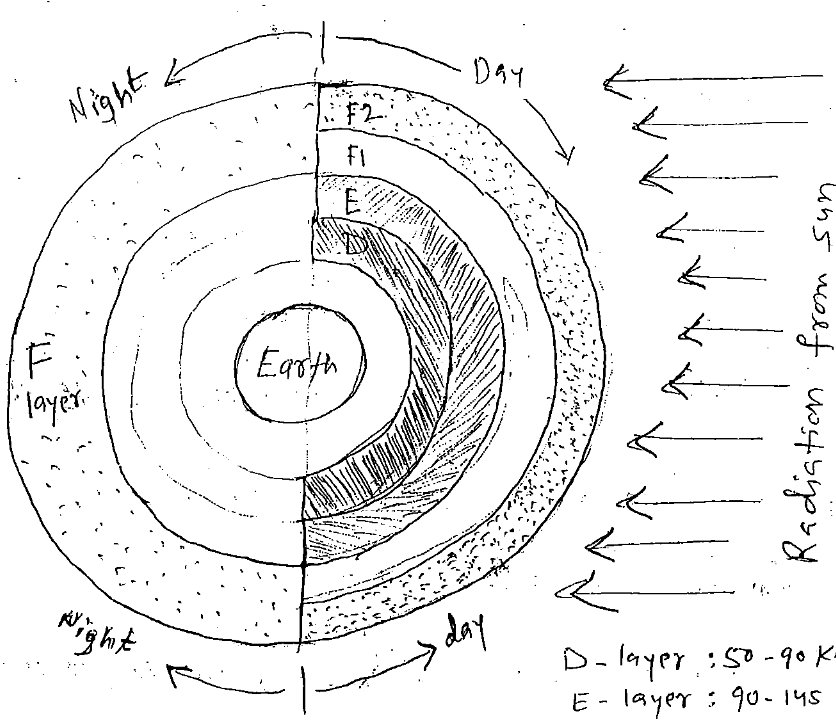
The F-layer exist from 145 Km to 400 Km above the earth. During day time this layer splits into two layers F1-layer and F2-layer. At night these two layers combine to form one single F-layer.

The F-layers are responsible for high frequency, long distance transmission due to refraction, & up for frequencies upto 30 MHz

explain How refraction of radio signals take place by ionosphere.

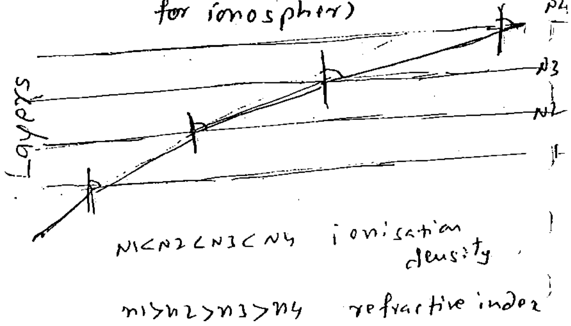
How does the reflection takes place? :- Actually reflection of radio signal does not take place. The refraction of signal takes from the ionosphere layers.

The ability of ionosphere to return a radio wave to the earth is based on the law of refraction and depends on ion density of the layers and, frequency of radio wave and angle of transmission. (or angle of incidence for ionospher)



D - layer : 50 - 90 Km
E - layer : 90 - 145 Km
F1 - layer : 145 - 300 Km
F2 - layer : 400 Km

(a)



Bending of wave using ionospher.

(b)

The radio waves are refracted from ionosphere ionized layers. When the radio signal goes into ~~inn~~ ionosphere, the ionization density increases, and refractive index of layers decreases. This causes the incident radio waves to be bent slowly. As the radio wave bends further away from normal then it becomes parallel to the layer and then bends downwards, shown in the figure (b). Due to this mechanism, the incident wave is reflected back to earth. The angle of incident and angle of reflection are equal.

Virtual Height :-

- The concept of virtual height can be understood by looking at Fig. 11.7.1(c). The incident wave returns back to earth due to refraction.
- In this process it bends down gradually and not sharply. But it is interesting to see that the incident and reflected rays follow exactly the same paths as those if the signal would have been reflected from a surface located at greater height.
- This height is called as the **virtual height**. If the virtual height of a layer is known then it is possible to find the angle of incidence required to return the wave to the ground at a selected point.

In respect to sky wave propagation explain the following terms.
(i) Virtual height
(ii) Critical frequency
(iii) MUF
(iv) Skip distance

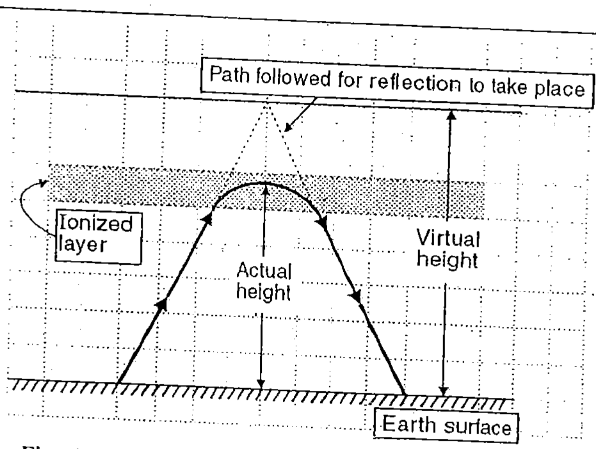


Fig. 11.7.1(c) : Virtual height of an ionized layer

Critical frequency (f_c) :- The critical frequency of a given ~~lay~~ ionospheric layer is defined as the maximum frequency that is returned back to the earth by that layer, when the wave is incident at angle 90° (normal) to it. The critical frequency for F2-layer is between 5 to 12 MHz.

The critical frequency (f_c) for a given layer is the highest frequency that will be returned down (back) to earth by that layer ~~after~~ when wave incident normal to the layer. The highest frequency that will be returned to the earth, when transmitted in a vertical direction is called as the critical frequency.

Maximum Usable Frequency (MUF):-

The Maximum Usable Frequency (MUF) is also called as limiting frequency & is defined for some specific angle of incidence θ other than the normal, as in case of critical frequency.

If angle of incidence is θ (between the incident ray and the normal), then MUF is given by

$$MUF = \frac{\text{Critical frequency}}{\cos\theta}$$

$$\boxed{MUF = f_c \cdot \sec\theta}$$

This equation is also known as the "secant law"

However the MUF is not defined in terms of angle θ in practice. Rather it is defined as MUF is the highest frequency that can be used for sky wave communication, between two ~~point~~ given points on the earth.

Normally the values of MUF are in the range of 8 to 35 MHz. The highest operating frequency between two points is selected to be less than MUF, but it is not very much less

Skip distance :

The skip distance is ~~the~~ defined as the shortest distance from transmitter, measured along the surface of the earth, at which a sky wave of fixed frequency (more than f_c) will be returned to earth back.

Following figure (a) shows the effect of variation in the angle of incidence θ, keeping the frequency constant.

(1) If the angle of incidence θ is quite large, for ray 1, the sky wave returns to ground at a long distance from the transmitter.

(2) As this angle is slowly reduced, then wave returns closer and closer to the transmitter, as shown by rays 2 and 3. But the penetration of the rays increases in the ionosphere.

(3) Now if angle of incidence is is made much smaller than that of ray 3,

If $f_c = 10\,MHz$. Angle of incidence is 45°

$$MUF = \frac{10\times10^6}{\cos 45} = \frac{10\times10^6}{0.7071} = 14.14\times10^6\,Hz = 14.14\,MHz$$

Explain the effect of ionosphere on the sky wave rays of varying the incident angles, and keeping freq. constant then rays 4 and 5 cannot returned back to the surface of earth and escape as shown in the figure

(5) Finally if the angle of incidence is only just smaller than that of ray 3, then wave may be returned, but at a distance farther than the return point of ray 3 (or this is a ray 6). It bends back very slowly, because ion-density is changing very slowly at this angle.

(6) Thus at a given frequency, the angle corresponding to ray 3 will result in the shorted distance upto the point of the return. Therefore this distance is the skip distance.

(7) Now if a higher frequency is beamed up at the angle 3, then it will not returned back to the earth. Thus MUF is the frequency, which makes a given distance to corresponding to the skip distance.

Multiple sky wave Propagation [Multiple Hop sky wave Propagation]:—

The sky wave propagation so far discussed till now is called as single hop transmission.

The transmission path is limited by skip distance and curvature of earth. The longest single hop distance is obtained when the ray is transmitted tangentially to the surface of the earth as shown in the following figure (b). The maximum range of the single hop transmission (maximum practical distance) is about 4000 km (due to skip distance and the curvature of earth).

The semicircumference of earth is about 20,000 km. The single hop transmission is not useful to cover this range, hence multiple hop transmission is generally preferred. This is shown in the following figure (b)

There is no problem in the multiple hop transmission in the north-south direction. But care is to be taken, when planning it for the east-west direction, if the transmitter and receiver happen to be either side of the terminator.

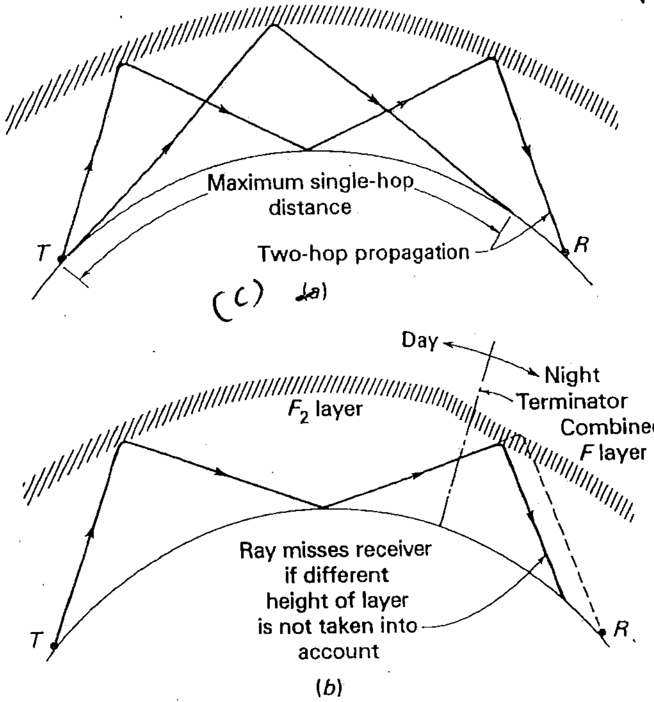
What is advantage of multiple hop skywave propagation ⇒ increases the distance between Transmitter & Receiver

What are sing hope and multiple hop sky wave propagation?

When a sky wave is transmitted, it is refracted by ionosphere & returns on the surface of earth, this is called single hop transmission.

When the ray is refracted by ionosphere coming back to the surface of earth where it is reflected back towards ionosphere and again it returns back to earth surface. This type of transmission is called multiple hop transmission.

As shown in the figure, there is "Day" on one side and "Night" on the other side of the terminator. The height of the combined F - layers at night is lower than that of the F_2 - layer in day time. Due to this the ray will return earlier than esepected and will miss the receiver as shown in the figure.



(b)

FIGURE 8-17 Long-distance sky-wave transmission paths. *(a)* North-south; *(b)* east-west.

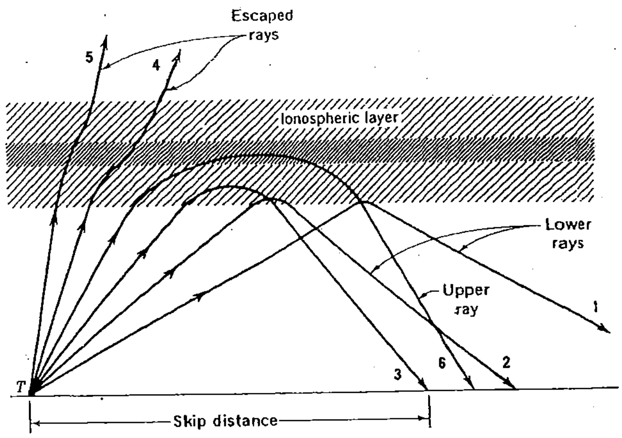


FIGURE 8-15 Effects of ionosphere on rays of varying incidence.

(a)

Concept of fading :

Fading means the variations or fluctuations in the signal strength at the receiver. The fading can be of following types

(i) Rapid or slow fading (ii) General or frequency selective fading

The fading of any type, takes place due to the interference between two waves, which follow different paths to travel from transmitter to receiver. Thus fading takes place due to multipath reception of the signal. Due to different path lengths, the two signals will undergo different phase shifts. At the receiver, the vector sum of them will take place. Therefore alternate cancellation and reinforcement will take place if the path difference is as large as $\lambda/2$. Such fluctuations are therefore more likely to occur at lower wavelengths or higher frequencies.

(11) What is fading? what are its causes?

Different reasons for fading:

(1) The fading can take place due to interference between the lower and upper rays of the sky wave.

(2) It can take place due to interference between waves arriving by different number of hops or paths.

(3) Due to interference between the ground waves and sky waves.

(4) Due to fluctuations of height or density in ionosphere layers.

As the fading is a frequency selective process, the signals very close to each other in the frequency domain will fade to a different extent.

The AM signal is very badly distorted due to such a frequency selective fading. The SSB signal is not affected to such an extent.

write applications of sky wave propagation

Applications:- The sky waves are useful for long distance communication, with a range of 3MHz to 30MHz frequencies. The short wave (SW) radio broadcast is possible due to sky wave propagation.

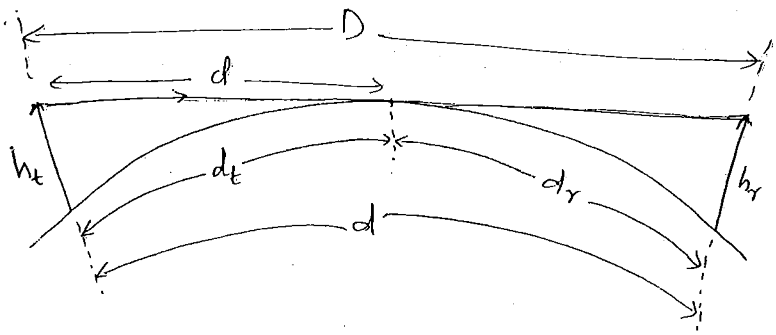
Space Wave Propagation (Line of sight Propagation)

The sky wave propagation cannot take place above the frequencies of 30MHz because the ionosphere cannot reflect back such high frequencies ~~and~~, whereas ground attenuates radio waves in a few hundred feet only. Hence the propagation of waves at these high frequencies are achieved by means of space wave propagation. Hence at radio frequencies above 30MHz, the space wave propagation is used.

The space wave propagation is the basic mode of propagation of higher radio frequencies like VHF, UHF and microwaves.

The space wave propagation take place by space waves or direct waves as shown in the following figure. These waves travel in a straight line directly from the transmitting antenna to the receiving antenna.

The direct or space waves are not refracted like sky waves nor they follow the curvature of earth like the ground waves.



Radio horizon :-

Radio horizon ~~is~~ for space wave is about four-third of the optical horizon (d). Let d be distance between transmitting antenna and horizon. $d = 4\sqrt{h_t}$. h_t = height of transmitting antenna above ground

lly $d = 4\sqrt{h_r}$ h_r = height of Receiving antenna above ground.

Thus for line of sight distance (D) ~~communication~~, as a function of ~~height~~ heights of transmitting and receiving antennas, and is given by

$$D = 4\sqrt{h_r} + 4\sqrt{h_t}$$

example: transmitting antenna height $h_t = 225m$ above ground level.
Receiving antenna height $h_r = 16m$ above ground level

then total distance between transmitting and receiving antennas is (i.e. Line of sight)

$$D = 4\times\sqrt{225} + 4\sqrt{16} = 76m.$$

Shadow zones.

What do you mean by shadow zone?

Write short note on Shadow zones

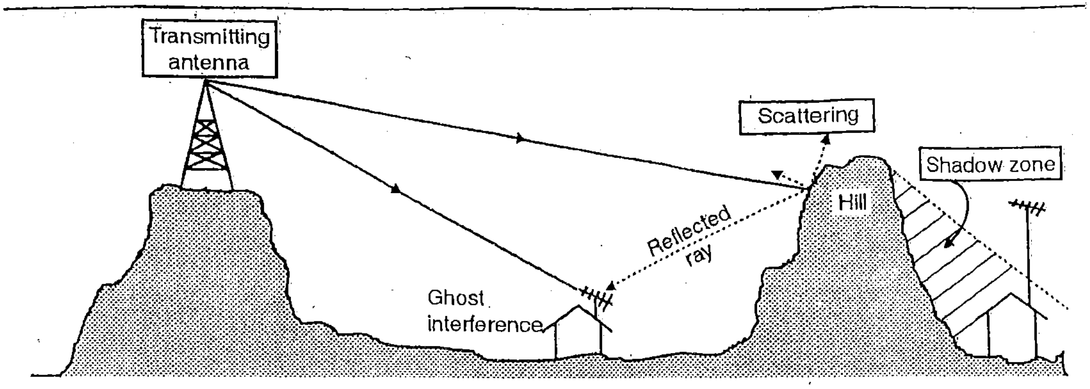
Since the space waves travels very close to the ground, any tall or massive objects such as hills, buildings, trees, etc will obstruct the space waves. These objects will absorb as well as scatter the energy. Therefore the shadow zones are formed.

The signal strength is very low in the shadow zones, therefore the receiving antennas need to be taller to receive adequate amount of signal.

On the other hand due to the reflections from the objects like buildings or trees, the "ghost" images are seen on the TV screens.

(13)

Write applications of space wave propagation (6m)



: Shadow zone in space wave propagation

6 Applications of Space Waves :

Applications of space wave propagation are :

- TV broadcasting
- FM radio broadcasting
- Microwave links
- Satellite communication

The ghost interference is caused by the difference in the path lengths (and hence in phase) between the direct and reflected rays.

Duct propagation: (microwave space-wave propagation) :-

What is duct propagation?
Under what circumstance it takes place?

Duct propagation is a special type of phenomenon which is also called as the "super refraction". It is observed at very high microwave frequencies. The duct propagation can be explained as follows :

(i) As the height above the earth increases, the air density decreases and the refractive index increases. The change in the refractive index is normally linear and gradual.

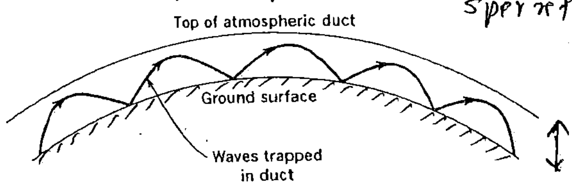
(ii) However under certain special atmospheric conditions, a layer of warm air may get trapped above the cooler air. This happens normally over the surface of water.

(iii) Due to this the refractive index will decrease more rapidly with height than usual. This happens near the ground normally within a distance of 30 meters above the surface.

(iv) Due to this rapid reduction of refractive index, the microwaves will completely bend back towards the earth surface, as shown in Fig. 11.9.1(a). This is similar to what happens in sky wave propagation. (multiple hop skywave propagation)

(v) Microwaves are thus continuously refracted inside the duct and reflected back by the conducting ground or water surface. These waves then propagates around the curvature of the earth over a distance of 1000 km or some times more.

The region in which the superrefraction takes place is called as duct. and superrefraction is known as ducting



atmospheric duct.

FIGURE 8-19 Superrefraction in atmospheric duct.

The main requirement for the formation of atmospheric ducts is so called temperature inversion. This is an increase of air temperature with height, instead of the decrease in temperature in standard atmosphere.

What is the main requirement for formation of atmospheric duct. (1m)

Tropospheric scatter Propagation:

- This type of propagation is also known as troposcatter or forward scatter propagation.
- It is a means to obtain beyond the horizon propagation of UHF signals.
- It uses the properties of the "troposphere" which is the nearest portion of the atmosphere. It is within about 15 km of the earth surface.

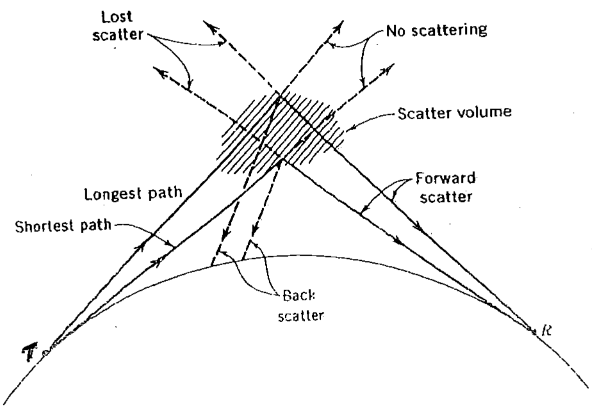


FIGURE 8-20 Tropospheric scatter propagation.

Draw the diagram for tropospheric scatter propagation,

Discuss tropo-scatter propagation

The tropospheric scatter propagation can be explained as follows :

(i) As shown in Fig. 11.10.1, two directional antennas are placed at points T and R so that their beams will intersect each other midway, above the horizon.

(ii) The transmitting UHF antenna at T beams up the energy. The energy will be scattered by the troposphere in different directions as shown in Fig. 11.10.1.

(iii) Sufficient radio energy is guided towards the receiving UHF antenna R. This happens by means of the forward scatter shown in Fig. 11.10.1. The receiving antenna will receive this radiation. Thus tropospheric scatter propagation can become a useful communication system.

(iv) The reason for the scattering is not fully understood. There are two theories suggested. One of them suggests that scattering takes place due to the reflections from the "blobs" in the atmosphere. This is identical to scattering of a search light beam by the dust particles. The other theory suggests that scattering is due to the reflections from the atmospheric layers.

(v) This phenomenon is a permanent and not a sporadic one. The frequencies most commonly used are 900 MHz, 2 GHz and 5 GHz.

(vi) The energy contents of the forward scatter which is received by the receiver is a very small percentage of the incident power. It is as small as may be one millionth of the incident power. Hence a very high transmitting power is needed.

11.10.1 Advantages and Applications :

- Advantage of troposcatter propagation is that it is not affected by the abnormal phenomena that affect the sky wave propagation. Hence it is a very reliable type of communication system.
- It is used to provide the long distance telephone and other communication links, in the unaccessible areas. It is used as an alternative to the microwave links or coaxial cables.

11.10.2 Fading in Troposcatter Communication :

- This type of propagation is subject to two forms of fading, Fast and Slow.
- The fast fading occurs many times per minute and the associated signal variation is more than 20 dB. It is known as **Rayleigh fading** and it is caused due to multipath propagation.
- The second type of fading is very slow. It occurs due to the variations in atmospheric conditions along the path. The space diversity systems are therefore employed to reduce the effects of fading.

State types of fading in troposcatter communication and explain them

Why tropospheric scatter propagation is a very reliable type of communication system. (2m)

11.11 Comparison of Ground, Sky and Space Wave Propagation :

	Sr. No.	Ground Wave Propagation	Sky Wave Propagation	Space Wave Propagation
4	1	It exists in the frequency range of 30 kHz to 3 MHz.	Exists in the range of 3 MHz to 30 MHz.	Used for frequencies above 30 MHz.
8	2	Used for radio broadcasting. (MW range).	Used for radio broadcast. (SW range).	Used for TV and FM broadcasting.
2	3	Ground waves are vertically polarized.	Vertically polarized.	Horizontally polarized.
Range of communication 9	4	Ground waves tilt progressively and eventually die. This limits the range of communication.	The transmission path is limited by the skip distance and curvature of earth.	The transmission path is limited by the line of sight and radio horizon.
1	5	Ground waves are surface waves which travel along the surface of the earth.	Sky waves are reflected from the ionosphere. This is how communication takes place.	Space waves travel in a straight line from transmitter to receiver through space.
3	6	The service range is a few hundred km.	Service range can be few thousand km.	Service range is not more than 100 km.
5	7	Power loss takes place due to absorption by ground and due to tilting of waves.	Power loss due to absorption of energy by the layers of ionosphere.	Power loss due to the power absorption and scattering by the tall and massive objects.
6	8	Problem of fading is not very severe.	Problem of fading is severe. Diversity reception is used.	Fading is not severe but shadow zones due to tall objects and ghost interference are serious problems.

calculate the critical frequency of E layer of ionosphere, if MUF is 3 MHz and angle of incidence is 20°

$MUF = \frac{f_c}{\cos\theta}$ $\therefore f_c = MUF \times \cos\theta$

$= 3\times10^6 \times \cos 20$

$= 3\times10^6 \times 0.9396 = 2.8188\times10^6$ Hz

$f_c = 2.8188$ MHz

Calculate the maximum radio horizon for transmitting antenna of height 200 m and receiving antenna of 2 meter above the ground surface

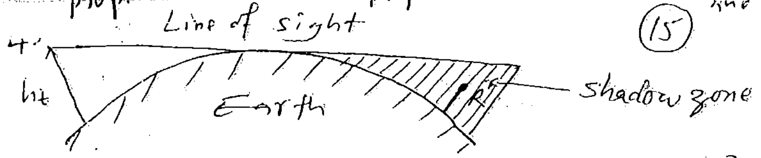
The maximum Radio horizon $= D = 4\sqrt{h_t} + 4\sqrt{h_r}$

$= 4\sqrt{200} + 4\sqrt{2}$

$= 4\times14.14 + 4\times1.44$

$= 56.56\cancel{m} + 5.656 = 62.224$ m

Compare sky waves and ground waves prop.   Compare ground wave propagation and space wave propagation (15)



The shaded area is known as the shadow zone and it is region where the field energy is not received.

Comparison of Ground wave, sky wave and space wave propagation:-

Parameter	Ground wave propagation	Sky wave propagation	Space wave propagation
(1) Wave propagation	Along the surface of earth	Refracted from ionosphere	In straight line (Line of sight)
(2) Polarisation of waves	Vertical	Vertical	Horizontal
(3) Range of service	Hundreds of Kms	Thousands of Kms	Limited to Radio horizons (few tens of Kms.
(4) Frequency used	30KHz - 3000KHz 3MHz Medium wave (mw) band	3MHz - 30MHz Shortwave (sw) band	30MHz onwards VHF/UHF Band
(5) Reason for attenuation	Ground surface behaves as conductor	some layers of ionosphere absorb sky waves	Creation of shadow zones due to tall and massive objects.
(6) Fading problem	Not a problem	Very serious	Not a problem
(7) Reliability of Communication	Most reliable since Ground conditions are Constant.	Not reliable because ionospher condition changes with time there are daily, seasonal and yearly variations so it is less reliable	Not reliable due to Ghost images and shadow zone.
(8) Applications	mw radio broadcast, Ameature radio, Marine and navigation aids	SW radio broadcast	Fm Radio, T.V. Radar

Compare Ground wave, sky wave and space wave propagations with respect following parameters.

Directional High frequency Antennas:-

HF antennas are differ from lower frequency Antennas due to two reasons: HF antennas are suitable for HF transmission/reception Since most of HF Communication is to be point to point, hence th. it require concentrated beam instead of omnidirectional (all directions) radiation. Such radiation pattern are obtained at HF, because of the shorter wavelengths.

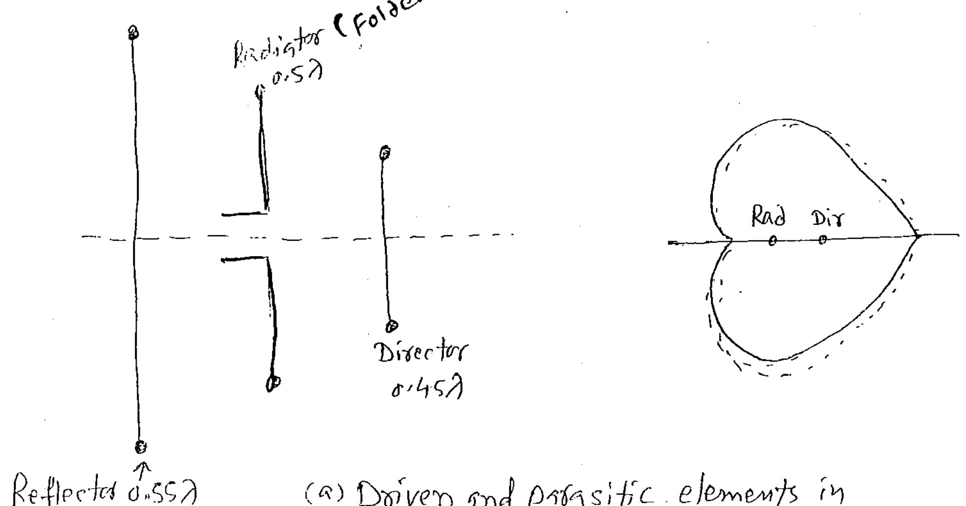
Dipole Arrays:

Explain dipole array.

An antenna array is a radiation system which consists of grouped radiators or elements as shown in the following figure.

These elements are placed close together, so that they will remain within each other induction field. Therefore they interact with one each other and produce a resulting radiation pattern, that is the vector sum of the indvidual ones.

Whether reinforcement or cancellation takes place in any given direction is determined not only by the individual characteristics of each element, but also by the spacing between elements. The gain of such structure is generally higher than 50.

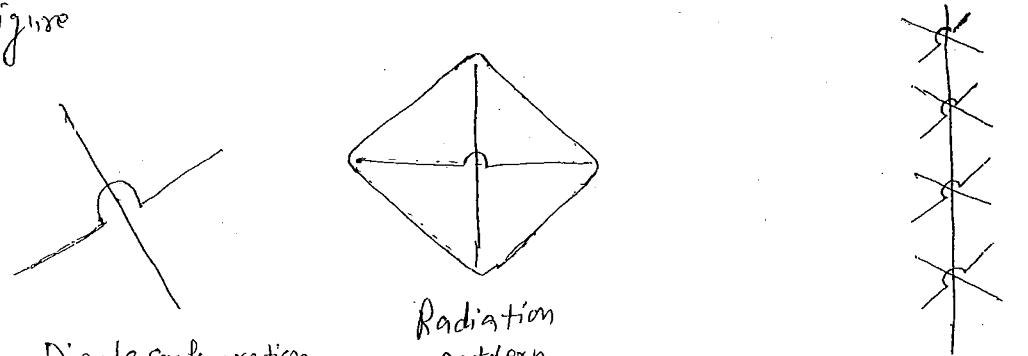


(a) Driven and parasitic elements in an array and radiation Pattern.

What is turnstile array antenna? Draw its Radiation pattern. 2

Draw the diagram for stacked array.

It is also possible to obtain an omnidirectional pattern in the horizontal plane by using the turnstile array as shown in the following figure



Dipole configuration Radiation pattern Four bay array

(b) horizontal dipole turnstile, radiation pattern and stacked array.

What do you mean by driven element and parasitic element?

Parasitic elements and driven element:

The radiating element (half wave dipole) is connected to the output of the transmitter, called as driven element. Where as the element, which is not connected to the transmitter output, is called parasitic element. Such parasitic element receives energy through the induction field created by a driven element, rather than by a direct connection to the transmission line.

A parasitic element longer than the driven element and close to it reduces signal strength in its own direction and increases it in the opposite direction. It is called as reflector. (behaves like concave mirror)

A parasitic element shorter than the driven element, it receives energy from driven element, tends to increase radiation in its own direction and therefore behaves like convergent convex lens, which is called as director.

State types of arrays:

Types of arrays: The large variety of types of array consists ~~as a~~ of dipoles arranged in specific physical patterns and excited in various ways as per requirement of condition.

Types of arrays are:

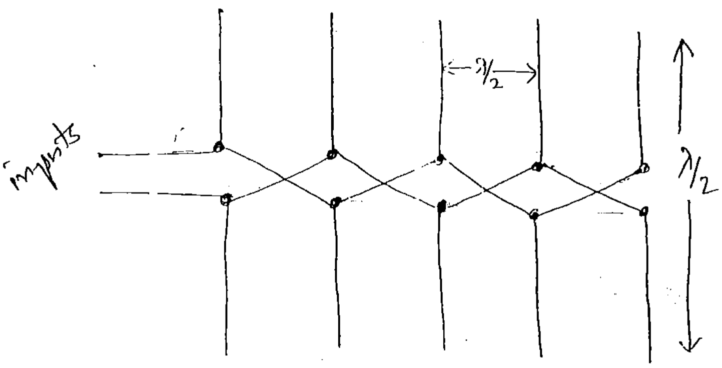
(i) Turnstile array (ii) Broad side array (iii) End fire array.

Broadside array:

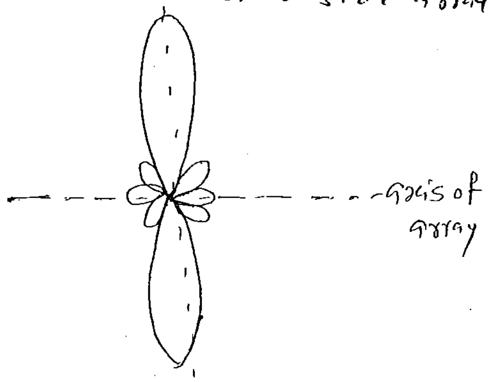
What is Broadside array?
Draw the diagram for Broadside array and explain it

It is the simplest array, consists of a number of dipoles of equal size, equally spaced along a straight line (i.e. collinear), with all dipoles fed in the same phase from the same source. Such an arrangement is called a broadside array, shown in the following figure (a) and fig. (b) shows radiation pattern.

Draw radiation pattern of Broad side array.



(a) Construction of Broadside array



(b) Radiation Pattern

The broadside array is strongly directional at right angles to the plane of the array, while radiating very little in the plane.

Typical antenna lengths in broadside array are from 2 to 10 wavelengths, typical spacing are $\lambda/2$ or λ. and dozens of elements may be used in the one array.

It is note that any array that is directional at right angles to the plane of array is said to have broadside action.

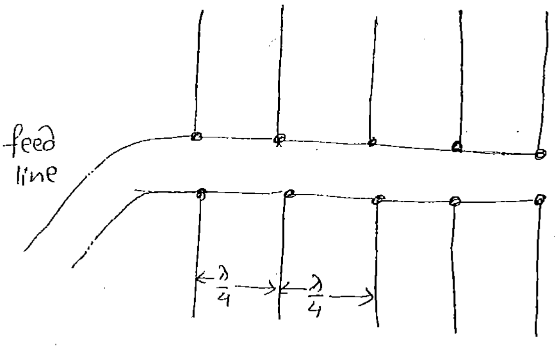
Explain Construction of Broadside array.

What is ~~End~~ fire array?
Draw the diagram of End fire array and explain its construction. Draw its Radiation Pattern (4)

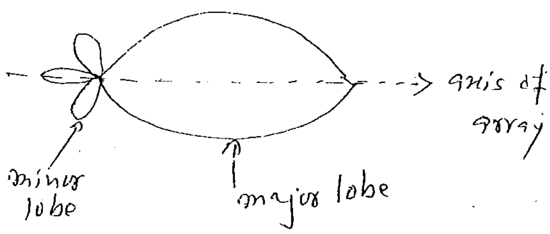
End-fire array :

The physical arrangement of the end-fire array is almost the same as that of the broadside array. Although the magnitude of the current in each element is still the same in every element, however there is a phase difference between these currents. This is progressive from left to right, in the following figure, as there is a phase lag between the succeeding elements. The radiation pattern of the end-fire array is shown in fig (b): which is quite different from that of the broadside array. It is in the plane of the array, not at right angles to it and it is unidirectional rather than bidirectional.

Give the difference between Broad side array and End fire array.



(a) Construction of End fire array



(b) Radiation Pattern.

The dipole spacing is λ/4 or 3λ/4 hence cancellation of radiation at right angles to the plane of the array but also in the direction from right to left. (Smaller lobes)

Both the end fire and broadside arrays are called linear, and both are resonant, because they consists of resonant elements.

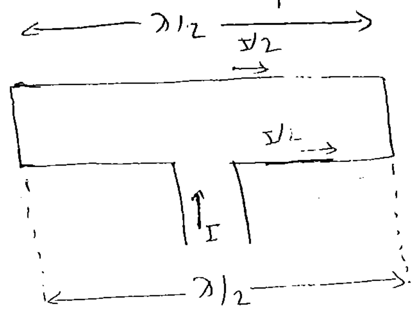
Write short note on End-fire array.

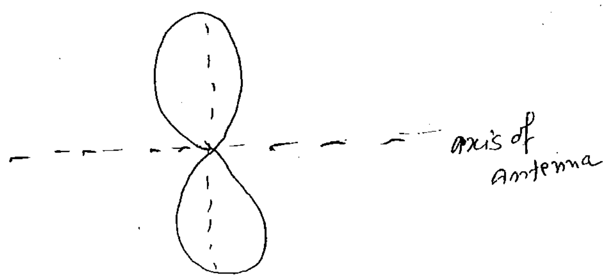
State advantages of folded dipole — 5

What is folded dipole. Explain folded dipole antenna & its radiation pattern.

Folded dipole and Applications

The folded dipole is a single antenna, but it consists of two halfwave dipole elements. The first element is feed directly, while the second is coupled inductively at the ends.





The radiation patter of folded dipole is the same that of straight dipoe i.e. figure of eight 8. But the input impedance of folded dipole is four times higher than that of straight single dipole. Thus input impedance of halfwave folded dipole is $4 \times 72 = 288\,\Omega$

{input impedance of single dipole antenna $= 72\,\Omega$}

Advantages of folded dipole:-

(i) Higher input resistance
(ii) Higher Bandwidth
(iii) Ease of construction
(iv) Cost efficient
(v) Impedance matching

Explain construction of Yagi-uda antenna.
Write its various applications

The Yagi-uda antenna (or Yagi antenna):-

This antenna is an array consisting of a driven element (dipole) and one or more parasitic elements.

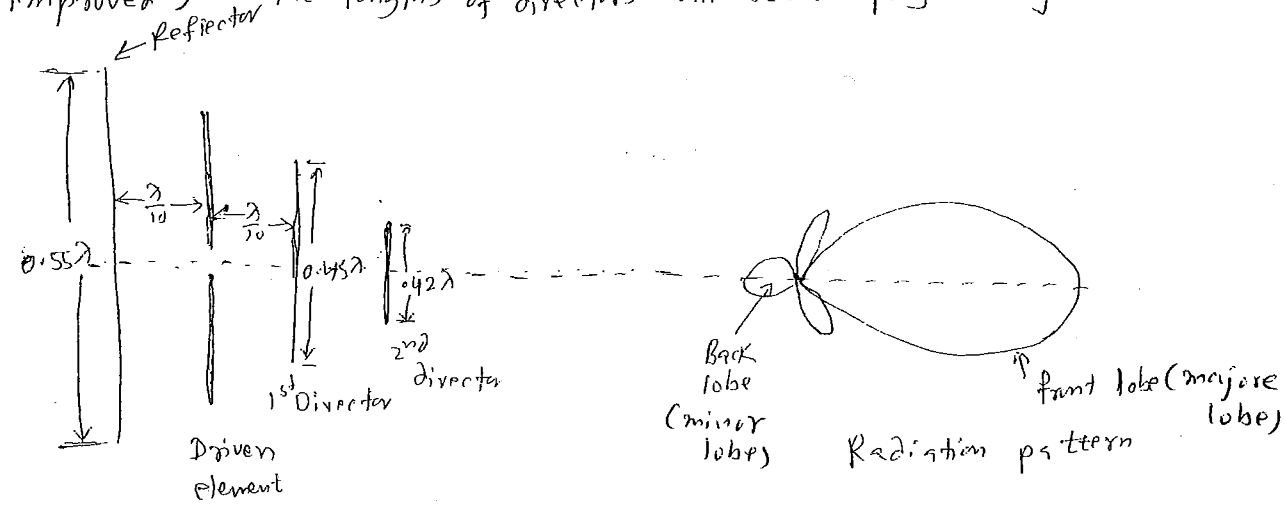
All these elements are arranged collinearly and close together as shown in following figure, along with optical equivalent and the radiation pattern

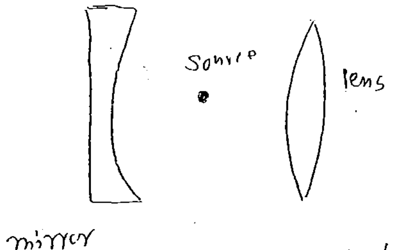
From the radiation pattern shows that the Yagi antenna is unidirectional antenna.

State advantages and disadvantages of Yagi antenna.

The radiation pattern of dipole is a figure of eight. But it is modified by the reflector and directors. The back lobe of Radiation Pattern is reduced by the reflector.

The directors elements are placed infront of dipole to increase radiation in forward direction (narrow beam of Radiation patter). Thus using large number of such elements, a very narrow beam of Radiation pattern can be obtained. In other words, such elements are used to improve the directivity in forward direction, therefore they are called as Directors. (i.e front to back ratio of antenna improved.) The lengths of directors will reduces progressively.





optical equivalent

Advantages:-

(1) It is a directional antenna.
(2) It has moderate gain
(3) It is very compact
(4) large bandwidth
(5) Can be used at high frequencies
(6) Adjustable front to back ratio

Disadvantages: (1) gain is not very high (2) Needs a large number of elements to be used

Applications: (1) Yagi antenna is used as HF transmitting antenna.

(2) It is also used at higher frequencies at VHF as TV receiving antenna

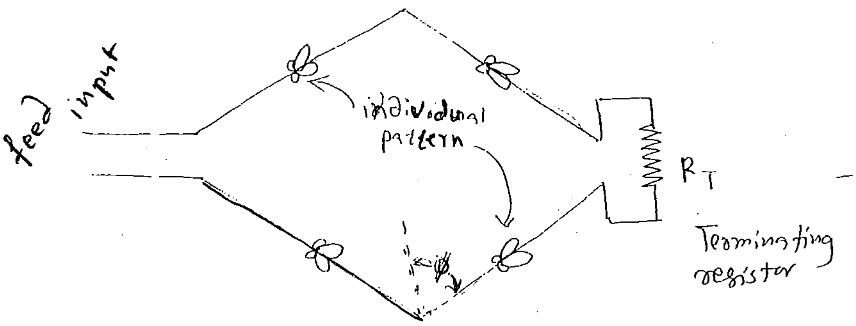
(3) A group of Yagi antennas can be used as a super gain antenna.

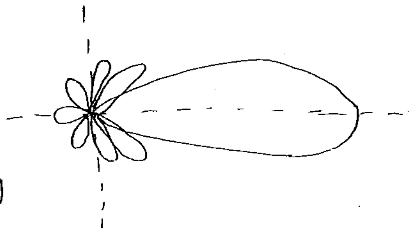
construction of Write anyone application of Rhombic antenna.
Explain the Rhombic antenna and draw its Radiation pattern

Nonresonant antennas: (Rhombic antenna):

A major requirement for HF is the need for a multiband antenna capable of operating satisfactorily over the frequency range of 3 to 30 MH, for either reception or transmission. Therefore array of nonresonant antennas ~~used~~ can be used, whose characteristics will not change very much over this frequency range.

A widely used antenna array, specially for point to point communications is shown in following figure. This is the rhombic antenna, which consists of nonresonant elements. It is a planar rhombus, which may be ~~con~~ thought as a piece of parallel wire transmission line bowed (bended) in the middle. The length of each ~~rew~~ radiators is equal and vary from 2 to 8 λ. and radiation angle ϕ varies from 40° to 45°





Radiation pattern in plane of antenna

The four legs or sides are considered as nonresonant antennas; The lengths of these sides and angle ϕ are interrelated and must be carefully chosen, so that the side lobes cancel properly, leaving only a single main lobe, lying along the main axis of Rhombic antenna.

The rhombic antenna is terminated by a resistor, called as terminating resistor. Its value is about 800 Ω

List various HF antennas?

What is a UHF antenna? (8)   write frequency range for microwave antenna.

Because the rhombic antenna is nonresonant, it does not have to be an integral number of half wavelengths long. Thus it is a broad band antenna.

The rhombic antenna is ideally suited for HF transmission and reception and it is a very popular antenna in commercial point-to-point communications

Why physical dimensions of microwave antenna is very small?
Write note on UHF and microwave antenna.

UHF and Microwave Antenna:

The frequency band from 0.3 to 3 GHz is called as UHF region. The antennas operating in this freq. band are called as the UHF antennas. The frequency band from 1 to 100 GHz is called as the microwave frequency region. The antennas operating in this frequency band are called as the microwave antennas. Both types of antenna are expected to be highly directional. The physical dimensions of both antennas should be many wavelengths, so as to have high gain.

At UHF and microwave frequencies, the wavelength is very small and so the physical dimensions of such antennas are very small.

A number of UHF and microwave applications, such as Radars, which are used in direction finding, hence directional antennas are useful. Several applications such as microwave communications links, which are point to point services & in which without interference between ~~sea~~ various services. The use of directional antennas are very useful for such applications in the areas where interference problems are serious (satellite TV reception).

At high frequencies, the performance of active devices goes down. The maximum output power decreases and device noise increases. It can be proved that all these problems can be sorted out, if we have high gain directional antennas.

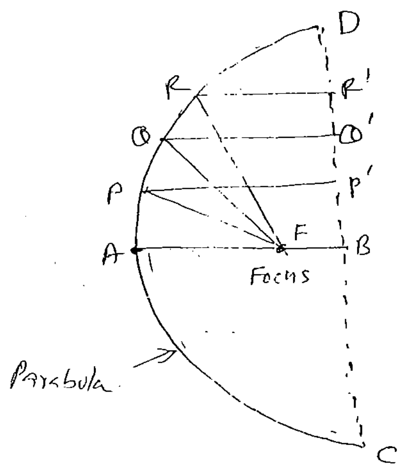
Antenna with Parabolic Reflectors: (dish antenna)

The parabola is a plane curve, defined as the locus of a point which moves so that its distance from another point (called the focus) plus its distance from a straight line (directrix) is constant. These special geometric properties of a parabolic reflector makes it very useful as a microwave or light reflector.

Geometry of the parabola:

Following figure shows a parabola CAD, whose focus is at F and whose axis is AB. From the defination of parabola that

$FP + PP' = FQ + QQ' = FR + RR' = K$



where K = constant and may be change if different shape of parabola is required

AF = focal length of parabola

The ratio of Focal length to the mouth diameter (AF/CD) is called aperture of the parabola.

⊗ Consider a source of radiation placed at the focus. All waves coming from the source and are reflected by the parabola equally from every point. It should be noted that all the reflected waves are in phase with each other. As a result they get added to give very strong and concentrated radiation along AB axis.

However cancellation of waves will take place in all other directions, because of path-length ~~differences~~ differences. This shows that the parabolic reflector leads to the production of concentrated beams of Radiation.

⊗ The dish antenna can be a transmitting antenna or receiving antenna. If it is a transmitting antenna

Explain geometry of parabolic reflector. 10

Write a short note on paraboloid.

A practical reflector uses the properties of the parabola, will be three dimensional bowl-shaped surface, which is obtained by revolving the parabola about the axis AB. The resulting three dimensional surface is called as <u>paraboloid</u> or a parabolic reflector or microwave dish.

We can use the dish antenna as a receiving antenna as well. Then this parabolic reflector is a high gain receiving directional antenna. (The concentration remains same but at focus used as receiver instead of transmitting source) The parabolic reflector will bring only those rays together which are coming in direction BA. These rays are brought together at focal point (because they are in phase). The rays arriving from any other direction are cancelled out (because of path differences between them)

<u>Principle of reciprocity</u> :-

This property state that the properties of an antenna remain the same and are independent of whether it is used as transmitting antenna or a receiving antenna.

The parabolic reflector increases the gain as well as the directivity of microwave antenna.

<u>properties of paraboloid reflectors</u> :

The directional pattern of an antenna using paraboloid reflector has a very sharp main lobe, surrounded by a number of minor lobes, which are much smaller, in the direction AB.

If the primary or feed antenna is nondirectional then paraboloid will produce a beam of radiation whose width is given by the formulas

$$\phi = \frac{70\lambda}{D} \quad \text{—(1)}$$

$$\phi_0 = 2\phi \quad \text{—(2)}$$

What is reciprocity of antenna ?

Write the properties of paraboloid reflector

Where λ = wavelength in meter
ϕ = beamwidth between half power points in degrees
ϕ_0 = beamwidth between nulls in degree
D = mouth diameter in meter.

Calculate the beamwidth between nulls of a 2m paraboloid reflector used at 6GHz.

$$\lambda = \frac{c}{f} = \frac{3 \times 10^8 \text{ m/sec}}{6 \times 10^9} = \frac{1}{2} \times 10^{-1} = 0.05 \text{ m}$$

$$\phi_0 = 2\phi$$

$$\phi_0 = 2 \times \frac{70\lambda}{D} = 140 \times \frac{0.05}{2}$$

$$= 3.5^0$$

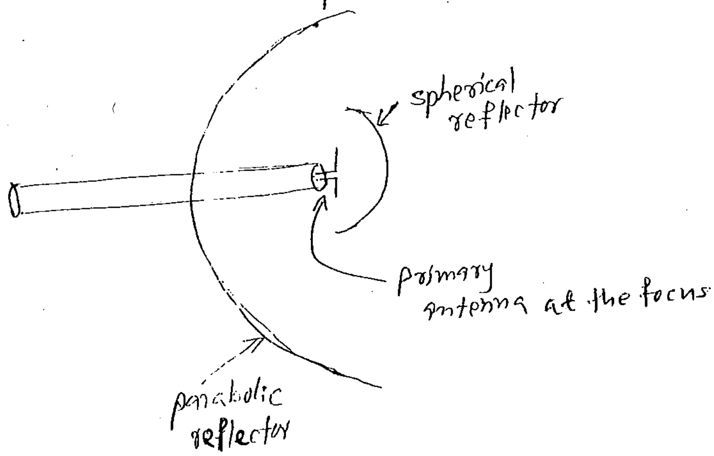
The gain of an antenna using paraboloid reflector is depends on aperture ratio (D/λ). If antenna is loss less, then power gain is given by $A_p = 6\left(\frac{D}{\lambda}\right)^2$

Calculate the power gain of the antenna in above example.

$$A_p = 6\left(\frac{200}{5}\right)^2 = 9600.$$

<u>feed mechanism</u>:-

(i) In paraboloid or dish antenna, the primary antenna is kept at the focus of the parabolic reflector and additional small spherical reflector is used as shown in the following figure for best transmission or ~~i.e.~~ reception.



Center-fed paraboloid reflector with spherical reflector (shell)

Explain different types of feed mechanisms (6m)

The direct radiation from the feed, which is not reflected by the paraboloid, tends to spread out in all directions and hence partially spoils the directivity. Several methods are used to prevent this. One of them is use of a small spherical reflector as shown in above figure. It is used for redirecting the all radiation back to the paraboloid, which is otherwise getting scattered in all directions.

Why small spherical reflector is used in paraboloid?

(2) Another method is to use a small dipole array at focus, such as Yagi-Uda or an end fire array, pointing at the paraboloid reflector.

(3) A horn antenna is pointing at the main reflector. It has directional pattern in the direction in which its mouth points. Therefore direct radiation from the feed antenna is avoided.

(4) Cassegrain feed:

Following figure shows another feeding arrangement called Cassegrain feed. It uses a hyperboloid secondary reflector. One of ~~the~~ its focal points of the hyperboloid coincides with the focus of the paraboloid

Explain Cassegrain feed mechanism

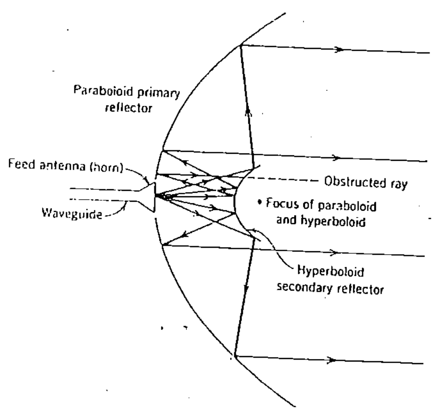


FIGURE 9-30 Geometry of the Cassegrain feed.

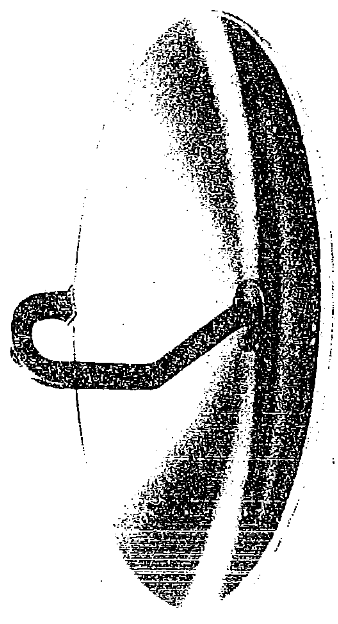


FIGURE 9-29 Paraboloid reflector with horn feed.

Because of this transmission action is shown in above figure
The rays emitted from the feed horn antenna are reflected from the paraboloid mirror (surface of hyperboloid secondary reflector). The effect on the main paraboloid reflector being the same as that of a feed antenna at the focus. Then main reflector gives the rays in parallel.

Other parabolic reflectors:

The full paraboloid (dish) is not the only practical reflector that uses the properties of parabola, but there are others also. Out of these three most commonly used parabolic reflectors are given in the following figure

list different types of parabolic reflector (iv)

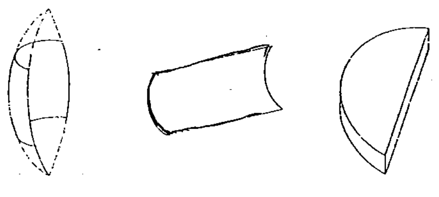

(a) (b) (c)

FIGURE 9-32 Parabolic reflectors. (a) Cut paraboloid; (b) paraboloid cylinder; (c) "pill-box."

State the drawback of pill box reflector

Each of them has an advantage over the full paraboloid antenna is that, they are much smaller, but beam is not as directional in one plane as compared to that of the full paraboloid.

With the pill box reflector, the beam is very narrow horizontally, but not directional vertically.

Another form of the cut paraboloid is shown in the following figure. This is called the offset paraboloid reflector. In which the focus is located outside the aperture. (in this case just below it) Now if an antenna feed is placed at the focus, the reflected and collimated rays will above it without any problem and thus removing any interference

Explain

This method is generally used, if the feed antenna is larger as compared with the reflector.

Explain offset parabolic reflector.

Draw the Radiation Pattern of paraboloid antenna.

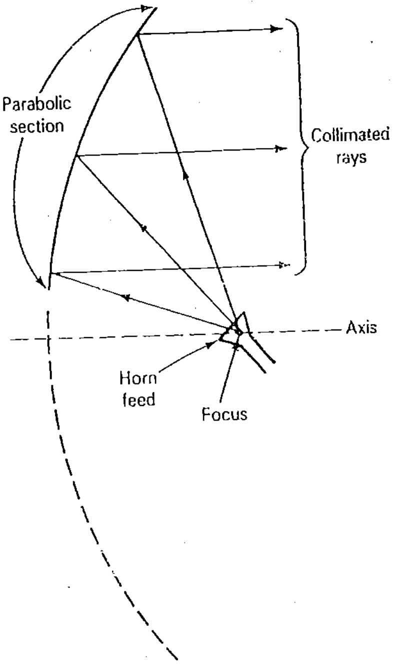
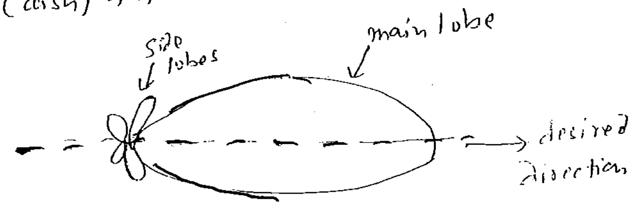


FIGURE 9-33 Offset paraboloid reflector.

The other common reflectors are hoghorn and Cass-horn reflectors

Radiation Pattern of paraboloid (dish) antenna:



It shows that radiation pattern include a narrow main lobe in the desired direction AB and small side lobes in all other directions. Radiation Pattern is ~~highly~~ shows that paraboloid or microwave dish antenna is highly directional.

State disadvantages of parabolic reflector antenna?

Disadvantages (shortcomings and difficulties) of paraboloid reflector antenna:

(1) Presence of side lobes.
(2) Presence of false echoes in radar, due to reflections from the direction of side lobes (specially when object is near)
(3) Increase in noise at the antenna terminals, caused by reception from undesirable sources. This is a serious problem in case of low noise receiving system e.g. radioastronomy.
(4) ~~Diffra~~ Side lobes created due to diffraction, at the edges of paraboloid, which producing interference.
(5) The finite size of the primary antenna also influences the beamwidth of antenna, using paraboloid reflectors.
(6) Broad main lobe.

Advantages:- (i) very high gain (ii) High directivity (iii) relatively narrow beam width.

List the advantages of parabolic reflector antenna.

Horn ~~Horn~~ Antennas:

A wave guide can radiate energy (~~like~~ acts like an antenna) into space if we excites it from one end and if its other end is left open. The waveguide radiates a large amount of energy as compared to that radiated by a two wire transmission line. But the problem with wave guides is that a very small amount of energy out of the total forward energy is radiated and large part of energy is reflected back by the open circuit.

Similar to transmission line, the open circuit at the far end actually acts as discontinuity. So wave guide is very poorly matched to the space. ~~A poor and non directional pattern~~

Diffraction around the edges of a waveguide gives a poor and non directional pattern

In order to overcome these difficulties, the mouth of the waveguide, to open out, same as in case of transmission line.

When a transmission line, is opened, it results into a dipole antenna, but when a wave guide is opened, it results in an electromagnetic horn, instead of the dipole.

Basic ~~horns~~ horns:

The wave guide is terminated into horn. The basic horns are available in different shapes as shown in the following figure

state and draw types of basic horns.

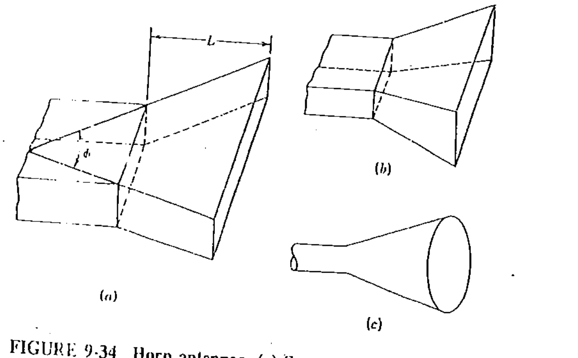


FIGURE 9-34 Horn antennas. (a) Sectoral; (b) pyramidal; (c) circular.

Compare waveguide and Transmission line

Compare: open out the mouth of wave guide, results into a dipole antenna

What is / are the advantage of horn

Due to the horns used, the abrupt discontinuity, that existed ~~is~~ ~~replaced~~ between a waveguide and free space, is gets converted into a gradual transformation (i.e Radiation in free space increases and reflections decreases). If the impedance matching is correct, then all the energy traveling in the forward direction in the waveguide will be radiated.

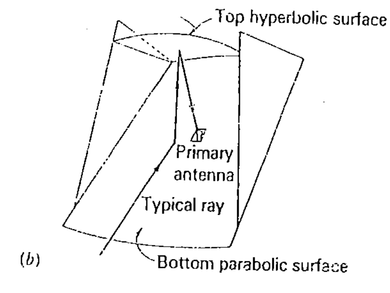
The directivity is will also be improved and diffraction is reduced.

The three most commonly used horn configurations are (1) Sectoral (2) Pyramidal (3) Circular. These are shown in following above figure.

The sectoral horn flare out in one direction only. and it is the equivalent of pillbox parabolic reflector. The pyramidal horn flares out in both directions. The conical circular horn is similar to it (Pyramidal horn) for a circular waveguide.

List two special horns.

Special horns: There are two antennas: Cass-horn and hoghorn antennas. Each consists of horn and parabolic reflector



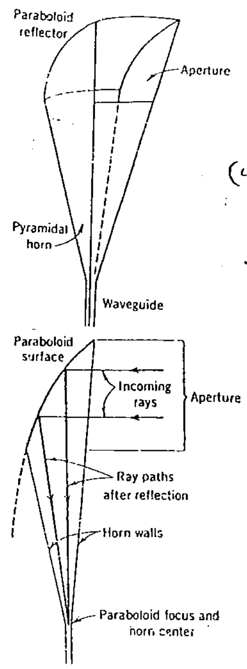
Cass-horn antenna. feeding the Cass-horn.

In case of the Cass-horn antenna, radio waves are collected by the large bottom surface (shown in the figure.), which is slightly curved (parabolically) and ~~are~~ wave are reflected upward at an angle of 45°. Upon hitting the top surface, which is a large hyperbolic cylinder, they are reflected downward to the focal point, situated in the centre of the bottom surface. They are collected by the conical horn placed at the fous.

This type of horn reflecter antenna has ~~more~~ gain and lo beamwidth comparable to those of a paraboloid reflecter of same diameter.

Explain with neat diagram Hoghorn antenna/Cass horn antenna.

The hoghorn antenna shown in the figure is another combination of paraboloid and horn. It is a low noise microwave antenna. It consists of a parabolic cylinder, joined to a pyramidal horn, with rays comes out from, or being received at the apex of the horn. An advantage of the hoghorn antenna is that the receiving point does not moves when the antenna is rotated about its axis.



(a) hoghorn antenna

(b) ray paths

Lens Antennas:

The antennas used optical principles for microwave antennas are : paraboloid reflector and Lens antenna.

Lens antenna is used as collimator at frequencies above 3 GHz.

Principle:- Following figure explained the operation of a dielectric lens antenna. From figure (a) we see that refraction takes place and rays at the edges are refracted more than those near the centre. A divergent beam is collimated and rays leaving the lense are parallel. It is assumed that the source is at the focal point of the lens. The reciprocity of antennae is also explained here. If a parallel beam is received, it will be converged for reception at the focal point.

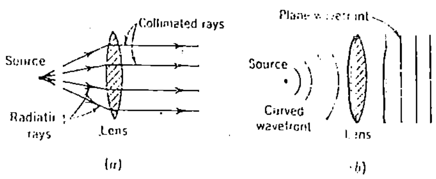


FIGURE 9-37 Operation of the lens antenna. (a) Optical explanation; (b) wavefront explanation.

It is note that curved wavefront is present on the source side of lens. We know that a plane wave front is required on the opposite side of the lens, to ensure correct phase relationship. As shown in fig (b) the lens does this.

Explain the working principle of Lens Antenna.

state types of antennas which uses optical principles.

state the types of special antennas. Explain any one of them.

Wide Band and special purpose Antennas :-

It is always desirable to have an antenna, capable of operating over a wide frequency range. This is because a number of widely spaced channels are used & as in ~~case~~ short wave transmission or reception or because only one wide channel is used, as in case of television transmission and reception.

In TV reception it is desirable to use the same receiving antenna for a group of neighboring channels.

Therefore it is required that for antennas whose radiation pattern and input impedance characteristics remain constant over a wide frequency range.

The horn (with or without paraboloid reflector), the rhombic and the ~~folded~~ folded dipole have the broad band properties for both impedance and radiation pattern.

The special purpose antennas ~~to be described~~ are discone, helical and log-periodic antennas and simpler loops used for direction finding.

List special purpose antenna.

List the broad Band antennas.

Folded Dipole (Bandwidth Compensation) :-

A simple compensating network for increasing the bandwidth of a dipole antenna is shown in figure ⓐ

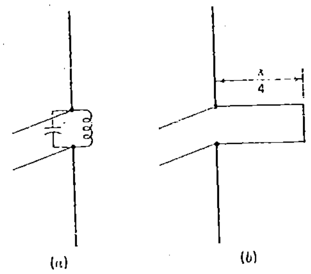


FIGURE 9-39 Impedance-bandwidth compensation for half-wave dipole. (a) LC circuit; (b) transmission line.

The LC circuit is parallel resonant at the half wave dipole resonant frequency. Therefore at this frequency, its impedance is a high resistance

Below this resonant frequency, the antenna reactance becomes capacitive, ~~while the reactance of LC circuit becomes inductive~~. Above the resonant frequency, the antenna reactance becomes inductive and LC tuned circuit becomes capacitive.

Over a small frequency range, near resonance, there is a tendency to compensate for the variations in antenna reactance and the total impedance remains resistive

This compensation ~~It~~ can be easily obtained with short circuited quarter wave transmission line as shown in fig(b)

The folded dipole also provides the same type of compensation.

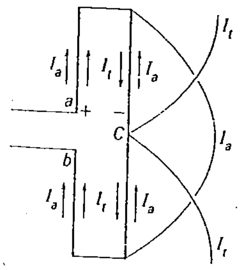


FIGURE 9-40 **Folded dipole showing antenna and line currents.**

Above figure shows folded dipole antenna, which may be considered as two short-circuited quarter wave transmission lines, connected together at C and feed in series. The transmission line currents are labelled I_t, whereas the antenna currents are labelled as I_a. When a voltage is applied between a and b, both sets of currents I_t and I_a flow. The antenna currents I_a only contribute to the radiation. The transmission line currents I_t flow in opposite directions and their radiations cancel.

It is note that the Yagi-Uda antenna is a broad band antenna, since ~~th~~ its driven element is a folded dipole.

Helical Antenna:

A helical antenna is a broadband VHF and UHF antenna, which is used, when circular polarization is required.

A helical Antenna consisting of a conducting wire wound in the form of a helix. In most cases, helical antennas are mounted over a ground plane. The feed line is connected between the bottom of helix and ground plane. It can be operated in two modes: normal mode and axial mode.

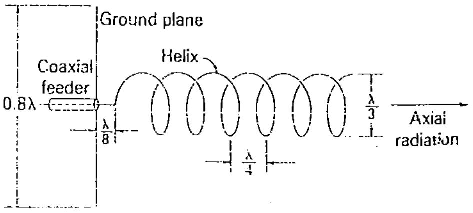
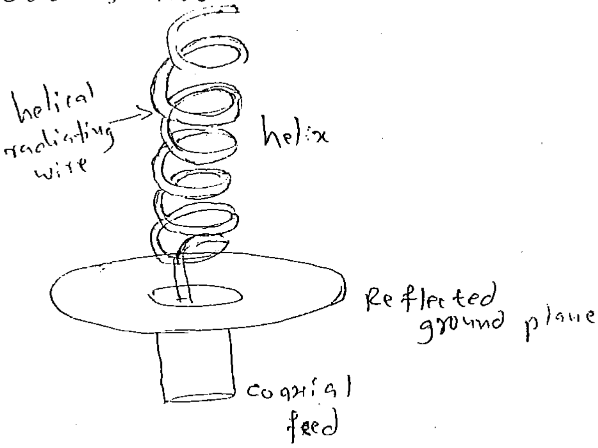


FIGURE 9-42 Dimensions of end-fire helical antenna.



In the first, radiation is in a direction at right angles to the axis of the helix. The second mode produces a broad band, directional radiation in the axial direction.

If the helix circumference becomes equal to wavelength, then wave travels around the turns of helix. and radiant lobe is circularly polarized.

The helical antenna is used as a single or in an array for transmission and reception of VHF signals through the ionosphere

The helical antenna is ~~generally~~ mostly used for satellite communication particularly for radiotelemetry.

Write the name of one antenna which used circular polarization.

Explain the construction of Helical Antenna

Write the two modes for operation of Helical Antenna

State atleast one application of Helical antenna

Explain construction of Discon antenna.

Write short note on Discon antenna.

Discone Antenna -

A discone antenna is a version of a biconical antenna, in which one of ~~of~~ the cones is replaced by a disc. It is usually mounted vertically, with disc at the top and cone at the bottom.

The Discone antenna is designed to radiate an omnidirectional (all directions) pattern in the horizontal plane, with vertical polarization. It is a broad band antenna.

The discone is a low gain antenna, but it is a omnidirectional. It is used as VHF and UHF receiving and transmitting antenna. It is used at airports, where communication must be maintained with aircraft that come from any direction

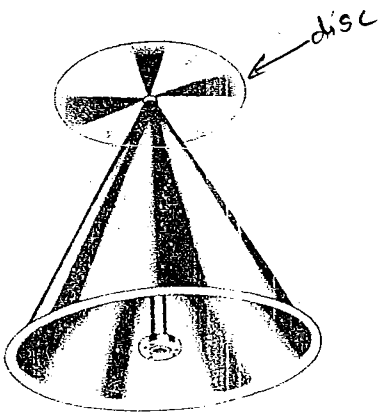


FIGURE 9-43 Discone antenna.

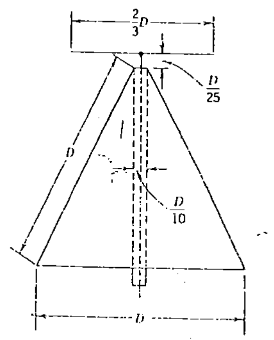


FIGURE 9-44 Dimensions of discone antenna.

Log - Periodic Antennas :

It is basically an array of dipole, fed with alternative phase, lined up along the axis of radiation. Following figure shows structure of Log-periodic antenna.

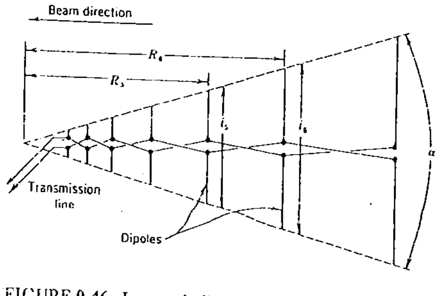


FIGURE 9-46 Log-periodic dipole array.

What is log-periodic antenna? 22

Draw the figure to indicate construction of log-periodic antenna

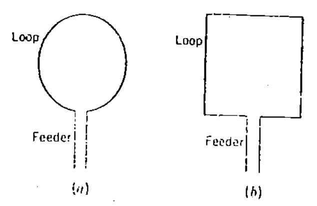
This antenna has unique feature that its impedance is a periodic function of the logarithm of the frequency, hence its name so. It is a multi element, directional antenna, designed to operate over a wide band of frequencies.

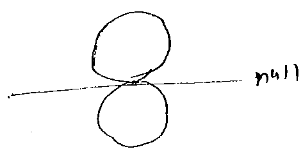
This type of antenna is used for mobile base station operations, where many channels must be handled over a single antenna system with good directive characteristics.

Loop antennas:

It is a single turn coil carrying RF current through it. The dimensions of the coil are smaller than the wavelength, hence current flowing through the coil has same phase

When the current flows through the loop, a magnetic field is generated around it. The magnetic field is perpendicular to the loop. Circular and square loop antennas are shown in the following figure





Draw the radiation pattern of loop antenna

FIGURE 9-47 Loop antennas. (a) Circular; (b) square. [Note: The direction of maximum radiation is perpendicular to the plane of the loop; the shape of the radiation pattern is very similar to that in Fig. 9-6 (a).]

The radiation pattern of Loop antenna is a doughnut pattern and no radiation is received that is normal to the plane of the Loop. This type of directional pattern is useful in the direction finding applications.

For DF, it is required to have an antenna that can indicate the direction of particular radiation. Although any of write important Application of loop antenna.

the highly directional antennas could be used for this purpose. But they have disadvantage of being very large, where as loop antenna is not.

Because the loop is very small and DF equipment is portable, hence loops have direction finding as their major application.

Advantages: (1) highly directive (2) smaller in size.

Disadvantage: (1) very low radiation efficiency

Applications: (1) direction finding (2) In portable receivers

Phased Arrays:

A phased array is an array or group of antennas, connected to the one transmitter or receiver, whose radiation beam can be adjusted electronically without any physically moving parts.

In this antenna phases of respective signals feeding the antennas are set in such a way that the effective radiation pattern of array is adjusted in a desired direction and suppressed in undesired directions.

The antennas may be radiators e.g. a large group of dipoles in an array (or array of arrays), pointing in the wanted direction

The main application of phased arrays is in Radar. Recently they have been considered for satellite communications.

State the main application of phased array.

What is phased array?

Blanced modulator gives DSB-FC at its output command

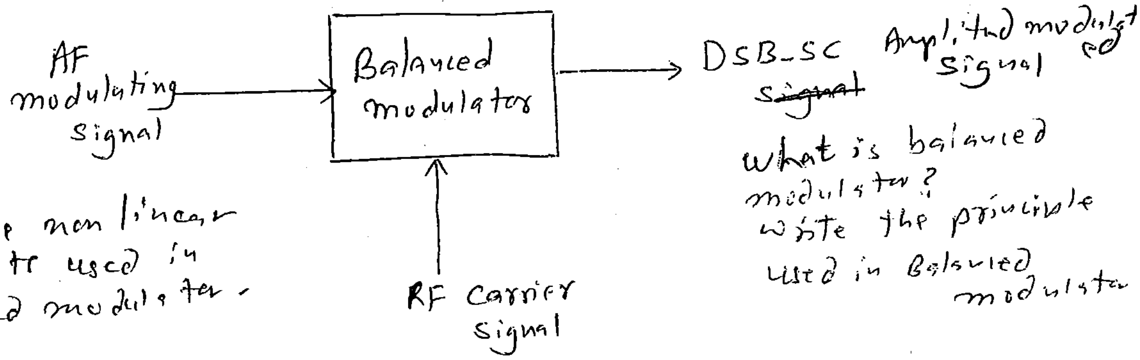
chap 2 Modulation and Demodulation techniques. (1)

Suppression of the carrier [Balanced modulator] gset

What is function of Balanced modulator?

The Balanced modulators are used to suppress the unwanted carrier from the Am signal.

The carrier and modulating signals are the inputs of the Balanced modulator. and we get the DSB signal with suppressed carrier at the output of the balanced modulator. Thus the output consists of upper and lower side bands only.



What is balanced modulator? write the principle used in Balanced modulator

List the non linear components used in Balanced modulator.

Principle used in Balanced modulator:

When two signals of different frequencies (i.e. carrier signal and modulating signal) are passed through a "nonlinear resistance", then at the output Am signal is generated with suppressed carrier.

A device having non linear resistance such as Diode, JFET or bipolar ~~BJT~~ transistor can be used in the Balanced modulator to generate Am signal with suppressed carrier.

Types of Balanced modulator:

Depends upon the methods used ~~to~~ for suppression of carrier, the types of Balanced modulator are given as follows:

(i) Using Diode ring modulator or lattice modulator.

(ii) Using JFET Balanced modulator.

List the Types of Balanced modulators

(2) explain working of balanced modulator using diodes.

Balanced Modulator using Diodes: [Diode Ring modulator]

The method of carrier suppression using the diode ring modulator, which is also called as the lattice modulator. Following figures (a) and (b) shows circuit diagrams of these modulators.

obtained across the secondary of the output transformer T_2.

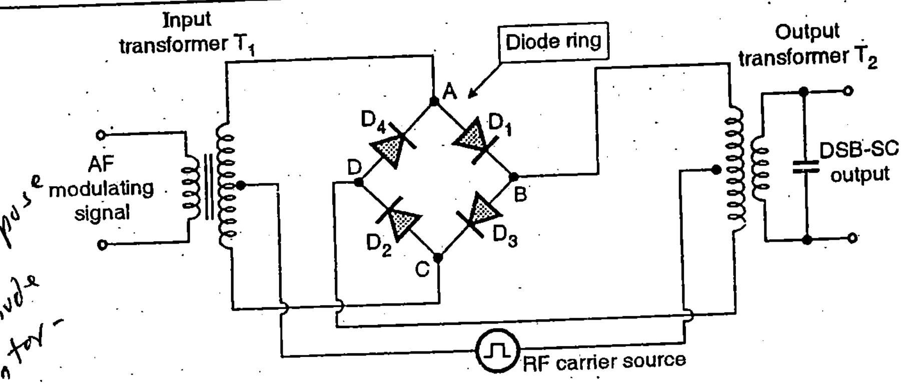


Fig. 4.3.3(a) : Diode ring balanced modulator

what is the purpose of Lattice Type Diode balanced modulator-

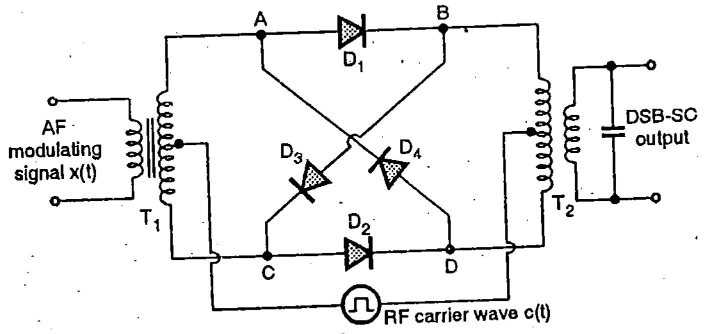


Fig. 4.3.3(b) : Lattice type diode balanced modulator

It consists of an input transformer T1, an output transformer T2 and four diodes connected in a bridge circuit.

The carrier signal is applied to the center taps of the input and output transformers and modulating signal is applied to the input transformer T1.

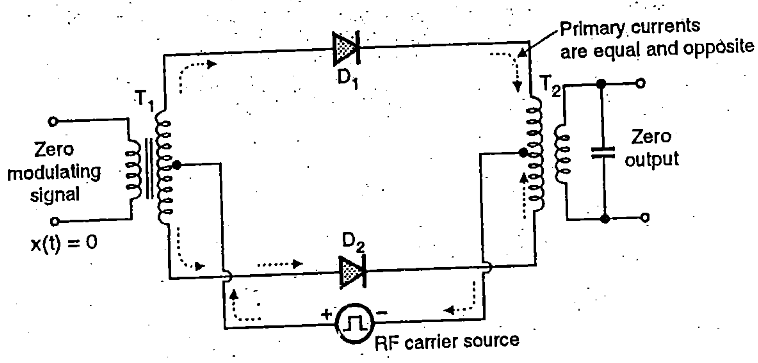
Operation of the Circuit :

mode 1: Carrier suppression: (operation in absence of modulating signal)

To understand how carrier suppression takes place, we assume that the modulating signal is zero. i.e. $x(t) = 0$

Operation in the positive half cycle of carrier:

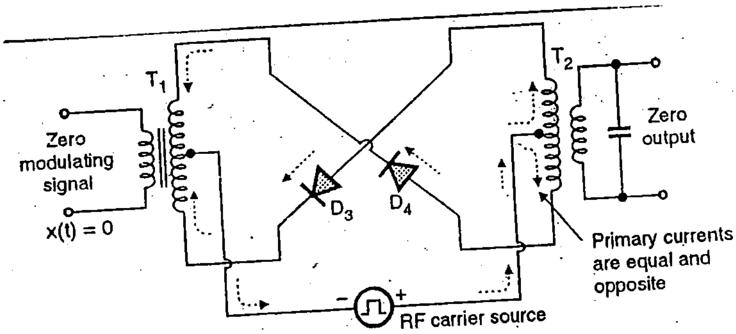
Let us assume that the modulating signal is zero and only the carrier signal is applied. During the positive half cycle of the carrier signal, diodes D1 and D2 are forward biased and diodes D3 and D4 are reverse biased, as shown in the following figure.



From the figure we can easily see that the directions of currents flowing through the primary windings of output transformer T2, are equal and opposite to each other. Therefore the magnetic fields produced by these currents are equal and opposite and cancel each other. Hence the induced voltage in secondary winding of ~~Tra~~ output transformer T2 is zero.

Thus the carrier is suppressed in the positive half cycle.

Operation in negative half cycle of the carrier:



We assume that the modulating signal is zero. In the negative half cycle of the carrier, the Diodes D3 and D4 are forward biased and diodes D1 and D2 are reverse biased.

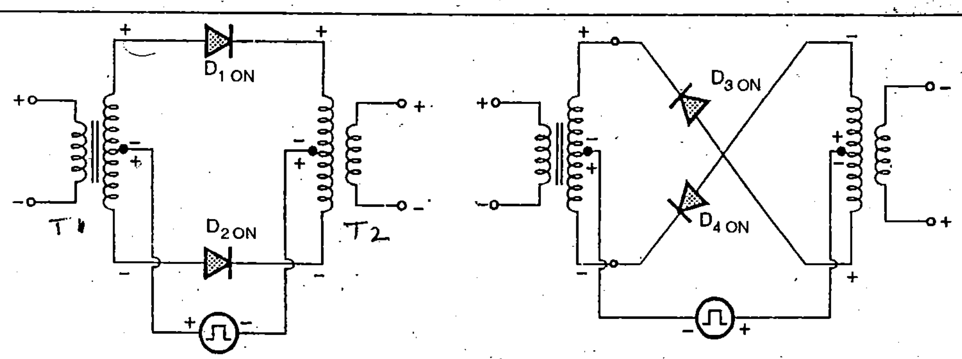
From the above figure, the currents flowing in the upper and lower halves of the primary winding of T2 transformer are again equal and in opposite directions. Hence Similar to positive half cycle, here also magnetic fields in the primary winding of T2 are equal and opposite, cancelling each other. Therefore output voltage at secondary winding of T2 is zero.

Thus the carrier is suppressed in the negative half cycle as well.

Mode II: operation in the presence of modulating signal:

Now let us consider the operation of the circuit when carrier and modulating signals, both are applied.

Operation in the positive half cycle of modulating signal :



(a) Equivalent circuit in the positive half modulating signal with carrier ~~negative~~ positive

(b) Equivalent circuit in the positive half cycle of modulating signal with carrier ~~positive~~ negative

Fig. 4.3.5

(1) As we apply the low frequency modulating signal through the input audio transformer T_1, there are many cycles of the carrier signal, in the positive half cycle of the modulating signal.

(2) In the positive half cycle of the carrier, D_1 and D_2 are on and secondary of T_1 is applied as it is across the primary of T_2. Hence during the positive half cycle of carrier the output of T_2 is positive as shown in Fig. 4.3.5(a).

(3) In the negative half cycle of the carrier, D_3 and D_4 are turned on and the secondary of T_1 is applied in a reversed manner across the primary of T_2 as shown in equivalent circuit of Fig. 4.3.5(b). Thus the primary voltage of T_2 is negative and output voltage also becomes negative.

Operation in the negative half cycle of modulating signal :

When modulating signal reverses the polarities the operation of the circuit is same as that in the positive half cycle of modulating signal discussed earlier. The only difference is now the diode pair D_3 D_4 will produce a positive output voltage whereas D_1 D_2 will produce a negative output voltage as shown in the waveforms of Fig. 4.3.5(c).

5

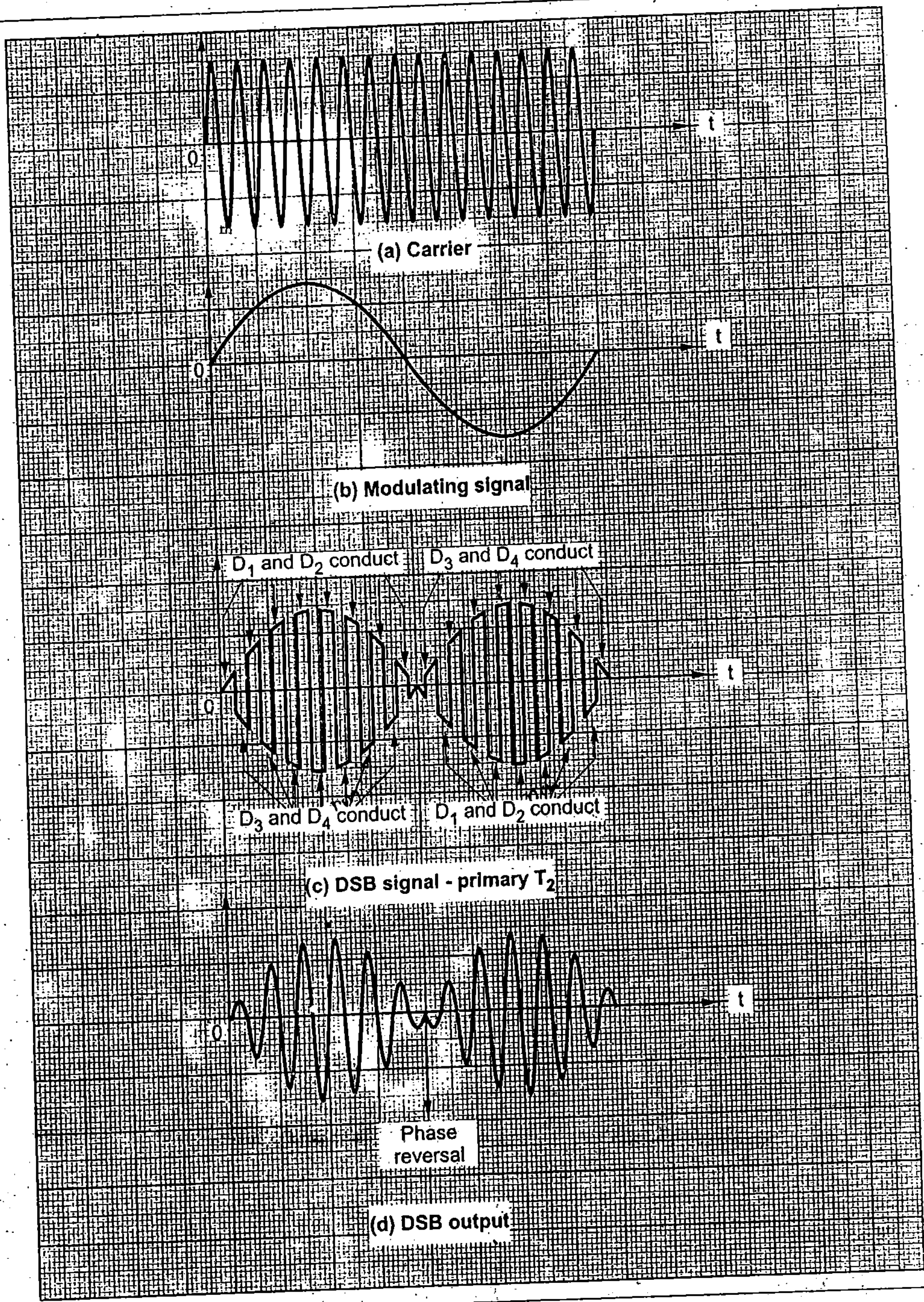
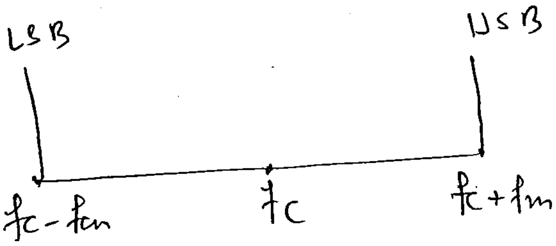


Fig. 3.8 Waveforms in the lattice type balanced modulator



The Am. DSB-SC Signal.

Balanced modulator using FETs (1)

Explain working of Balanced modulator using FETs. Derive the expression for output voltage.

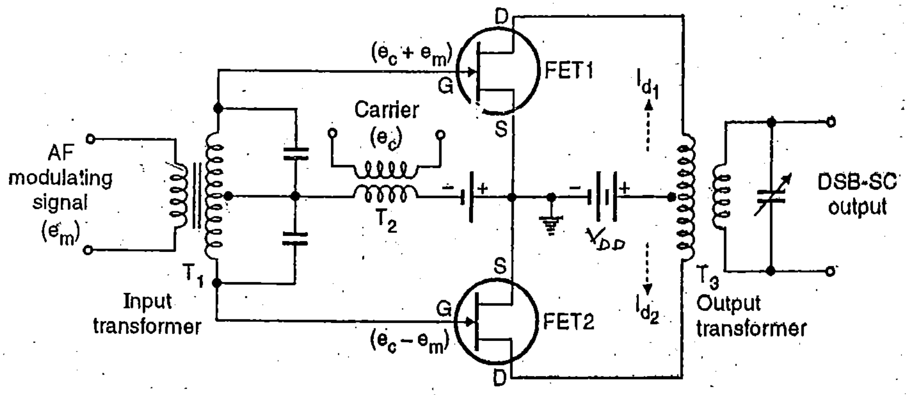


fig (a)

It consists of an input transformers T1 and T2 and an output transformer T3 and two FETS.

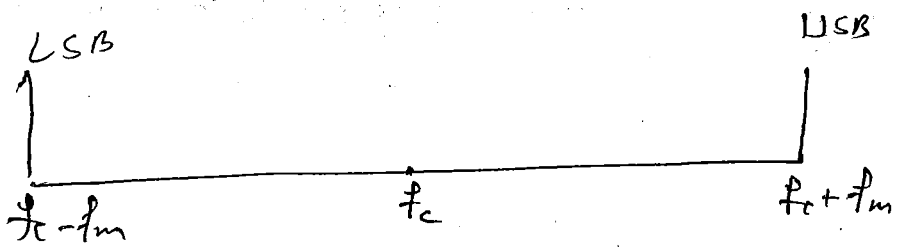
The carrier signal is applied to the center taps of input transformer T1 and output transformer T3, through the transformer T2. The carrier ~~voltage~~ signal is applied to the two gates of FET'S, in phase. The modulating signal is applied to the input transformer T1. The modulating ~~signal voltage~~ signal appears 180° out of phase at the gates of FETS, since these are at the opposite ends of a center tapped transformer T1

Mode I (Operation in the absence of modulating signal) :

- In the absence of modulating signal, both the FETs conduct simultaneously due to the inphase carrier voltage applied to their gates.
- Their drain currents are equal in magnitude but opposite in direction through the primary of output transformer T_3 as shown in the Fig. ~~4.3.6.~~ (a)
- Due to this their magnetic fields cancel each other, inducing a zero secondary voltage. Thus the output of transformer T_3 is zero. The carrier is thus suppressed.

Mode II (Carrier and modulating signal both present) :

- When the modulating signal is applied, the drain currents of the two FETs flow due to the combined effect of carrier and the modulating signal.
- The FET currents due to carrier are equal and opposite and hence cancel each other.
- However FET currents due to the modulating signal are equal but not opposite so they do not cancel out.
- This is because the modulating signal is applied 180° out of phase to the two FETs.
- Hence at the output of the circuit we get a DSB-SC signal.
- For the 100 % suppression of the carrier, both the FETs must have the identical characteristics i.e. it should be a matched pair and the transformer centre taps must be exactly at the centre of the windings.
- Practically this is not possible hence carrier will be heavily suppressed but not completely removed.



Proof :

(7)

~~**Soln. :**~~

where e_m = modulating signal
e_c : carrier signal

- Refer Fig. ~~4.2.6~~ (2) of the FET balanced modulator. As shown, the input voltage at gate of FET 1 will be $(e_c + e_m)$ and it will be $(e_c - e_m)$ at the gate of FET 2.
- The relation between the drain current i_d and gate to source voltage V_{gs} of FET is nonlinear, infact it is almost parabolic. Therefore it is expressed as,

$$i_d = I_D + a\,V_{gs} + b\,V_{gs}^2 \qquad \ldots(1)$$

Here, I_D = dc component of drain current (Q point drain current)

a and b are constants.

- So the drain currents for the two FET are expressed as :

$$i_{d1} = I_{D1} + a\,V_{gs1} + b\,V_{gs1}^2 \qquad \ldots(2)$$

- Substitute $V_{gs1} = (e_c + e_m)$

$$\therefore \quad i_{d1} = I_{D1} + a\,(e_c + e_m) + b\,(e_c + e_m)^2 \qquad \ldots(3)$$

And

$$i_{d2} = I_{D2} + a\,V_{gs2} + b\,V_{gs2}^2 \qquad \ldots(4)$$

Substitute $V_{gs2} = (e_c - e_m)$

$$\therefore \quad i_{d2} = I_{D2} + a\,(e_c - e_m) + b\,(e_c - e_m)^2 \qquad \ldots(5)$$

- Note here that we have considered only the ac voltages while writing the Equations from 1 to 5.

The primary current of output transformer T_3 is given by the difference between i_{d1} and i_{d2}.

$$\therefore \quad i_p = i_{d1} - i_{d2} \qquad \ldots(6)$$

$$= I_{D1} + a\,(e_c + e_m) + b\,(e_c + e_m)^2 - I_{D2} - a\,(e_c - e_m) - b\,(e_c - e_m)^2$$

$$= I_{D1} + a\,(e_c + e_m) + b\,(e_c^2 + e_m^2 + 2\,e_c\,e_m) - I_{D2} - a\,(e_c - e_m) - b\,(e_c^2 - 2\,e_c\,e_m + e\,m^2)$$

$$= I_{D1} + a\,e_c + a\,e_m + b\,e_c^2 + b\,e_m^2 + 2b\,e_c\,e_m - I_{D2} - a\,e_c + a\,e_m - b\,e_c^2 + 2b\,e_c\,e_m - be_m^2$$

$$= I_{D1} + 2\,be_c\,e_m - I_{D2} + 2\,be_c\,e_m + 2a\,e_m$$

$$\therefore \quad i_p = I_{D1} - I_{D2} + 2a\,e_m + 4\,b\,e_c\,e_m$$

- Assuming perfect symmetry,

$$I_{D1} = I_{D2} \qquad \therefore \quad I_{D1} - I_{D2} = 0$$

$$\therefore \quad i_p = 2\,a\,e_m + 4\,b\,e_c\,e_m \qquad \ldots(7)$$

dulator

- But the output voltage is proportional to the primary current i_p.

$$\therefore \quad v_o = x\, i_p$$

where x = constant of proportionality.

- Now substitute i_p from Equation (7) into Equation (8).

$$\therefore \quad \text{output voltage } v_o = 2\, a x e_m + 4\, b x e_c e_m$$

where E_m = maxi peak of m

Where E_m = maximum or peak value of modulating signal

E_c = maximum or peak value of carrier signal

$\omega_m = 2\pi f_m$.

f_m = modulating AF freq.

$\omega_c = 2\pi f_c$

f_c = carrier frequency.

- Now substitute the expressions for modulating and carrier signals $e_m = E_m \cos \omega_m t$ and $e_c = E_c \cos \omega_c t$ in Equation ( ),

$$\therefore \quad v_o = 2\, a x E_m \cos \omega_m t + 4\, b x E_c E_m \cos \omega_c t \cos \omega_m t$$

$$\therefore \quad v_o = A \cos \omega_m t + 2B \cos \omega_m t \cos \omega_c t \quad \text{...(10)}$$

where $A = 2\, a x E_m$ and $B = 2\, b x E_m E_c$

- Simplifying Equation (10) we get,

$$v_o = \underbrace{A \cos \omega_m t}_{\text{Modulating signal}} + \underbrace{B \cos(\omega_c + \omega_m) t}_{\text{Upper sideband}} + \underbrace{B \cos(\omega_c - \omega_m) t}_{\text{Lower sideband}}$$

- The modulating signal in the output will be removed by tuning variable capaci of the output tra
- (T3) Equation (11) proves that the carrier is completely removed and the output consi upper and lower sidebands.

Balanuled modulator is nothing a mixer. Comme

4.3.8 Balanced Modulator as Mixer :

- As seen from Equation (11), the sum and difference frequency compon ts $(f_c + f_m)$ & $(f_c - f_m)$ are produced at the output of a balanced modulator.
- You will study the operation of a mixer in chapter 5 where you will understand signals at different frequencies are applied at the input of a mixer, the sum frequency components are obtained at its output. Heterodyne action
- This is identical to the output of a balanced modulator. Infact both these circ its same principle of nonlinear resistance.

When two signals of different frequencies are applied at the input of mixer then it gives the sum and difference freq. components at its output (Heterodyne action). The same thing is obtained at the output of Balanced modulator i.e. (f_c+f_m) & (f_c-f_m). Hence Balanced modulator is nothing but mixer. Both Balanced modulator and mixer circuits operates on same principle of nonlinear resistance.

4.4 Methods to Suppress the Unwanted Sideband : (SSB Transmitters)

To get single side band signal we have to suppress one of the sideb signal available at the output of Balanced modulator.

➤➤ [University Exam - D

There are three practical methods of suppressing the unwanted sidebands. They are as

(i) Filter method (ii) The phase shift method (iii) The "third" method.

- In all these methods balanced modulators are used to suppress the carrier but ea a different technique to remove the unwanted sideband.
- All the three systems can remove the upper or lower sideband to get a SSB from DSBSC signal. Let us understand the methods one by one.

List the methods to suppress the unwanted sideband.

Why balanced modulator is called as mixer?

Methods to suppress the unwanted side band:
(SSB Transmitter)

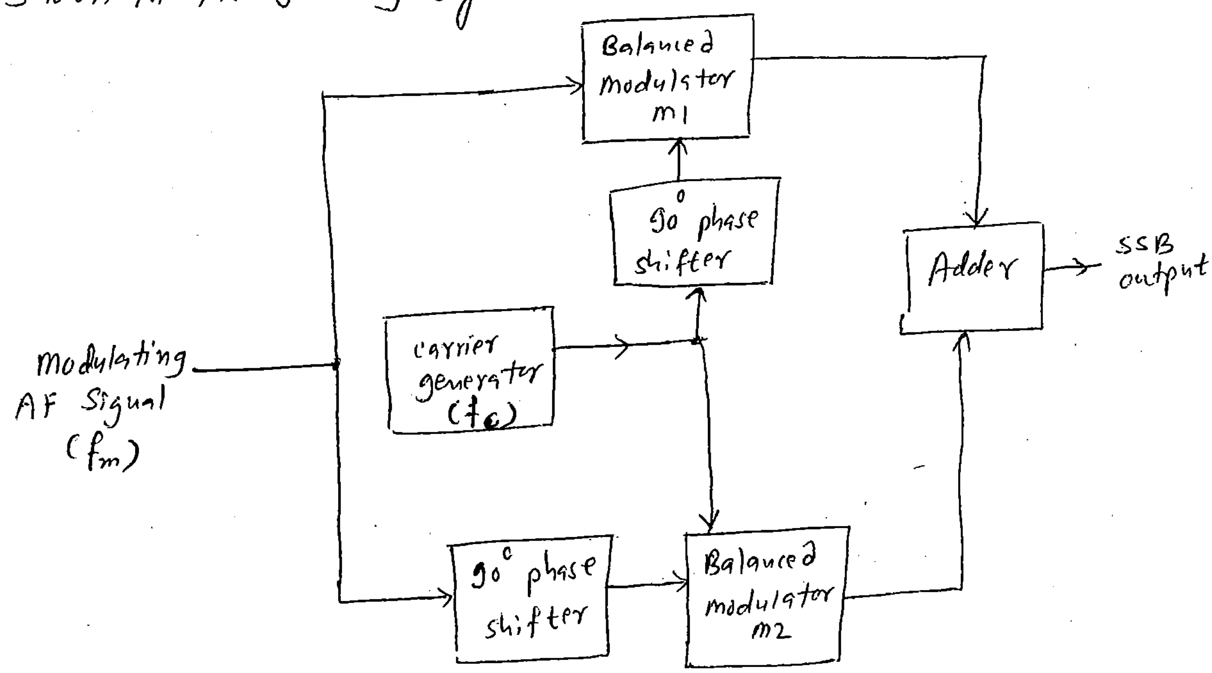
Ex.

Explain phase shift method of side band suppression. Derive the expression for output.

The phase shift method of side band suppression:

The phase shift method of SSB generation uses a phase shift technique that causes one of the sidebands to be cancelled out. This method is used for the suppression of lower sideband (LSB).

The phase shift method uses two balanced modulators m1 and m2 and two 90° phase shifting networks, and adder as shown in the following figure.



Operation :

The balanced modulator m1 has two inputs (i) the modulating signal without any phase shift and (ii) carrier signal with 90° phase shift.

The other balanced modulator m2 has two inputs (i) the modulating signal with a 90° phase shift and carrier signal without any phase shift.

Both modulators m1 and m2 suppress the carrier and produces two sidebands at their outputs. (DSB-SC signals)

The upper sidebands (USBs) at the outputs of both the balanced modulators leads the carrier by 90°. However the LSB (Lower side Band) at the output of Balanced modulator m1 leads the carrier by 90° and the LSB at the output of Balanced modulator m2 lags behind the carrier by 90°. Thus ~~these~~ LSBs are 180° out of phase. Therefore when the outputs of m1 and m2 are applied to the adder the LSBs are cancelled out and the output of the adder consists of only upper sideband (USB) and no carrier and LSB.

Mathematical proof of sideband suppression :

① The inputs to the balanced modulator m1 are

inputs to m1 $\begin{cases} \cos\omega_m t \text{ ----- modulating signal as it is} \\ \cos(\omega_c t + 90^\circ) \text{ ----- } 90^\circ \text{ phase shifted carrier.} \end{cases}$

② The inputs to the balanced modulator m2 are

inputs to m2 $\begin{cases} \cos(\omega_m t + 90^\circ) \text{ ----- } 90^\circ \text{ shifted modulating signal.} \\ \cos\omega_c t \text{ ----- Carrier as it is.} \end{cases}$

(11)

(3) So the output of Balanced modulator M1

$$= \cos(w_c t + 90) \cdot \cos w_m t$$

$$= \frac{1}{2}\cos[w_c t + 90^\circ + w_m t] + \frac{1}{2}\cos[w_c t + 90^\circ - w_m t]$$

$$= \underbrace{\frac{1}{2}\cos[(w_c + w_m)t + 90^\circ]}_{\text{USB with } 90^\circ \text{ advance (leads)}} + \underbrace{\frac{1}{2}\cos[(w_c - w_m)t + 90^\circ]}_{\text{LSB with } 90^\circ \text{ advance (leads)}}$$

(4) and the output of Balanced modulator M2

$$= \cos(w_c t) \cdot \cos(w_m t + 90^\circ)$$

$$= \frac{1}{2}\cos[w_c t + w_m t + 90^\circ] + \frac{1}{2}\cos[w_c t - (w_m t + 90)]$$

$$= \underbrace{\frac{1}{2}\cos[(w_c + w_m)t + 90^\circ]}_{\text{USB with } 90^\circ \text{ advance (leads)}} + \underbrace{\frac{1}{2}\cos[(w_c - w_m)t - 90^\circ]}_{\text{LSB with } 90^\circ \text{ delay (lags)}}$$

(5) output of adder = output of M1 + output of M2

$$= \cos[(w_c + w_m)t + 90^\circ]$$

USB only

Where $W_m = 2\pi f_m$

$W_c = 2\pi f_c$

f_m: Frequency of modulating signal

f_c: Frequency of carrier signal.

(3) The Third method: (12)

The block diagram of the "Third method" of SSB generation is given as follows:

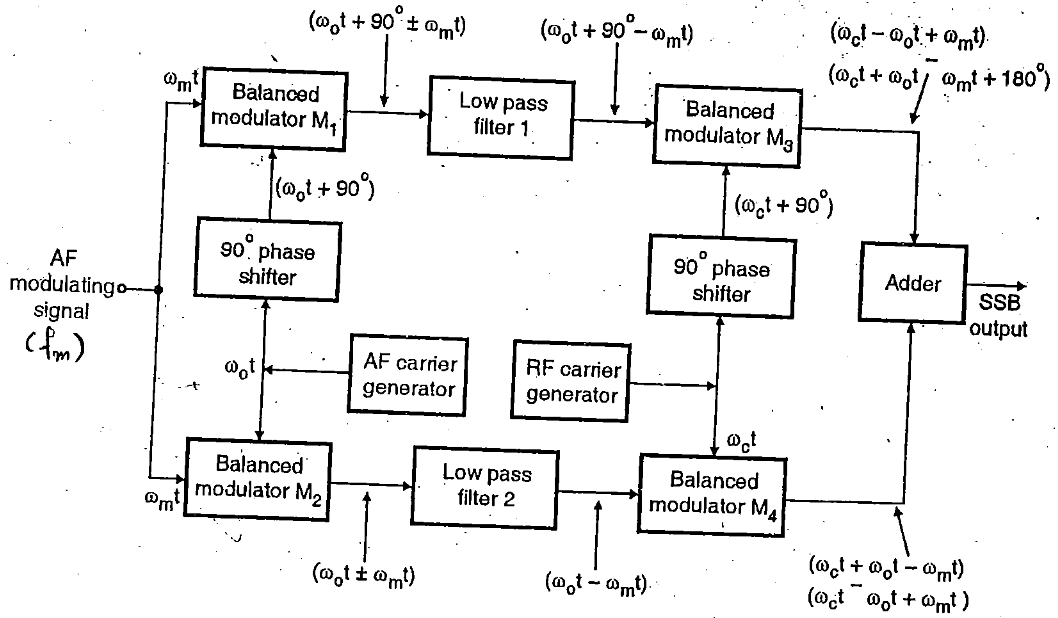


Fig. 4.4.1(e) : Third method to generate the SSB

- This method is very much similar to the phase shift method and it has all the advantages of the phase shift method.
- However it has an added advantage over the phase shift method that the 90° phase shifter for the AF modulating signal is not at all required.

This method uses four balanced modulators, two carrier generators and two 90° phase shifters and adder as shown in Fig. 4.4.1(e).

State advantage of third method over phase shift method to generate AM SSB SC signal

Operation :

The operation of the system is as follows :

(i) The modulating signal does not undergo any phase shift in this method. Instead of that an AF carrier at frequency f_o is generated. f_o is generally at the middle of the AF range (say 1650 Hz). This signal is applied as it is to balanced modulator M_2 and after a 90° phase shift it is applied to the balanced modulator M_1.

(ii) The modulating signal is applied to both M_1 and M_2.

Their outputs are as follows :

$$\text{output of } M_1 = \cos[\omega_o t \pm \omega_m t + 90^\circ] \quad .....(4.4.8)$$

$$\text{output of } M_2 = \cos[\omega_o t \pm \omega_m t] \quad .....(4.4.9)$$

(iii) The low pass filters after M_1 and M_2 have a cutoff frequency at f_o. So they will pass the LSBs and attenuate USBs.

$$\text{output of } LPF_1 = \cos\left[\omega_o t - \omega_m t + 90^\circ\right] \text{ and} \quad .....(4.4.10)$$

Explain the third method of side band suppression. Derive the expression for output compare between phase shift and third method of side band suppression. (4m)

$$\text{output of } LPF_2 = \cos\left[\omega_o t - \omega_m t\right] \quad (13) \qquad .....(4.4.11)$$

(iv) The other pair of the balanced modulators M_3 and M_4 receives the carrier without and with 90° shift respectively along with the filter outputs. Their outputs are :

$$\text{output of } M_3 = \cos\left[\omega_c t + \omega_o t - \omega_m t + 90^\circ + 90^\circ\right] \quad \text{USB}$$

$$\text{and} \quad \cos\left[\omega_c t - \omega_o t + \omega_m t - 90^\circ + 90^\circ\right] \quad \text{LSB}$$

$$\therefore \quad \text{output of } M_3 = \cos\left[\omega_c t + \omega_o t - \omega_m t + 180^\circ\right] \quad \text{USB} \qquad .....(4.4.12)$$

$$\text{and} \quad \cos\left[\omega_c t - \omega_o t + \omega_m t\right] \quad \text{LSB} \qquad .....(4.4.13)$$

M3

$$\text{and output of } M_4 = \cos\left[\omega_c t + \omega_o t - \omega_m t\right] \quad \text{USB} \qquad .....(4.4.14)$$

$$\text{and} \quad \cos\left[\omega_c t - \omega_o t + \omega_m t\right] \quad \text{LSB} \qquad .....(4.4.15)$$

M4

(v) All these signals are added in the adder circuit. As seen from Equations (4.4.12) and (4.4.14) the USBs are out of phase hence cancel each other. Equations (4.4.13) and (4.4.15) show that the LSBs are in phase hence add together.

Thus this circuit will suppress the USB and generate SSB with LSB in the output.

$$\therefore \quad \text{Adder output} = \cos\left[\omega_c t - (\omega_o t - \omega_m t)\right] \qquad .....(4.4.16)$$

LSB only

Advantages of third method :

- The third method has all the advantages of phase shift method.
- The additional advantage is that as the modulating signal need not be phase shifted, the design of 90° phase shifter becomes very simple. [This circuit has to provide the 90° phase shift at only one frequency i.e. f_o]

Disadvantages : This system is complicated and rarely used.

4.4.4 Comparison between the Sideband Suppression Methods :

➤➤➤ [University Exam - May 98, Dec. 2003 !!!]

Sr. No.	Parameter	Filter method	Phase shift method	Third method
(1)	Method to cancel the unwanted sideband.	Using a filter.	By shifting AF and RF signals to BM by 90°.	Same as phase shift method in principle.
(2)	Design of 90° shifter at modulating frequency.	Not Applicable	Design is critical.	Design is easy as it is to be done at a single frequency.
(3)	Possibility of SSB generation at any frequency.	Not possible to generate at any frequency.	Possible.	Possible.
(4)	Need of up conversion.	Needed.	Not needed.	Not needed.
(5)	Use of low modulating frequencies.	Not possible.	Possible.	Possible.
(6)	Need of linear amplifiers.	Needed.	Needed.	Needed.
(7)	Critical points in system design.	Filter characteristics, its size and weight, cutoff frequency.	Design of 90° phase shifter for modulating frequency. Symmetry of balanced modulators.	Symmetry of balanced modulators for the proper carrier cancellation.

List advantages and disadvantages of third method of side band suppression.

Compare the All the methods of sideband suppression.

4.4.1 Filter Method :

➤➤➤ [University Exam - May 98, May 2006 !!!]

The block diagram for the filter method is as shown in Fig. 4.4.1(a).

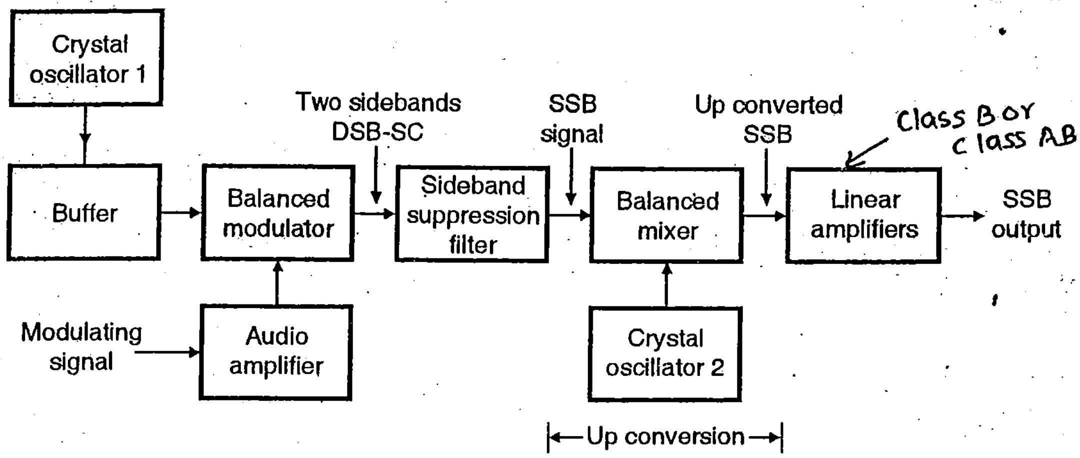


Fig. 4.4.1(a) : Filter method for sideband suppression

Operation :

(i) The modulating signal is amplified using an audio amplifier and applied to the balanced modulator. The other input to the balanced modulator is the carrier signal generated by the crystal oscillator 1. This carrier frequency is much less as compared to the carrier which is to be actually transmitted.

(ii) The balanced modulator will suppress the carrier and will produce upper and lower sidebands at its output.

(iii) Out of the two sidebands the unwanted sideband is heavily attenuated by the sideband suppression filter and the other sideband is passed through without much attenuation.

(iv) The frequency of this sideband is very low. So the frequency is boosted up to the transmitter frequency by the combination of the balanced mixer and second crystal oscillator. This is called as frequency up conversion.

(v) This signal is then amplified using linear amplifiers (class B or AB). Linear amplifiers are used to avoid any waveform distortion.

Why to modulate at low frequency and use frequency up conversion ?

- The filter used to suppress the unwanted sideband must have a flat pass band and very high attenuation outside the passband. ~~And~~ The filter response must change from zero attenuation to full attenuation over a range of ~~about 600 Hz~~ Low frequencies. The filter operation is best at ~~low~~ low frequencies
- To fulfill these requirements at very high operating frequency the Q of the tuned circuit used in the filter must be very high. This is practically not achievable.
- Hence the modulation is carried out at low frequency and then the frequency is raised by means of up conversion.

Types of filters used :

- The LC filters, crystal filters, ceramic or mechanical filters can be used for removing the unwanted sideband.
- The crystal or ceramic filters are cheap but technically better only above 1 MHz operating frequency.
- The mechanical filter has the best properties i.e. small size, good passband and good attenuation characteristics.

Advantages of filter method :

(i) This method gives the adequate sideband suppression. The sideband filter also helps to attenuate the carrier.

(ii) The bandwidth is sufficiently flat and wide.

(iii) This method is simpler as compared to other methods.

Disadvantages :

(i) Due to the inability of the system to generate SSB at high radio frequencies, the frequency up conversion is necessary.

(ii) Low audio frequencies can not be used as the filter becomes bulky.

(iii) Two expensive filters are to be used one for each sideband.

Frequency spectrums at various points :

- The frequency spectrums at various points for the filter method are as shown in Fig. 4.4.1(b).

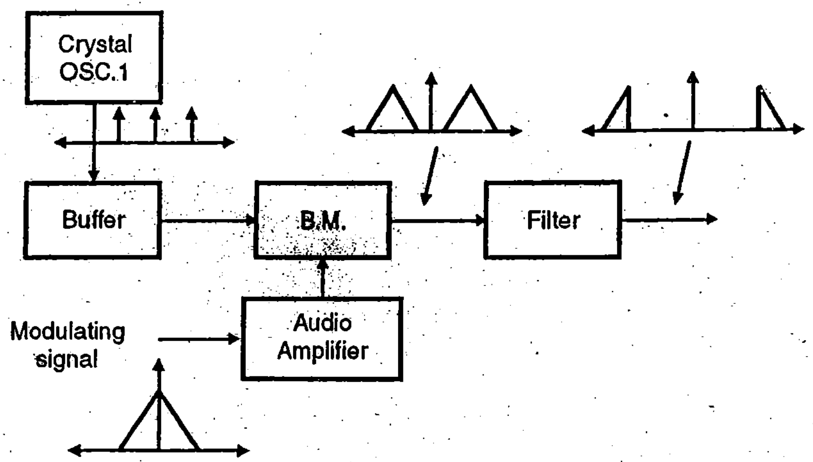


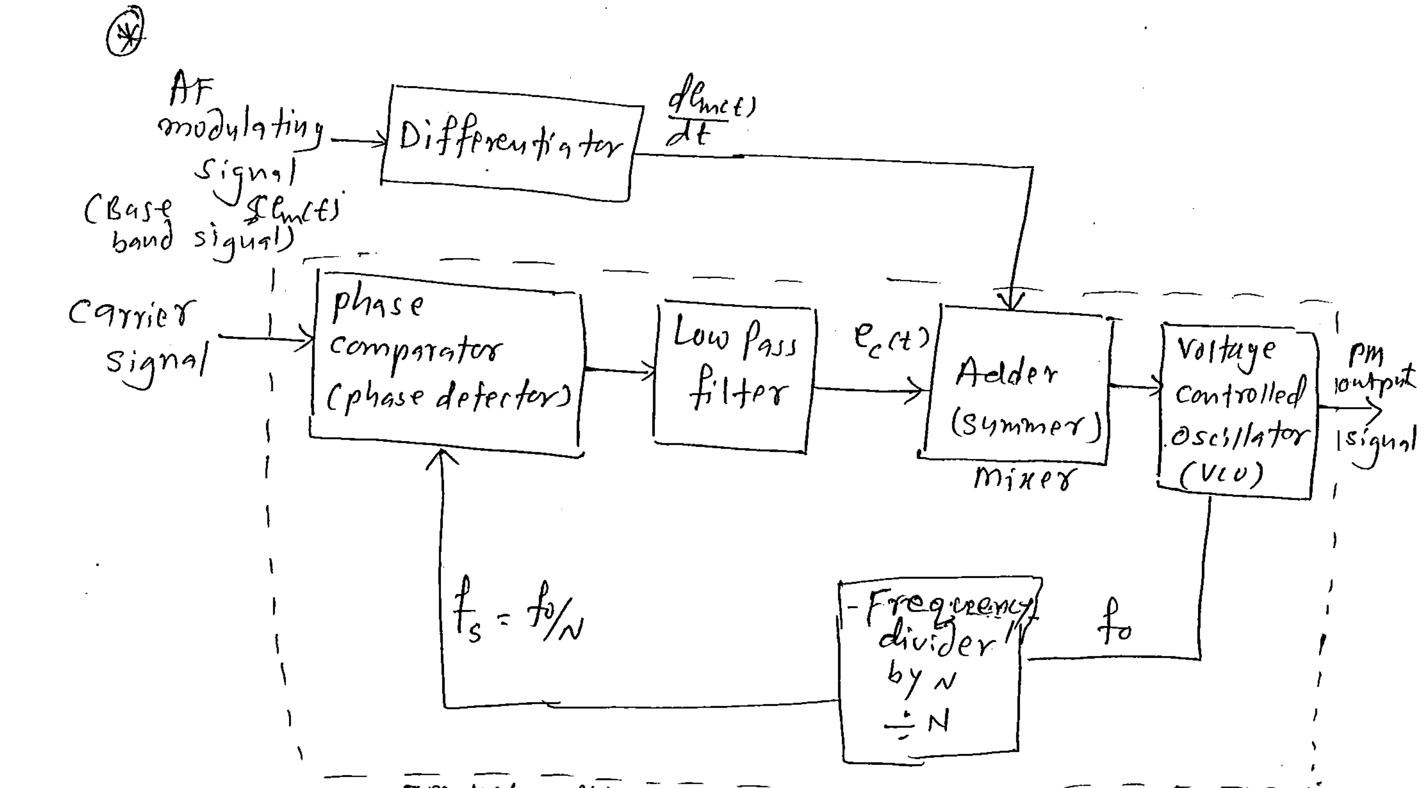
Fig. 4.4.1(b) : Frequency spectrums at various points for filter method

4.4.2 Phase Shift Method :

➤➤➤ [University Exam - May 98, May 99, May 2004 !!!]

- The block diagram for the phase shift method of SSB generation is as shown in Fig. 4.4.1(c). This system is used for the suppression of lower sideband.
- This system uses two balanced modulators M_1 and M_2 and two 90° phase shifting networks as shown in the Fig. 4.4.1(c).

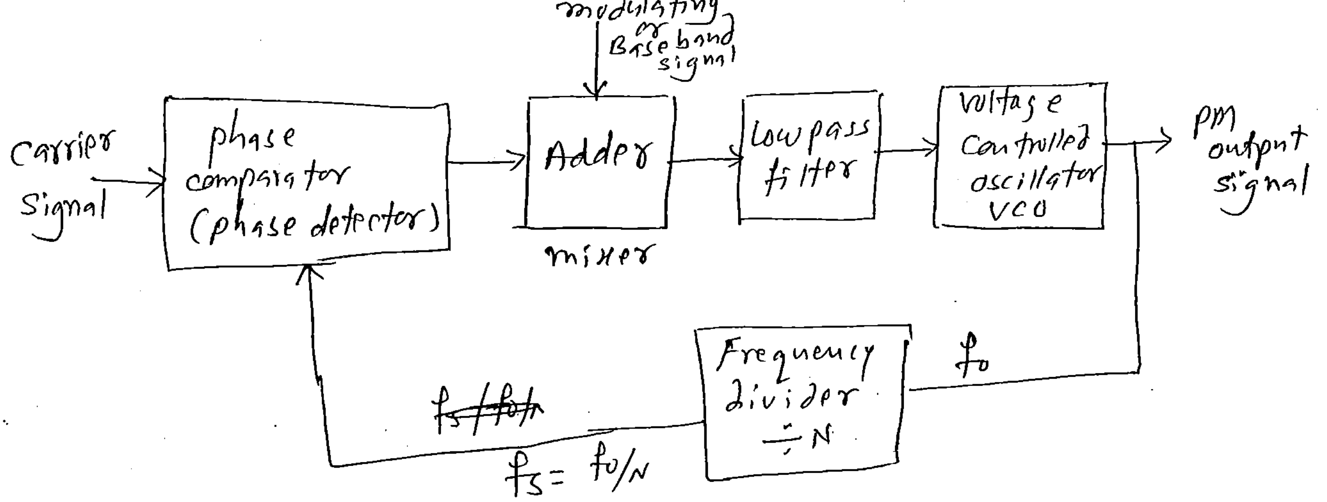
Phase modulation using Phase Locked Loop (PLL) :-



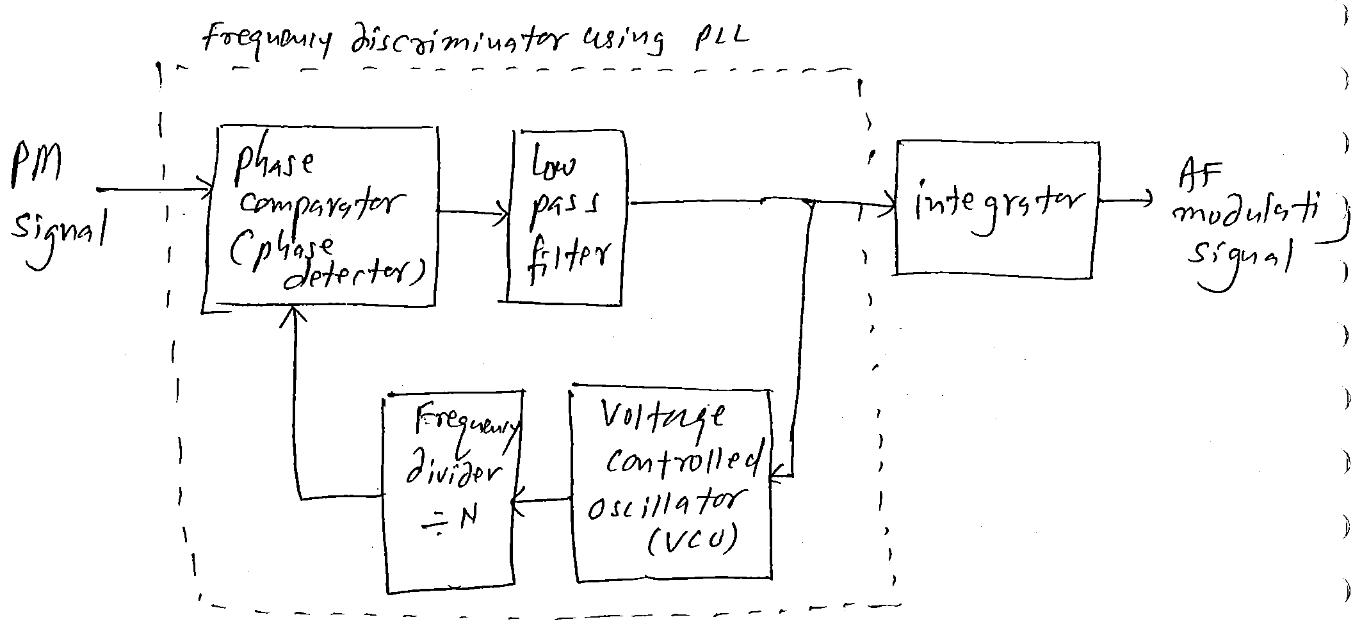
FM using PLL

The AF modulating signal $e_m(t)$ is first differentiated. The carrier signal is applied to phase comparator (detector) of PLL. The output of VCO of PLL gives phase modulated (PM) signal. Here PLL is used as frequency modulator.

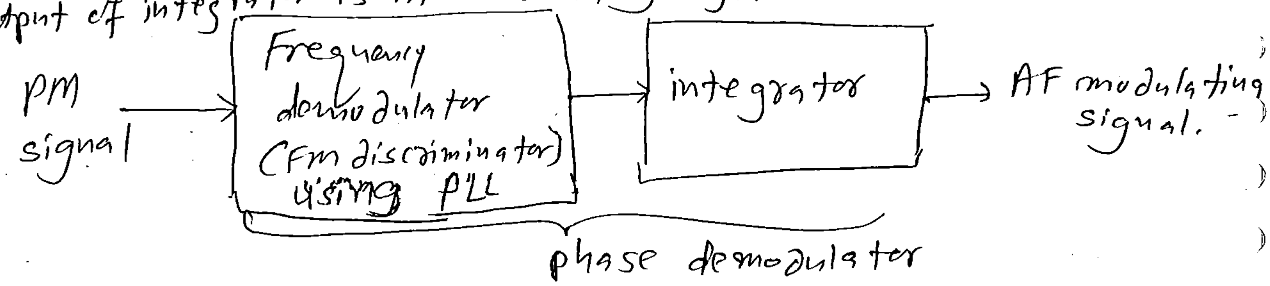
Another method is that AF modulating or baseband signal is added before low pass filter, then there is no need for the differentiator.

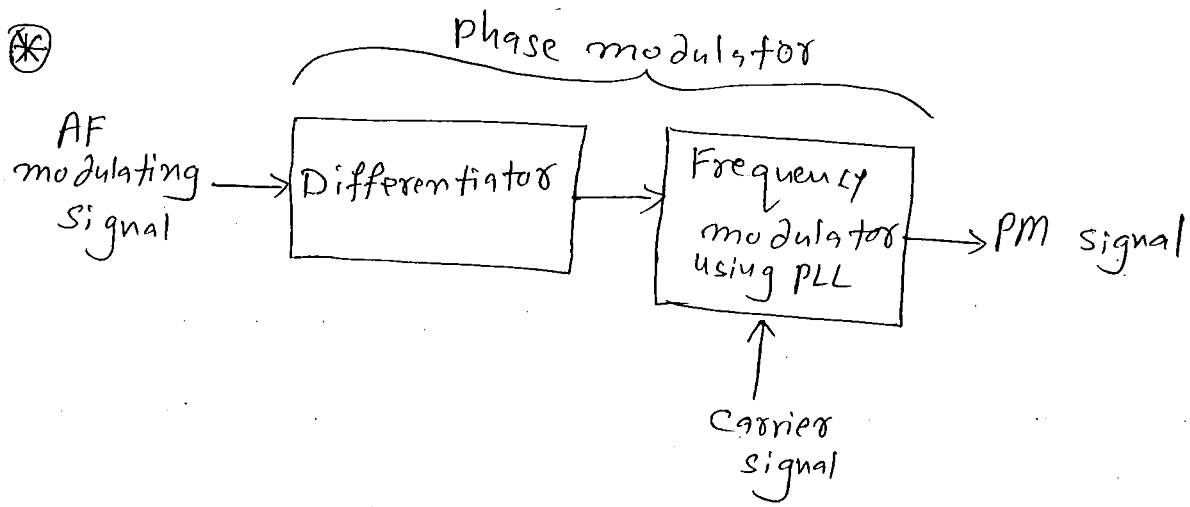


phase demodulation using PLL :



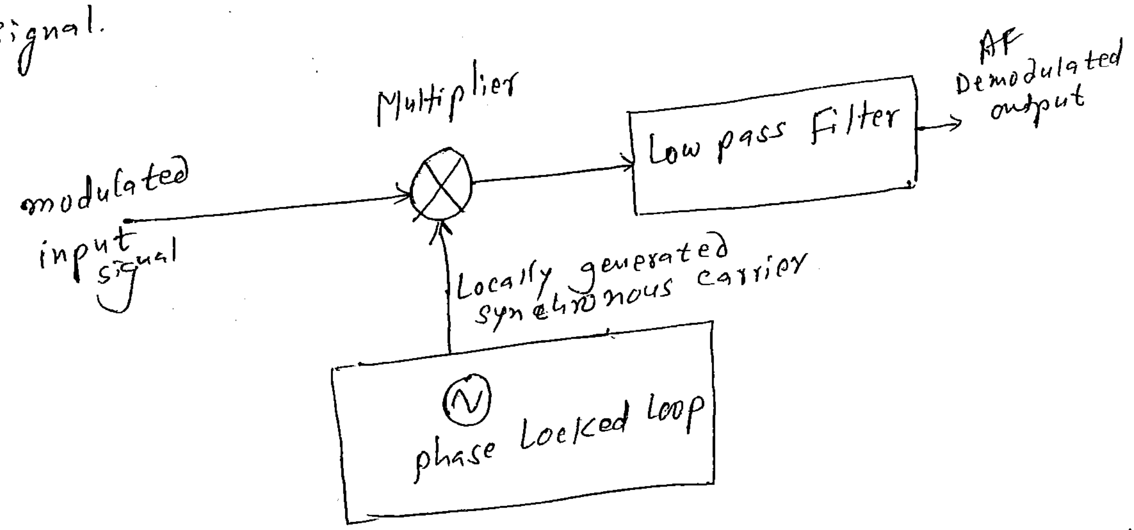
The phase modulated signal is ~~first~~ applied to phase comparator of PLL. Here PLL is used as Frequency demodulator. The output of low pass filter of PLL is given to integrator circuit. The output of integrator is AF modulating signal.

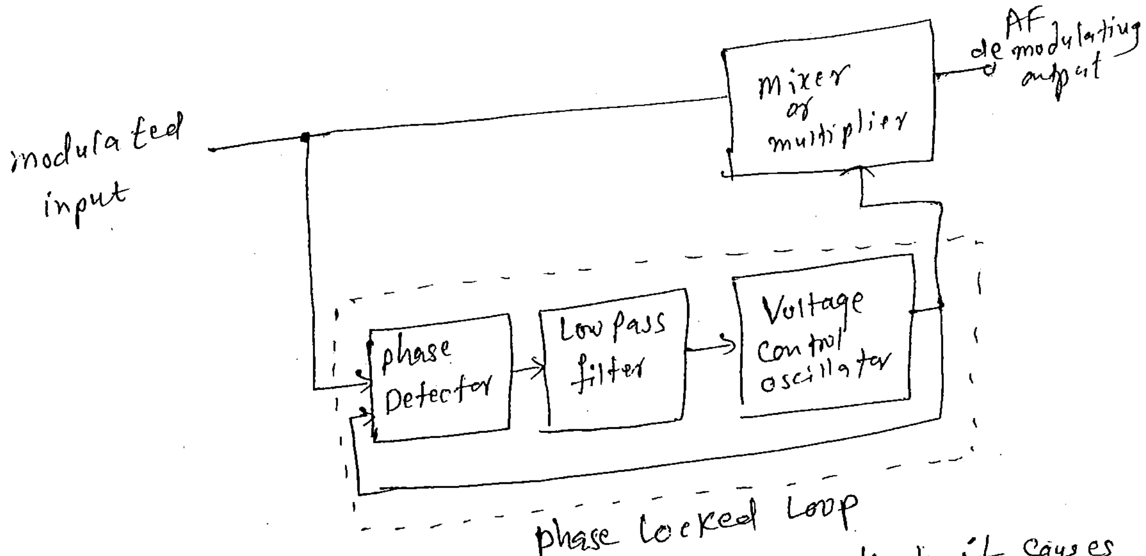




Envelope detectors are the simplest form of detector, whereas synchronous detectors are considerably more complex than envelope detectors. They consists of phase locked loop (PLL) and multiplier circuit.

Demodulation is performed by multiplying the modulated input by a sine wave that is phase locked to the carrier of incoming modulated signal.





The advantage of synchronous detection is that it causes less distortion than envelope detection and works well with single sideband signals

...rnal resistance R ... of the detector is supplied by a source

The sync. detectors provides increased immunity from selective fading of the carrier relative to the side bands. This advantage can be significant, when the reception of sky wave signals is considered.

Advantages: (i) cancellation and reduction in noise and ~~inter~~ interference.

(2) increased immunity from selective fading and phase shifting between side bands and

(3) improved signal to noise ratio.

In other words selective fading, overmodulation due to carrier fading, ~~heterodyning~~, phase shift distortion and adjacent channel interference can be reduced

One ~~nice~~ advantage thing about envelope detectors (in addition to their low cost) is that they are not sensitive to the phase of carrier of modulated signal.

"Synchronous Detector is ~~used~~ works well with DSB-FC"

Comment

6.8.2 Envelope Detector :

(AM Demodulator)

- The envelope detector is a simple and very efficient device which is suitable for the detection of a ~~narrowband~~ AM signal. A ~~narrowband~~ AM wave is the one in which the carrier frequency f_c is much higher as compared to the ~~bandwidth~~ frequency of the modulating signal. (f_m)
- An envelope detector produces an output signal that follows the envelope of the input AM signal exactly.
- The envelope detector is used in all the commercial AM radio receivers.

(14)

Circuit Diagram :

- The circuit diagram of the envelope detector is shown in Fig. 6.8.3. It consists of a diode and RC filter.

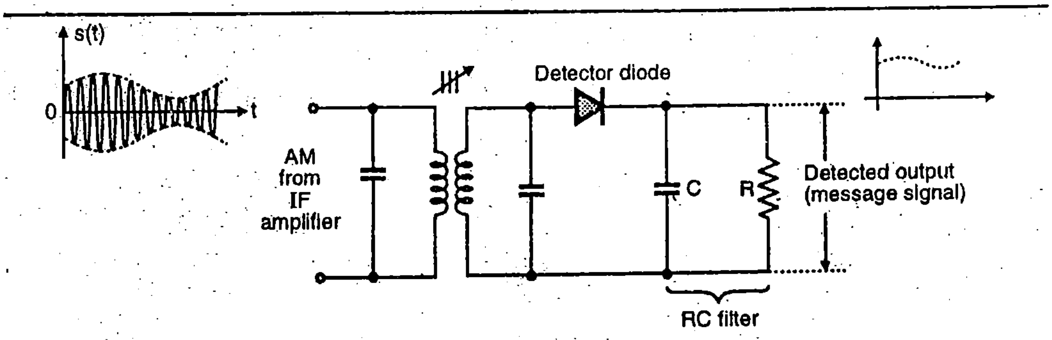


Fig. 6.8.3 : Envelope detector for detection of AM wave

Operation of Envelope Detector :

1. The standard AM wave is applied at the input of the detector.
2. In every positive half cycle of the input the detector diode is forward biased. It will charge the filter capacitor C connected across the load resistance R to almost the peak value of the input voltage. As soon as the capacitor charges to the peak value, the diode stops conducting. The capacitor will discharge through R between the positive peaks as shown in the Fig. 6.8.4.
3. The discharging process continues until the next positive half cycle. When the input signal becomes greater than the capacitor voltage, the diode conducts again and the process repeats itself.

Waveforms :

- The input-output waveforms for the envelope detector are shown in Fig. 6.8.4. It shows the charging discharging of the filter capacitor and the approximate output voltage.

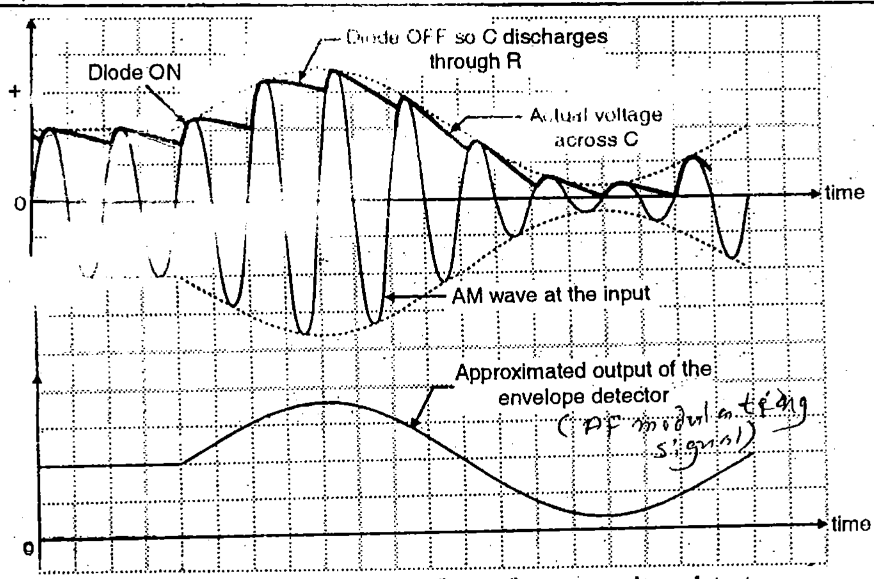


Fig. 6.8.4 : Input output waveforms for an envelope detector

- It can be seen from these waveforms, that the envelope of the AM wave is being recovered successfully.
- We assume that the diode is ideal which presents a zero resistance when it is ON and infinite resistance when it is OFF.
- We also assume that the AM wave applied to the input of the detector is supplied by a source having internal resistance R_s.

(15)

6.8.5 Synchronous Detection :

- This is another type of detection/demodulation method. for AM signal
- Fig. 6.8.8 shows the block diagram for the synchronous demodulator.

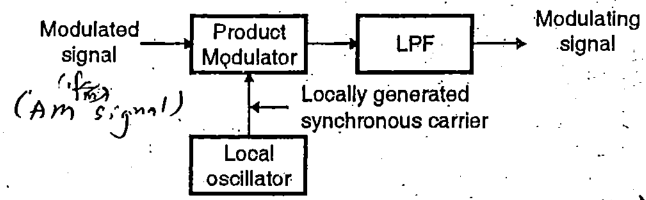


Fig. 6.8.8 : Synchronous Detection (AM)

- The principle of synchronous detection states that the input modulated signal is multiplied with a locally generated phase and frequency synchronous carrier at the receiver.
- The product is passed through a low pass filter to pass only the modulating signals.
- The local oscillator of Fig. 6.8.8 generates a carrier wave, the frequency of which is equal to the carrier frequency and it is phase synchronised with the carrier at the transmitter.
- Product modulator is basically a multiplier which multiplies the input modulated signal and the locally generated carrier.
- At the output of the multiplier we obtain various frequency components one of which is the modulating (baseband) signal.
- It is allowed to pass through by the LPF.

1) Explain synchronous detection with suitable block diagram.
2) List the advantages of -
3) List the disadvantages of synchronous detection
4) Compare synchronous and envelope detection.

Advantages :

1. Better quality of demodulation
2. Less effect of noise.

Disadvantages :

1. Synchronization between transmitter and receiver is necessary.
2. Phase and frequency errors may get introduced.
3. More complexity.

6.8.6 Comparison of Synchronous and Envelope Detection :

➤➤➤ [University Exam - Dec. 2006 !!!]

Refer Table 6.8.1 for the comparison.

Table 6.8.1

Sr. No.	Parameter	Synchronous Detection	Evelope Detection
1.	Locally generated synchrounous carrier.	Used	Not required
2.	Block schematic	Refer Fig. A	Refer Fig. B.
3.	Types of errors	Frequency and phase error.	Envelope ~~distortion~~ and diagonal clipping.
4.	Operating principle	Multiply the incoming modulating AM signal with locally generated sync carrier and pass the product through a LPF.	Rectify the incoming modulating AM signal and pass it through a LPF.
5.	Complexity	High	Simple
6.	Synchronization with the transmitter	Necessary	Not necessary.
7.	Used for	DSB-SC, SSB, VSB	DSB-FC.

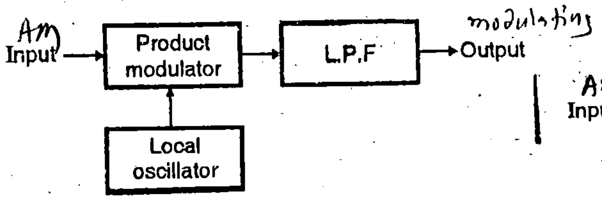


Fig. A : Synchronous detection

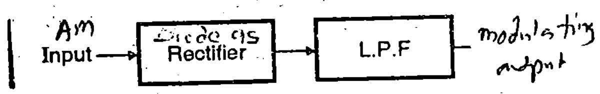


Fig. B : Envelope detection.

Write the operating principle of synchronous Detector. (2m)

AM Synchronous Demodulation / Detection:

Synchronous Am demodulation is generally used for higher performance radio receivers.

The simplest form of detection for amplitude modulated signal uses a simple diode rectifier. To obtained improved performance, a ~~demodulation~~ Synchronous demodulation can be used.

Advantages and Disadvantages of Am synchronous Demodulators:

Advantages:

(1) Increased Linearity

(2) Lower levels of distortion

(3) Considerably less affected by selective fading experienced on the medium and short wave bands.

(4) Improved sensitivity.

(5) Improved signal to noise ratio.

Disadvantages:

(1) Complexity of the circuit, although this is not an important consideration, if the synchronous detector can be included in an IC

Product detector:

A product detector is a type of demodulator used for AM and SSB signal.

Rather than converting the envelope of the signal into the decoded waveform like an envelope detector, the product detector takes the product of the modulated signal and a local oscillator signal, hence the name.

A product detector is a frequency mixer.

Demodulation of SSB:

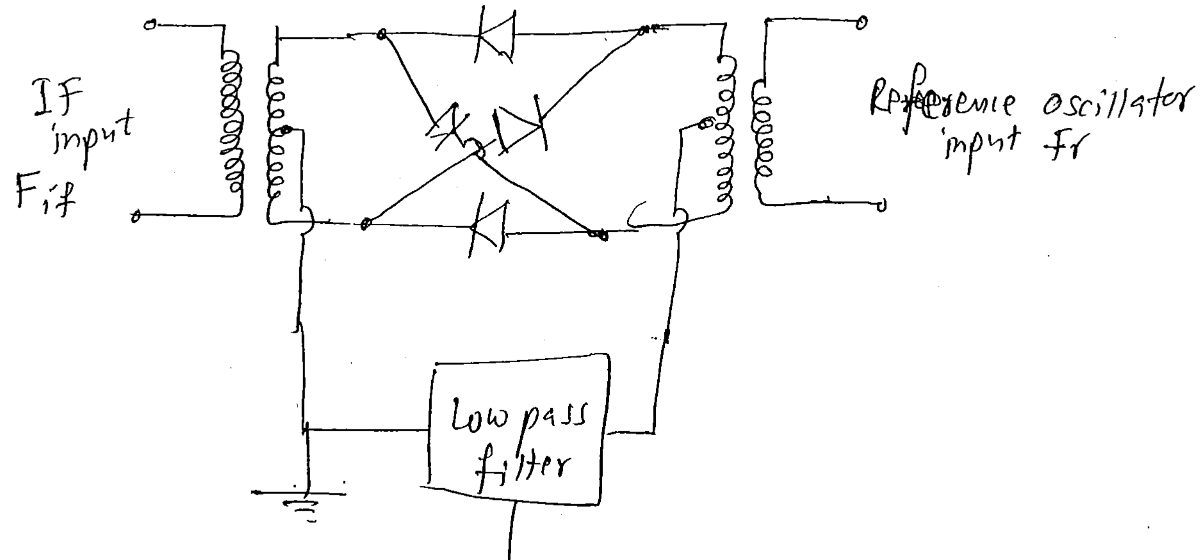
Demodulation of SSB is different from ordinary AM detection. The basic SSB demodulation circuit is the product detector, which is ~~ver~~ similar to an ordinary mixer.

A product detector is simply a balanced modulator or balanced mixer circuit. A product detector can be used to demodulates SSB signals, DSB signals, and standard AM signals.

The mixer may be ~~eg~~ either a diode mixer or transistor mixer.

It has one input from the IF amplifier output and the second input from reference crystal or other oscillator.

The frequency of the reference oscillator is the same as the carrier frequency. The difference frequency for the mixer is the desired modulating frequency, which can be obtained using a simple low pass filter.



Demodulation of a single side band (SSB) signal is obtained by multiplying it with a locally generated synchronous carrier signal at the receiver. Detectors using this principle are ~~f~~ called product detectors and balanced modulator circuits are used for this purpose.

It is important that the carrier closely synchronized in frequency and phase with the original carrier as possible to avoid distortion of ~~modulation~~ modulated output.

To show that the multiplying process does demodulate an SSB signal.

~~Consider lower side band frequency~~ signal

Consider the Lower side Band (LSB) signal $E_{max} \cdot \cos(\omega_c - \omega_m)t$ multiplied by a local oscillator signal $E_{Cmax} \cdot \cos\omega_c t$ using a balanced modulator, with gain K

thus $e_{out} = K \cdot E_{max} \cos(\omega_c - \omega_m)t \cdot \cos\omega_c t$

$$= \frac{K \cdot E_{max}}{2} \cdot \left[\cos\omega_m t + \cos(2\omega_c t - \omega_m)t\right]$$

The first term on the right of the equation is the required information signal, while the second term is the lower side frequency at the second harmonic of local carrier frequency. Low pass filtering easily removes this, leaving only the demodulated information (base band) signal as

$$e_{bb}(t) = \frac{K \cdot E_{max}}{2} \cdot \cos \omega_m t$$

Draw backs of product demodulator:

(1) The frequency of local oscillator must be the same as the frequency of original ~~frequency~~ carrier signal, otherwise demodulated output will fade in case of AM or frequency shifted in case of SSB.

(2) Once the frequency is matched, the phase of carrier signal must be matched with phase of original carrier signal, Otherwise demodulated signal will be attenuated and noise remains as it is.

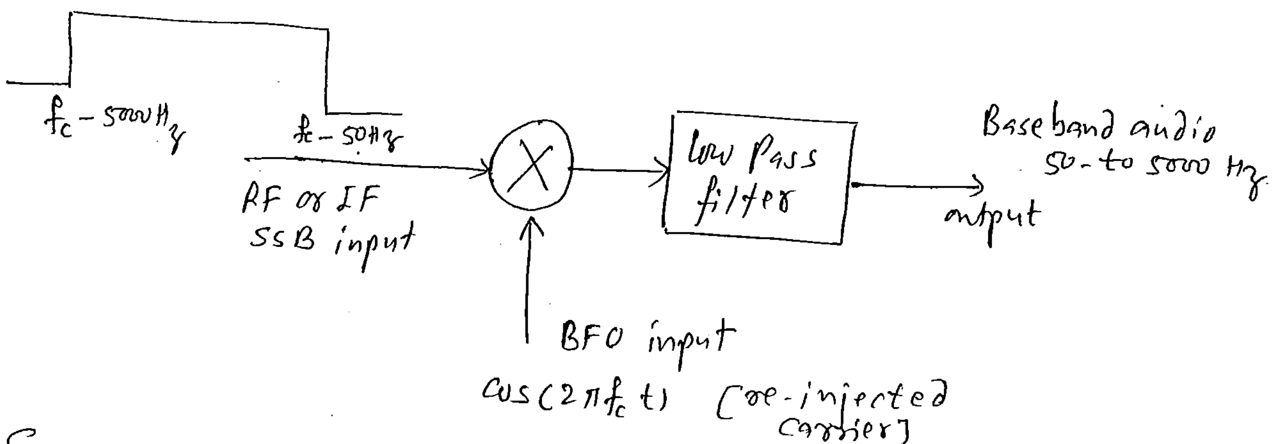
A product detector for SSB, DSB demodulation:

A product detector is a mixer used to down convert an input modulated signal to orignal or baseband signal. The term product detector is generally used, when referring to Single side Band (SSB) or Double side band (DSB) demodulation.

It is a detector, whose output is approximately equal to the product of the beat frequency oscillator (BFO) and RF modulated signals applied to it.

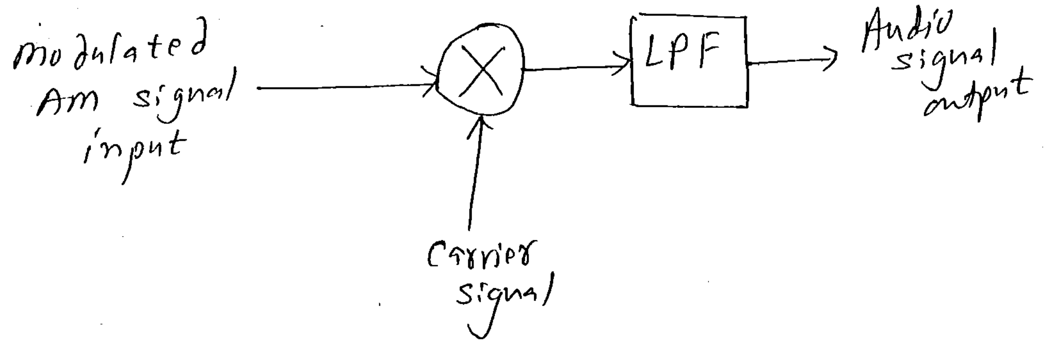
The output from the product detector is modulating AF signal. Following figure shows a product detector used to demodulate a SSB signal.

The low pass filter that follows the mixer, passes only the down conversion or difference frequency band.



Since the SSB or DSB the signal is transmitted with no carrier or with supressed carrier, the frequency of re-injected carrier in above figure will not be exactly the same as that of carrier, that was suppressed in the generation of SSB or DSB signal. If the frequency of the re-injected carrier is not sufficiently close, then there will be error at the audio output signal.

For Am demodulator, a balance modulator can be used. Then this type of modulator is called synchronous detector or product detector. Following figure shows block diagram of product detector. Here modulated Am signal is multiply by synchronised carrier signal.



let $e_{Am}(t)$ be modulated Am signal and $e_c(t)$ be the carrier signal.

$$e_{Am}(t) = A_{DC}\left[1 + m_a \cdot \cos(2\pi f_m t)\right] \cdot \left[A_c \cdot \cos(2\pi f_c t)\right] \quad (1)$$

$$e_c(t) = A_c \cdot \cos(2\pi f_c t) \quad (2)$$

When these two signals are given to the balanced modulator, then output signal of balanced modulator is as follows

$$e_{out}(t) = K \cdot e_c(t) \times e_{Am}(t)$$

$$= K \cdot A_{DC} A_c^2 \left[1 + m_a \cdot \cos(2\pi f_m t)\right] \cos^2(2\pi f_c t)$$

$$= \frac{K \cdot A_{DC} A_c^2}{2} + \frac{K A_{DC} A_c^2}{2} \cdot m_a \cos(2\pi f_m t) + \frac{K A_{DC} A_c^2}{2}\left[1 + m_a \cos(2\pi f_m t)\right] \cos(4 \quad (3)$$

Where K represent gain of balanced modulator.
the first term is the DC signal. The second term is the audio signal and third term is the second harmonic of modulat Am signal. The third term is filter out by low pass filter.

and thus at the output there is only demodulated AM signal or audio signal. (or original modulating ~~AF~~ signal) ~~can~~

$$e_{out}(t) = \frac{K \cdot A_{DC} A_c^2}{2} \cdot m_a \cos(2\pi f_m t)$$

Generation of FM signal

There are two basic methods of generating the FM signal:
(i) Direct methods and (ii) Indirect methods

Methods of FM generation
- Direct methods
  - Reactance modulators
  - Varactor diode modulators
- Indirect methods
  - Armstrong method

The main advantage of direct FM generation is the simplicity of modulators and their low cost.

Disadvantages of Direct method:

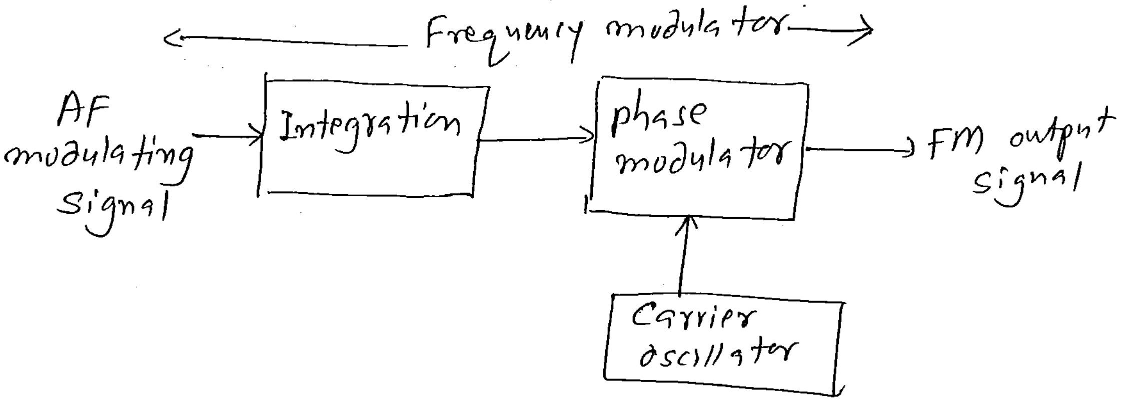
In the direct method of FM generation, we have to use LC oscillator. The LC oscillator frequency is not stable enough. Therefore it is not possible to use such oscillators for communication or broadcast purpose. Therefore we have to use indirect method of FM generation, in which we can use the crystal oscillator to control the carrier frequency.

Indirect method (Armstrong method) of FM generation:

In the direct methods of generation of FM signal, LC oscillators are used and crystal oscillators can not be used. But LC oscillators are not stable for communications or broadcast purpose (frequency stability problem). Thus direct methods generally not used for broadcast applications. The alternative method is to use the indirect method called as the Armstrong method of FM generation.

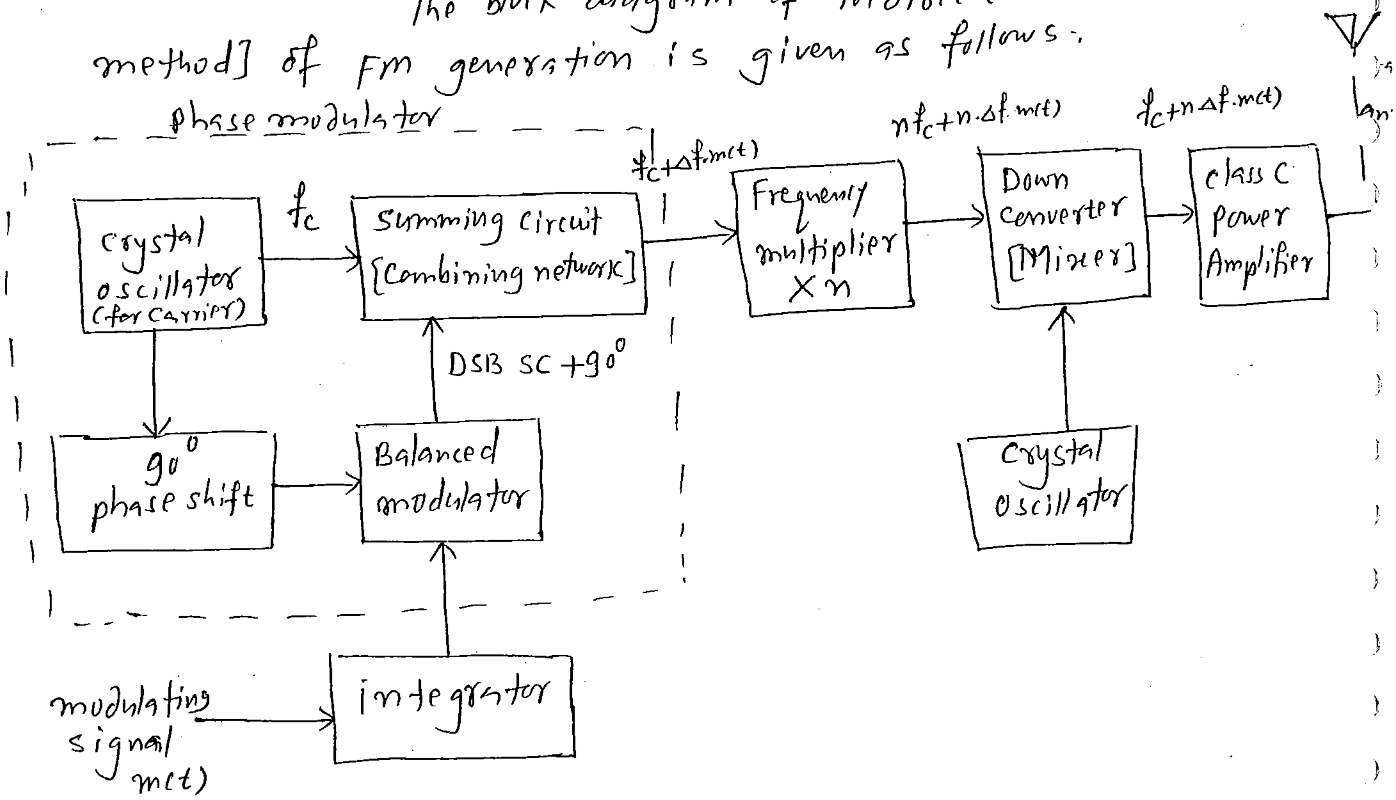
phase modulation is used to get frequency modulation by indirect method.

It is necessary to integrate the modulating signal, before applying it to the phase modulator and at the output of phase modulator, we get FM signal.



A crystal oscillator can be used, hence the frequency stability is very high. This method is widely used in practice.

The block diagram of indirect method (Armstrong method) of FM generation is given as follows.



The crystal oscillator produces a stable unmodulated carrier signal with frequency f_c. This carrier signal is applied to the Balanced modulator, after shifted by 90^0. The audio modulating signal is passed through an integrator circuit and then applied to the Balanced modulator.

At the output of balanced modulator, we get double side bands with ~~s~~ carrier suppressed (DSB-SC) signal i.e. [AM signal without carrier] with 90^0 phase shift.

The DSB-SC $+ 90^0$ signal from Balanced modulator and unmodulated carrier signal (f_c) are combined in a combination network.

The output of combining network gives FM signal $f_c + \Delta f . m(t)$

where f_c = carrier frequency.

Δf = frequency deviation = change in carrier frequency.

$m(t)$ = instantaneous modulating signal.

The FM signal produced at the output of phase modulator has a low carrier frequency and low frequency deviation. They are increased to a high value with the help of frequency multiplier ($\times n$). Thus output of frequency multiplier is $n . f_c + n . \Delta f . m(t)$.

The Down converter (mixer) along with other crystal oscillator used to increase frequency deviation without increasing the centre (original) carrier frequency ($f_c + n . \Delta f . m(t)$).

Finally the FM signal with carrier frequency f_c and high frequency deviation is given to class C power amplifier, to raise the power level of FM signal.

The power amplifier drives the Transmitting antenna.

Chapt 3: Transmitters and Receivers

FM Transmitters

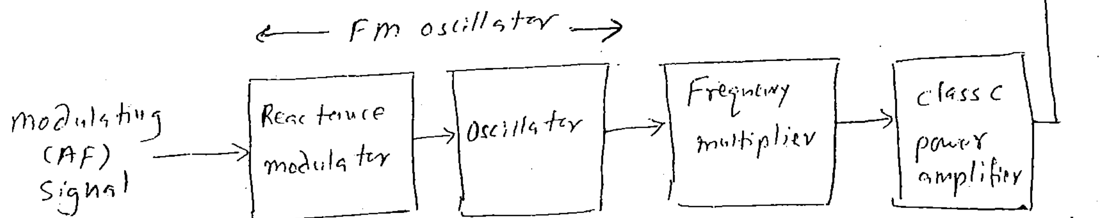
Explain with block diagram directly modulated FM Transmitter. (1)

(1) Directly Modulated FM Transmitter:

A block diagram of directly modulated FM Transmitter is ~~shown~~ given as follows:

The direct frequency modulation can be obtained by using varactor diode ~~Varactor diod~~ modulator or reactance modulator. The maximum frequency deviation is kept small and then modulated signal from FM oscillator is passed to a frequency multiplier circuit, which increases the output frequency of the desired carrier frequency and the desired frequency deviation. A power amplifier used to ~~increase~~ increase the power ~~leval~~ level of FM signal and it drives the Transmitting antenna.

What is the role of freq. multiplier in FM Transmitter? (1m)



Direct frequency modulation of carrier frequency with LC oscillator could be used, with ~~no~~ the multiplier stage omitted, but the problems of this methods are (1) to obtain sufficient frequency deviation (ii) and maintaining high frequency stability is ~~raises~~.

What are the problems of FM Transmitter without using frequency multiplier?

Crystal oscillators (for high freq. stability) can be directly frequency modulated, but to get ~~frequency~~ sufficient frequency deviation, a frequency multiplication factor (frequency multiplier) is necessary.

Why direct freq modulation is used only for narrow band Fm. ? (2m) (2)

Thus direct frequency modulation is only used for narrow band FM, having relatively small frequency deviations.

Effect of Heterodyne (mixer) and frequency multiplication on FM signal:

The heterodyne (mixer) and frequency multiplication are very commonly used processes in FM Transmitters.

Thus distinction between frequency multiplication and mixing is important in FM Transmitter systems.

Let us consider FM signal at the carrier frequency f_c with frequency deviation of $\pm \Delta f$. It is applied as one input of the mixer, while the other input is from local oscillator with frequency f_L.

Assume that local oscillator frequency f_L is less than the carrier frequency f_c. The output of mixer will be sum and difference components i.e ~~$(f_c \pm f_L)$~~ $\pm \Delta$

$(f_c \pm \Delta f) \pm f_L$ i.e.

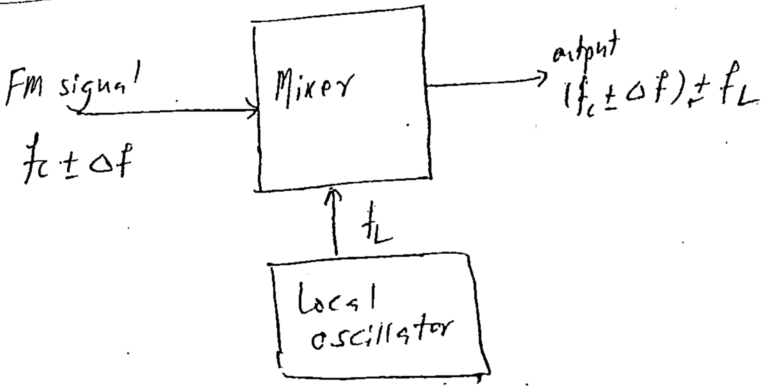
Sum Component: $(f_c + f_L) \pm \Delta f$

Difference Component: $(f_c - f_L) \pm \Delta f$

This shows that mixing process either increases or decreases the carrier frequency f_c, however keeping the frequency deviation constant. Thus heterodyne process has no effect on frequency deviation of Fm. (in FM Transmitter) Comment

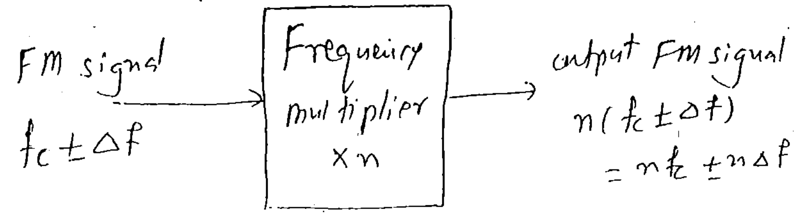
Explain role of

Comment



in FM Transmitter (2m)

(3)

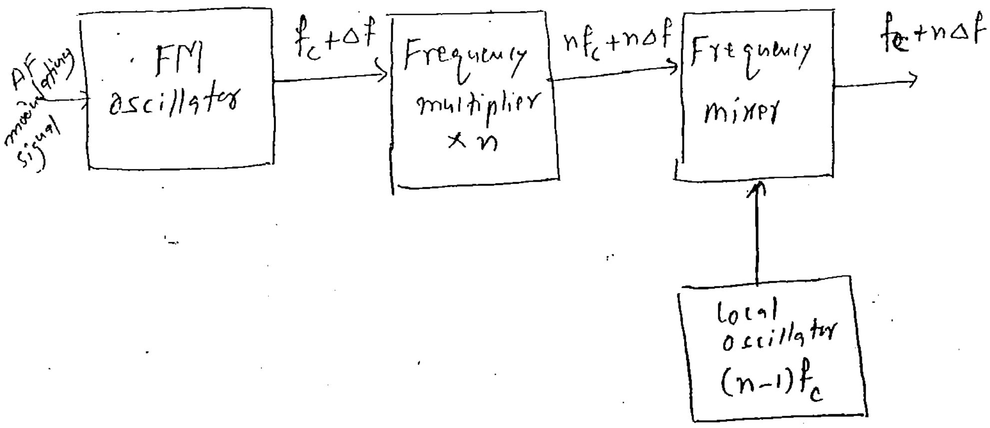


When the FM signal with carrier frequency f_c and frequency deviation $\pm \Delta f$ is applied at the input of a frequency multiplier (say $\times n$), then both the carrier frequency and frequency deviation are multiplied by multiplying factor (n) of frequency multiplier. In this case the output FM signal will be at center carrier frequency nf_c and frequency deviation of $\pm n\Delta f$.

Following figure shows how a combination of frequency multiplication and mixing can be used to increase freq. deviation without increasing the nominal FM oscillator frequency [center frequency] f_c

(1) Explain the role of frequency multiplier and mixer in case of FM Transmitter.

(2) Write the difference between frequency multiplier and mixer.



$$nf_c + n\Delta f - [(n-1)f_c]$$

$$= \cancel{nf_c} + n\Delta f - \cancel{nf_c} + f_c$$

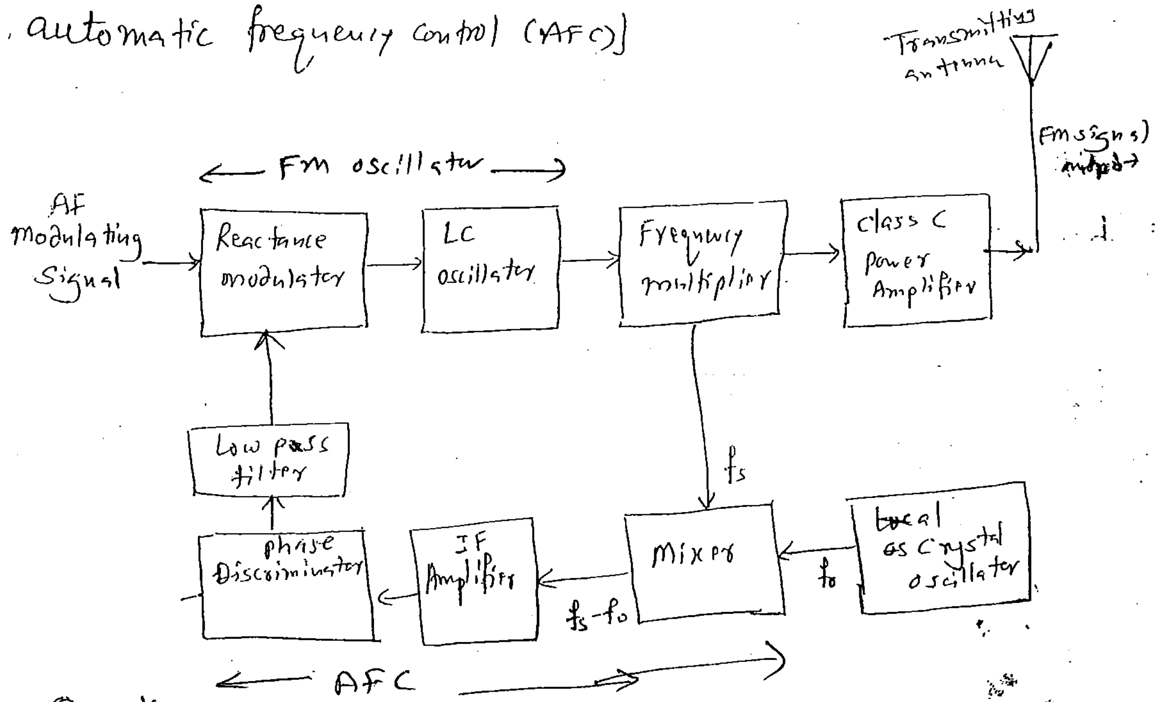
$$= f_c + n\Delta f$$

Explain with block diagram AFC, directly modulated FM transmitter. (4) (4M)

Frequency stabilized Reactance modulator (AFC) for directly modulated FM Transmitter —

The reactance modulator along with LC oscillator may be directly frequency modulated to produce relatively large ~~deviati~~ frequency deviation, but then it may be difficult to maintain stability of center freq. (unmodulated carrier frequency). But if it to be used in the broadcast applications, then it must have ~~a~~ high frequency stability.

A high frequency stability can be obtained by using the frequency stabilized reactance modulator [i.e. Frequency stability can be improved by using automatic frequency control (AFC)]



Operation:

(1) The reactance modulator gets the modulating signal as input. It operates on the tank circuit of an LC oscillator.

(2) The output of LC oscillator is the FM signal. It is passed through frequency multiplier and then applied to the class C amplifier, which drives the transmitting antenna.

What do you mean by indirectly modulated FM Transmitter?

The class C power amplifier is used to amplify FM signal. (5)

The fraction of FM output is taken from Frequency multiplier is fed to the mixer. The mixer also receives the signal from a crystal oscillator. (f_o)

The difference frequency signal ($f_s - f_o$) at the output of mixer is selected. It is usually $1/10^{th}$ of LC oscillator frequency. This signal is amplified using IF amplifier.

The amplified signal is then applied to a phase discriminator (phase demodulator). The output of phase discriminator is a dc voltage, applied to reactance modulator for correcting any drift in the frequency of LC oscillator. (due to temperature etc)

(6)

3.10 AM Transmitters :

➤➤➤ [University Exam - May 99, Dec. 2000 !!!]

The AM transmitters can be of two types :

(i) High level modulated transmitter or

(ii) Low level modulated transmitter.

Let us see their operation one by one.

3.10.1 Low Level Modulated AM Transmitter :

➤➤➤ [University Exam - Dec. 2006, May 2008 !!!]

- (1) The block diagram of a low level AM transmitter is as shown in Fig. 3.10.1(a).
- (2) The RF oscillator produces the carrier signal. The RF oscillator is stabilized in order to maintain the carrier frequency deviation within a prescribed limit. The carrier frequency is equal to the transmitter frequency. (carrier freq stability)
- 4 The amplified modulating signal is applied to the modulator along with the carrier signal (fc). At the output of the modulator we get the AM ~~wave~~ signal

3 The AF modulating signal is first passed through processing and filtering circuit and amplified by class A AF amplifier. Then this 3

Explain with the block diagram working of low level modulated/ High level modulated Am transmitter.

Compare Low level and High level modulated Am Transmitters

Depending on the way in which, the value of angle $\theta(t)$ of carrier signal changes, there are two types of angle modulations:
(i) phase modulation (pm) (ii) Frequency modulation (F.m)

phase modulation:

It is a type of angle modulation, in which phase angle $\theta(t)$ of carrier is changed in linear proportion with instantaneous amplitude of modulating signal or base band signal (message signal) $x(t)$.

Mathematically $\theta(t) = w_c t + \phi$

$$\theta(t) = w_c t + K_p \cdot x(t)$$

$$= \underbrace{2\pi f_c t}_{\text{carrier frequency}} + \underbrace{K_p}_{\text{phase sensitivity}} \cdot \underbrace{x(t)}_{\text{modulating signal}}$$

$w_c = 2\pi f_c t$ = angulare frequency of unmodulated carrier i.e. when $x(t)=0$

The second term $K_p \cdot x(t)$ represent the proportional phase angle change

The phase modulated signal mathematically expressed as

$$s(t) = E_c \cdot \cos\theta(t)$$

$$= \underbrace{E_c}_{\text{Amplitude Constant}} \cdot \cos[\underbrace{2\pi f_c t}_{\text{carrier freq. constant}} + \underbrace{K_p x(t)}_{\text{p varying phase angle with modulating signal } x(t)}] \quad \text{——— (A)}$$

This expression shows that P.M. signal has constant amplitude equal to that (E_c), of carrier signal, but phaseshift is the function of modulating signal $x(t)$.

Frequency Modulation (FM):

The frequency modulation (FM) is a type of angle modulation in which the frequency of carrier is varied in proportion with instantaneous amplitude of modulating signal $x(t)$.

Mathematically it is expressed as

$$\underbrace{f_i(t)}_{\text{change in frequency of carrier}} = \underbrace{f_c}_{\substack{\text{unmodulated carrier frequency} \\ \text{(centre carrier freq.)}}} + \underbrace{K_f}_{\text{frequency sensitivity}} \cdot \underbrace{x(t)}_{\text{modulating signal}}$$

Since frequency of FM signal $f_i(t)$ is changing continuously with time, we have to take integration of $x(t)$ over a duration of 0 to t, to write the expression of angle $\theta(t)$ for F.M.

$$\theta(t) = 2\pi f_c t + 2\pi K_f \int_0^t x(t)\, dt$$

Frequency modulated signal, mathematically, expressed as

$$s(t) = E_c \cdot \cos\theta(t)$$

$$= \underbrace{E_c}_{\text{constant amplitude}} \cdot \cos\left[\underbrace{2\pi f_c t}_{\text{unmodulated carrier}} + \underbrace{2\pi K_f \int_0^t x(t)\, dt}_{\text{continous change in modulating signal } x(t)}\right] \quad \text{—— (B)}$$

explain how frequency modulation is obtained by phase modulation.

Conclusion:

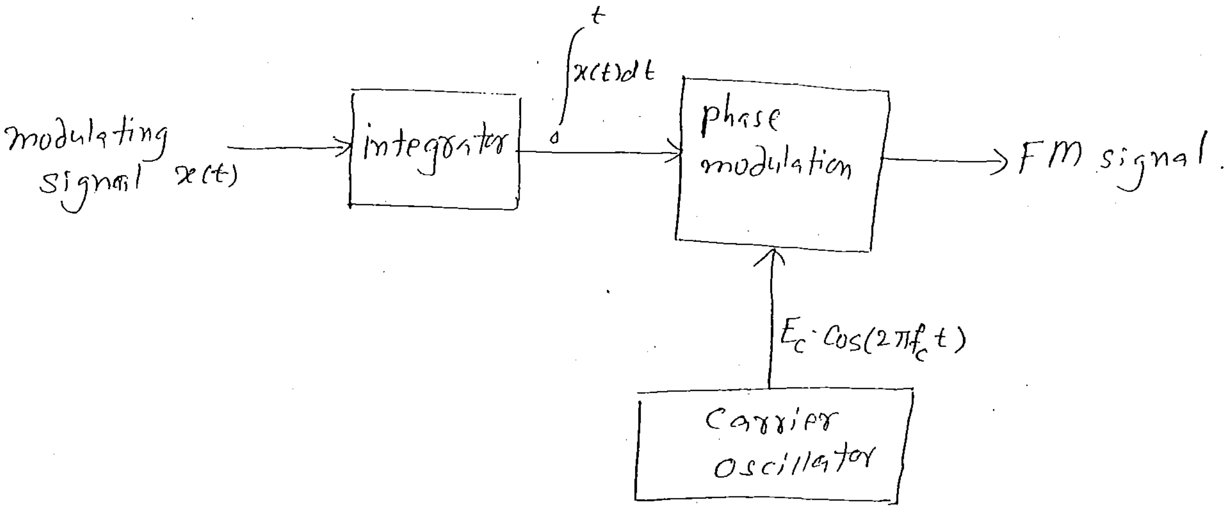
(1) In FM and PM, the angular argument $\theta(t)$ is allowed to become a functio of the baseband signal or message signal $x(t)$

(2) FM and PM are interrelated, one cannot change without the other changing. The information signal $x(t)$ frequency also deviates the carrier

carrier frequency in pm.

Generation of FM using phase modulator:

Comparing the expressions (A) and (B), we can conclude that FM signal is actually a pm signal having modulating signal is $\int_0^t x(t)\,dt$ instead of $x(t)$. This means that, we can generate FM signal by applying the integrated version of $x(t)$ to a phase modulator as shown in the following figure



Since the FM is produced by pm, ∴ such FM is referred as indirect FM.

FM achieved through phase modulation:-

[Indirectly modulated FM transmitter]:

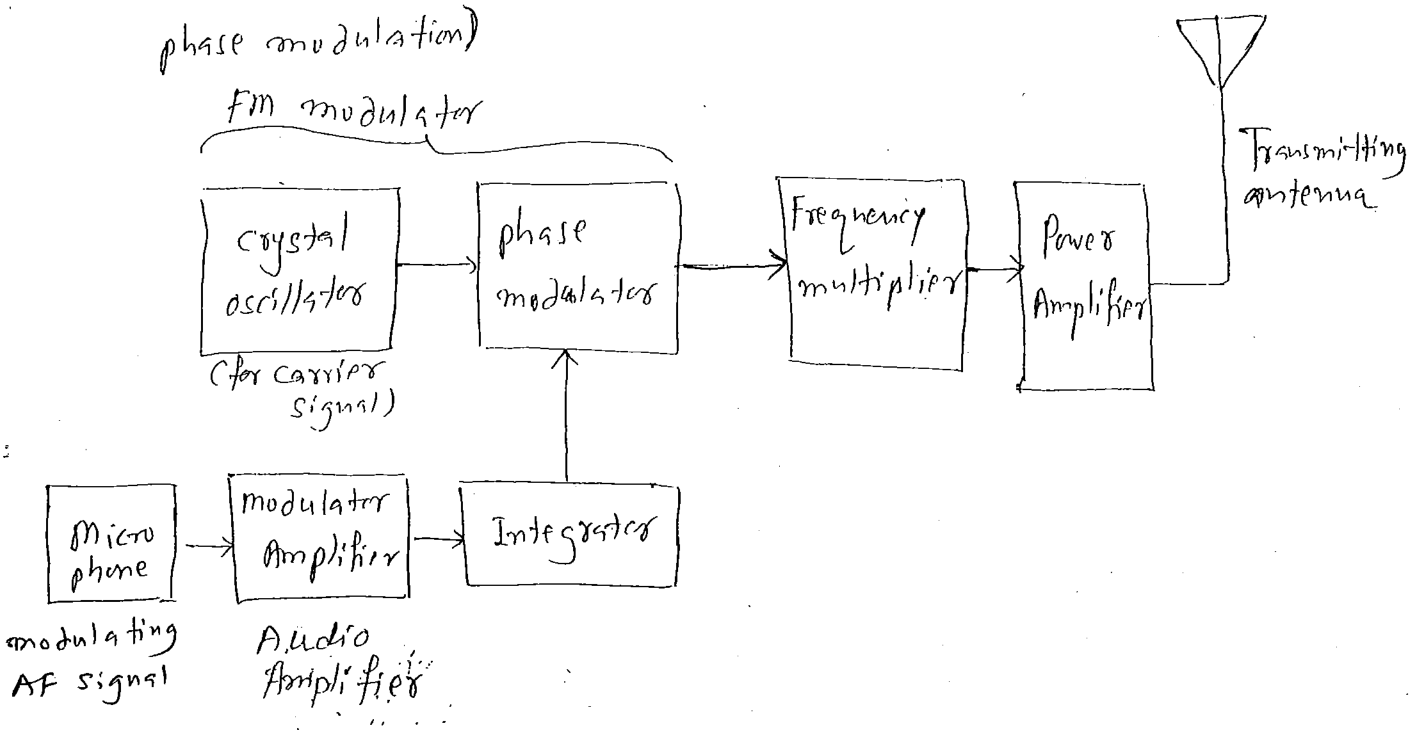
A very popular indirect method of achieving FM through phase modulation is Known as Armstrong method. This method is widely used in VHF and UHF radio telephone equipments.

phase modulation may be used to get frequency modulation by indirect method. It is necessary only to integrate the modulating signal before applying it to the phase modulator

Draw the block diagram of FM Transmitter where FM achieved through phase modulation and explain its working.

and at the output of phase modulator, we get FM signal.

Following figure shows block diagram of indirectly modulated FM transmitter (i.e. FM achieved through phase modulation)



The FM signal is then applied to frequency multiplier (multiplying by n) and then applied to power amplifier, which is used to amplify FM signal. Finally the power amplifier drives the transmitting antenna, which radiates the FM signal in electromagnetic form.

AM Transmitting

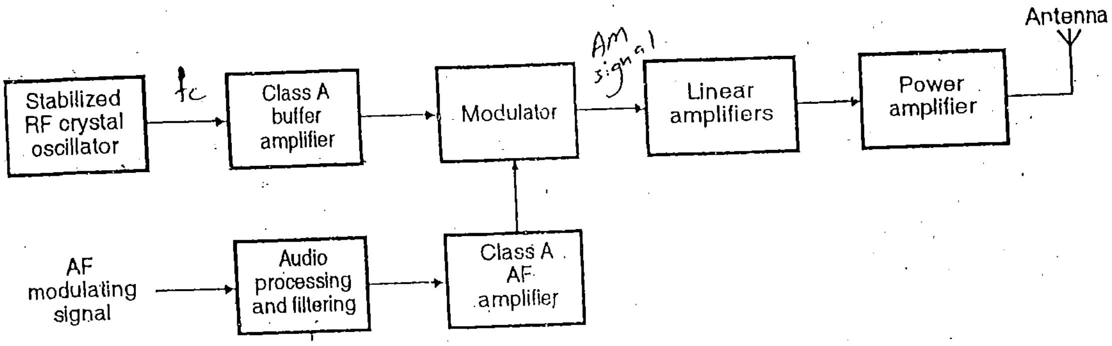


Fig. 3.10.1(a) : Low level AM transmitter

- This AM signal is then amplified using a chain of linear amplifiers to raise its power level.
- The linear amplifiers can be class A, AB or B type amplifiers. The linear amplifiers are used in order to avoid the waveform distortion in AM ~~signal~~. Then an AM signal is amplified given to power amplifier to raise the power level. (Class C power Amplifier)
- Finally The amplitude modulated signal is then transmitted using transmitting antenna.
- The transistorized modulator circuits can be used for low level modulator due to the low power which is to be handled.
- The low level transmitter does not require a large AF modulator power so its design is simplified.
- However the overall efficiency is much lower compared to high level modulation. This is due to the use of less efficient linear amplifiers.

3.10.2 High Level Modulated AM Transmitter :

➤➤➤ [University Exam - May 2005, May 2007 !!!]

- Many of the AM transmitters use the high level modulation technique. The block diagram of high level AM transmitter is as shown in Fig. 3.10.1(b).

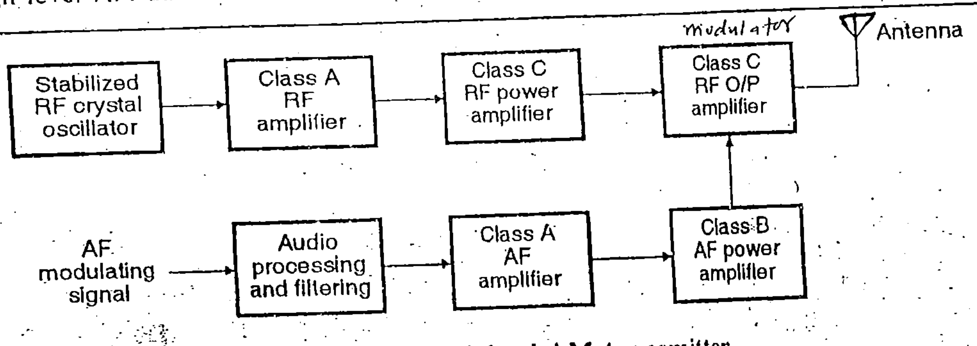


Fig. 3.10.1(b) : High level AM transmitter

Operation :

(i) Here the carrier generated by the stabilized crystal oscillator is first amplified to the adequate power level using class ~~A~~ RF power amplifiers. and class C RF power amplifier.

(8)

i) The modulating signal also is amplified to a high power level before modulation takes place. If we want 100 % modulation then the power of modulating signal must be 33 % of the total power. So if 1500 W total power is to be transmitted, the modulating power will be ~~500~~ 520 W. nearly 450

ii) The modulation takes place in the last class C RF amplifier. The modulator output is AM wave which can be ~~directly transmitted~~. ~~amplified by linear amplifiers and power amplifier and then transmitted through transmitting antenna;~~

v) The collector modulated transistorized circuit or plate modulated vacuum tube modulator is used as modulator stage.

i). The advantage of high level modulation is its high efficiency due to the use of highly efficient class C amplifiers.

i) The disadvantage is that a large AF power amplifier is needed to raise the modulating signal to the adequate power level.

.10.3 Comparison of High Level and Low Level Modulation : (Am Transmitters)

➤➤➤ [University Exam - Dec. 99, Dec. 2001, Dec. 2003, May 2005, Dec. 2006 !!!]

Sr. No.	Parameter	High level modulation	Low level modulation
1	Power level	Modulation takes place at high power level.	Modulation takes place at a low power level.
2	Types of amplifiers	Highly efficient class C amplifiers are used.	Linear amplifiers (A, AB or B) are used after modulation.
3	Efficiency	Very high.	Lower than high level modulators.
4	Devices used	Vacuum tubes or transistors for medium power applications.	Transistors, JFET, OP-AMPs.
5	Design of AF power amplifier	Complex due to very high power involved.	Easy as it is to be done at low power.
6	Applications	High power broadcast transmitters.	Sometimes used in TV transmitters (~~IF modulation~~). (Low power broadcast transmitters)

.10.4 AM Transmitter with LOW/HIGH Level Modulation :

Fig. 3.10.2 shows block diagram of an AM transmitter which may be either low level or high level modulation. It can be seen that there are a lot of common features. Both have a stable RF source, (crystal oscillator) buffer amplifiers and subsequent RF power amplifiers. In both modulation systems

In both types of transmitters the AF signal is processed, and filtered audio and audio power amplifiers are present. The only difference is the point at which the modulation takes place.

An amplifier is shown here following the modulated RF amplifier for low level modulation. This amplifier is a linear class B amplifier. (class C RF Amplifier)

No such amplifier is needed for the high level modulator. Hence the modulator output is directly connected to the transmitting antenna. (class C RF output amplifier)

For compari~~son of high and low level modulation techniques refer to section 3.10.3.~~

Draw the block diagram of AM Transmitter with Low and High level modulation and Explain its working.

3.

After pre-emphasis the modulating signal is applied to FM modulator. The other input for FM modulator is the carrier frequency signal from crystal oscillator. The carrier frequency stability is obtained using Automatic frequency control (AFC) circuit. Thus the FM signal from FM modulator is $f_c + \Delta f$. Since this FM signal has low carrier frequency and low frequency deviation (`$\Delta f$`) they are increased to high values with the help of frequency multiplier. i.e. $nf_c + n\Delta f$

The power amplifier is used to increase the power level of FM signal. Finally the power Amplifier drives the Transmitting antenna, which transmits FM signal in Electromagnetic form.

Q: Explain the working of FM Transmitter including Pre-emphasis Circuit.

What is the role of pre-emphasis circuit in case of FM Transmitter.

"Pre-emphasis Circuit is necessary in FM Transmitter" Comment.

"De emphasis Circuit is necessary in FM Transmitter". (2m) Comment

What is the purpose of Pre emphasis Circuit in FM Transmitter.

(10)

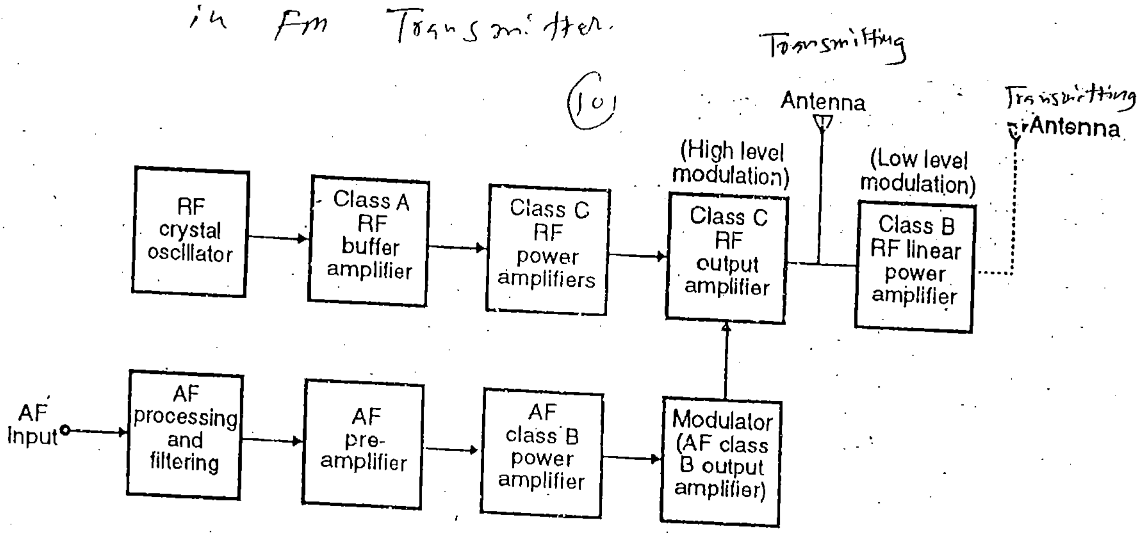
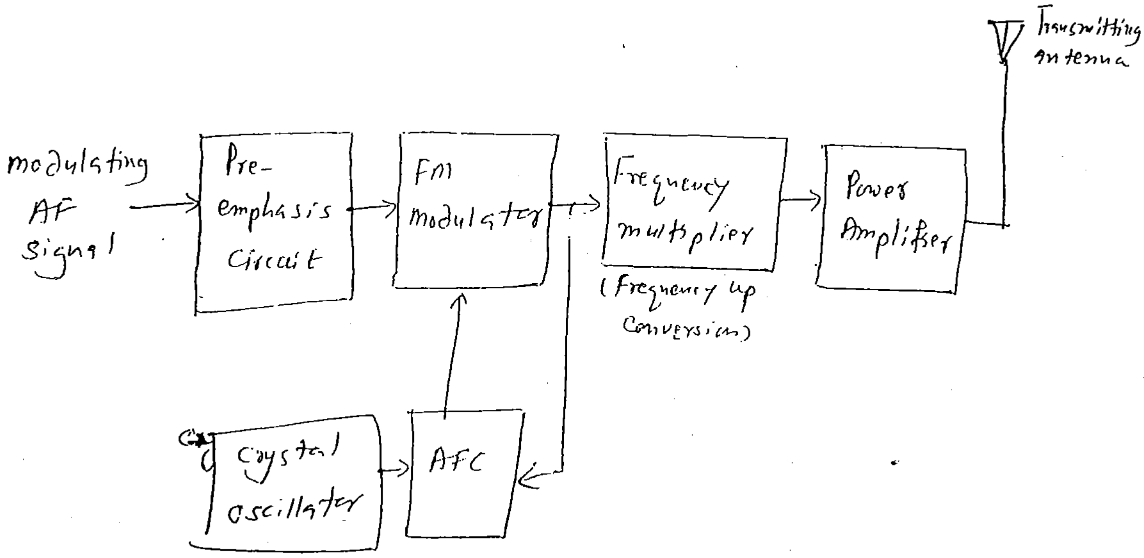


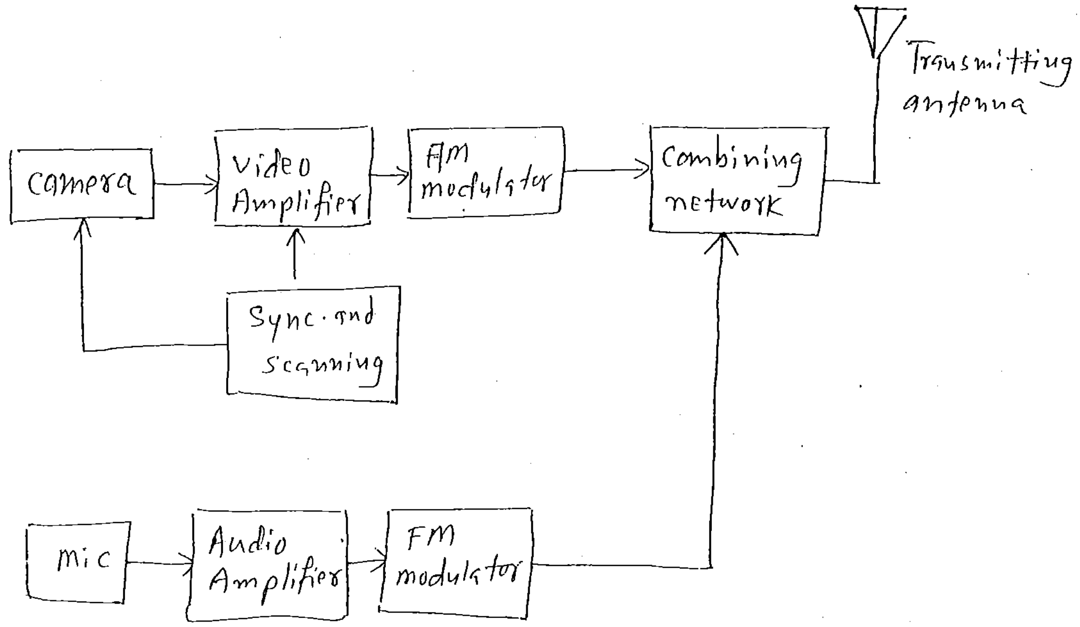
Fig. 3.10.2

(3) FM Transmitter including Pre-emphasis Circuit :—



The pre-emphasis circuit is used to boost the amplitude of higher frequency AF modulating signals, to improve the noise immunity at higher modulating frequencies. (to improve signal to Noise ratio).

[The boosted higher frequency AF modulating signals are brought to their original amplitude using de-emphasis Circuit at receiver]



Simplified block diagram of monochrome T.V.

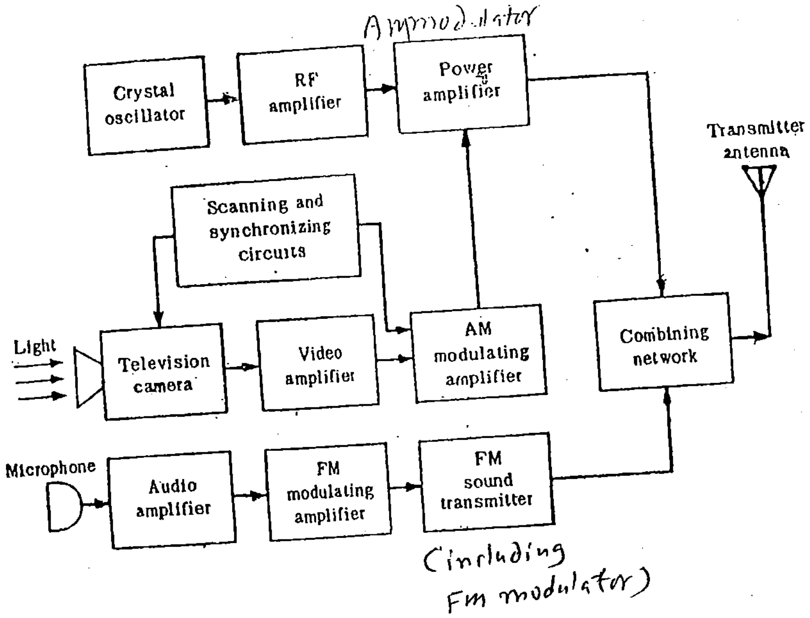
Why combining network is necessary in Television Transmitter (1m)

(11)

Television Transmitters

Simplified block diagram of monochrome television broadcasting system:

Explain the working of monochrome T.V. Transmitter using simplified block diagram.



The term broadcast means to send out in all directions. The television camera converts the visual information (optical information) into a corresponding electrical signal, whose amplitude varies in accordance with the variations of brightness. These electrical variations from the camera tube are known as video signal, which contains the desired picture information.

This video signal is amplified [using video Amplifier] and used as modulating signal for class c power amplifier [Am modulator]

The carrier signal for Am modulator is provided by crystal oscillator, it is amplified by RF amplifier and then given to Am modulator (class c power Amplifier)

The video signal is amplitude modulated, raised to the sufficient power and coupled to the transmitting antenna through combining network.

The scanning circuit is used to scan the scene rapidly in the horizontal and vertical directions simultaneously to provide sufficient number of complete pictures or frames per second

The synchronization circuit provides horizontal and vertical synchronization signals, to synchronize transmitter and receiver.

The synchronizing signals are transmitted along with picture information for receiver.

The sound waves incident on the microphone are converted into electrical audio signal. Then this audio signal is amplified and frequency modulated a high ~~carrier~~ frequency carrier. [separate carriers are used for the picture signal and sound signal]

The AM picture and FM sound signals are radiated by the same transmitting antenna using combining network.

One basis of modulation level, classify the Monochrome TV transmitter (13)
Compare low level T.V transmitter on modulation with high level modulation T.V. transmitter

Monochrome TV Transmitter:

On the basis of modulation level, monochrome TV transmitters are classified in two classes:

(a) TV transmitters with high level modulation.

(b) TV transmitters with low level modulation

In high level modulation, modulating signals (video and audio) and carrier signals are first amplified to high levels and then modulation is achieved. Class C modulated amplifiers are used to achieve these modulation and vacuum tubes are required to handle large powers.

In Low level modulation, modulating signal (audio, video) and carrier signal are first amplified to low level and then modulation is achieved. Semiconductor devices like BJT, FET, MOSFET can be used for low level modulation. The operation of semiconductor devices at low power is more linear and more reliable. Therefore Low Level modulation is widely used. High level modulation is hardly used now a days.

State the advantage of low level modulation technique in T.V. transmitter

TV transmitters are also classified according to the carrier frequencies used:

(a) TV transmitter with RF modulation and

(b) TV transmitter with IF modulation.

Block diagram of Monochrome TV Transmitter with RF modulation:

The TV camera converts the visual information (related to scene) in to electrical signal (video signal). Its amplitude varies according to variations of brightness of light present in the scene.

Thus video signal contains the picture information.

Along with this video signal, Horizontal deflection, vertical deflection, horizontal blanking, vertical blanking, horizontal synchronizing and vertical synchronizing signals added (as per CCIR standards) to make video signal as composite video signal.

The composite video signal is amplified by Camera Amplifier (Pre video Amplifier). The levels of composite video signal is set by using signal level controller, to make the amplitude of synchronizing pulses 100% and that of blanking pulses 75% of peak value of composite video signal.

The composite video signal is further amplified by another video amplifier. The picture related to this signal can be observed on TV monitor (M). Then make necessary changes, if required, by observing the quality of picture on the TV monitor.

This video signal and similarly the video signals from other cameras, may be from same studios or other studios or from out door cameras or from VCP or from cine film projector are feed to distributor and switcher circuit. Only one signal is selected at a time and given to transmitter through Coaxial Cable. It is again amplified by video amplifier. The DC clamping circuit adds DC level to this amplified video composite signal. This adjustment can be done by observing the picture on another TV monitor.

Highly stable crystal oscillator produces sinewave carrier frequency signal. ~~This~~ The frequency of this signal is multiplied by frequency multiplier and made equal to the picture carrier of the channel on which TV signals are to be transmitted. It is amplified by RF amplifier. It may be VHF or UHF amplifier depending on the channel used. These UHF/VHF picture carrier signal are modulated by composite video signals, in ~~RF~~ modulated RF amplifier. The negative amplitude modulation is used. Amplitude modulated VHF/UHF signals are further amplified by VHF/UHF power amplifier (i.e. class C tuned power Amplifier). Its output includes both side bands and carrier term. VSB (vestigial side band) filter is used to reduce bandwidth to 7MHz. This transmits complete USB and some part of LSB.

Microphone converts sound waves into AF signals. They are processed by audio processing unit and amplified by audio amplifier. Audio signals from various microphones placed in various studios, similarly audio signals from VCR, from outdoor mices, from cine film projector are added to distributor and switcher circuit. Audio signals from one source is selected at a time and transmitted. Selection of audio signals depends on picture selected. Finally selected audio signals are carried to transmitter through coaxial cable. These are further amplified by Audio amplifier. The audio signals are pre-emphasized. This technique improves S/N ratio in the receiver. The carrier frequency is frequency modulated by audio signals. As stable oscillator is not used in frequency modulation, automatic frequency control (AFC) circuit is used. Frequency of FM signals is multiplied by frequency multiplier. These signals are amplified by VHF/UHF class C tuned power amplifier. Ratio of video to audio (sound) power is 5:1. Modulated video and audio signals are fed to combining network. Generally duplexer is used as combining network. It combines video and audio powers and directs them to antenna. It enables us to use only one antenna to transmit video and audio signals. Combined signals are carried to transmitting antenna through coaxial cable.

Stacked turn style antenna is very commonly used for TV Transmission.
(turn stile antenna)
Antenna converts electricall power into electromagnetic waves and radiates them in the space almost equally in all directions.
Impedance of antenna must be equal to characteristics impedance of coaxial Cable.

Picture carrier and sound carrier frequencies are directly modulated by video and audio signals respectively Hence the method is called as RF modulation.

~~Ves.~~ Vestigial side band (VSB) filter:

The amplitude modulated composite video signal is fed to the VSB filter, which is designed to pass uppersite band ~~and~~ completely and some part of ~~ts~~ Lower side Band, resulting in saving bandwidth.

Write role of following in T.V. Transmitter.
i) Duplexe (2) VSB Filter

Draw the block diagram of TV transmitter and explain its working in detail. (8m)

(17)

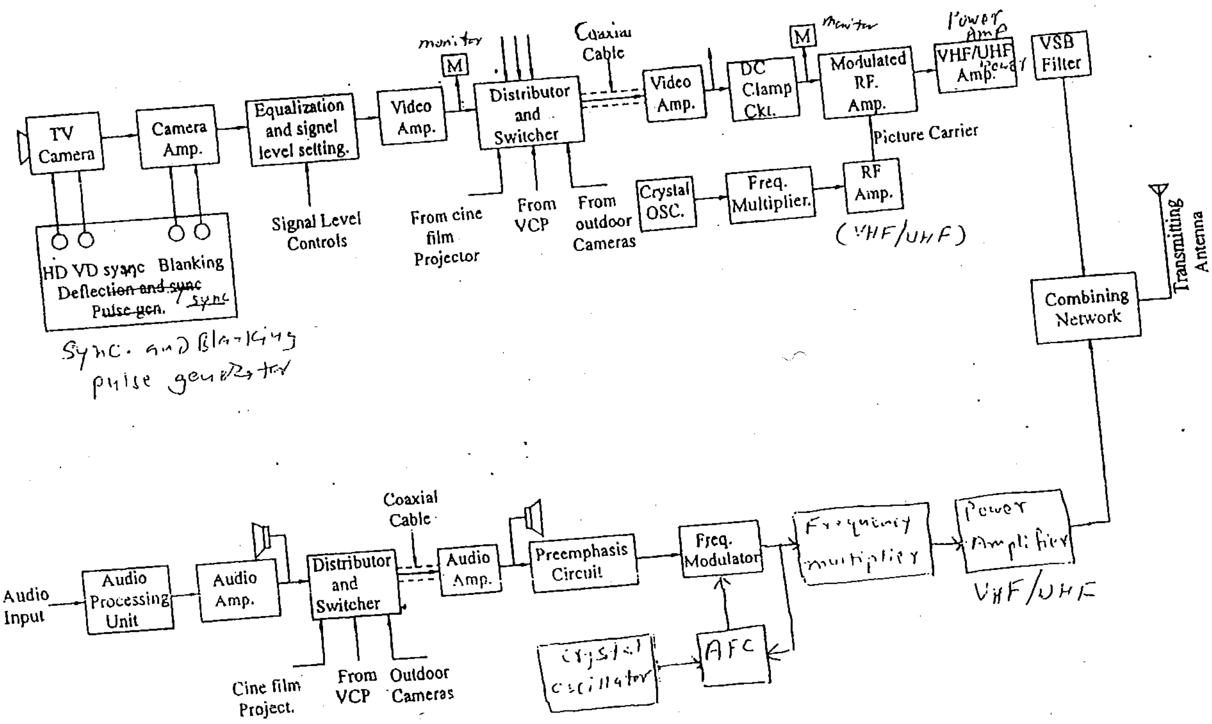
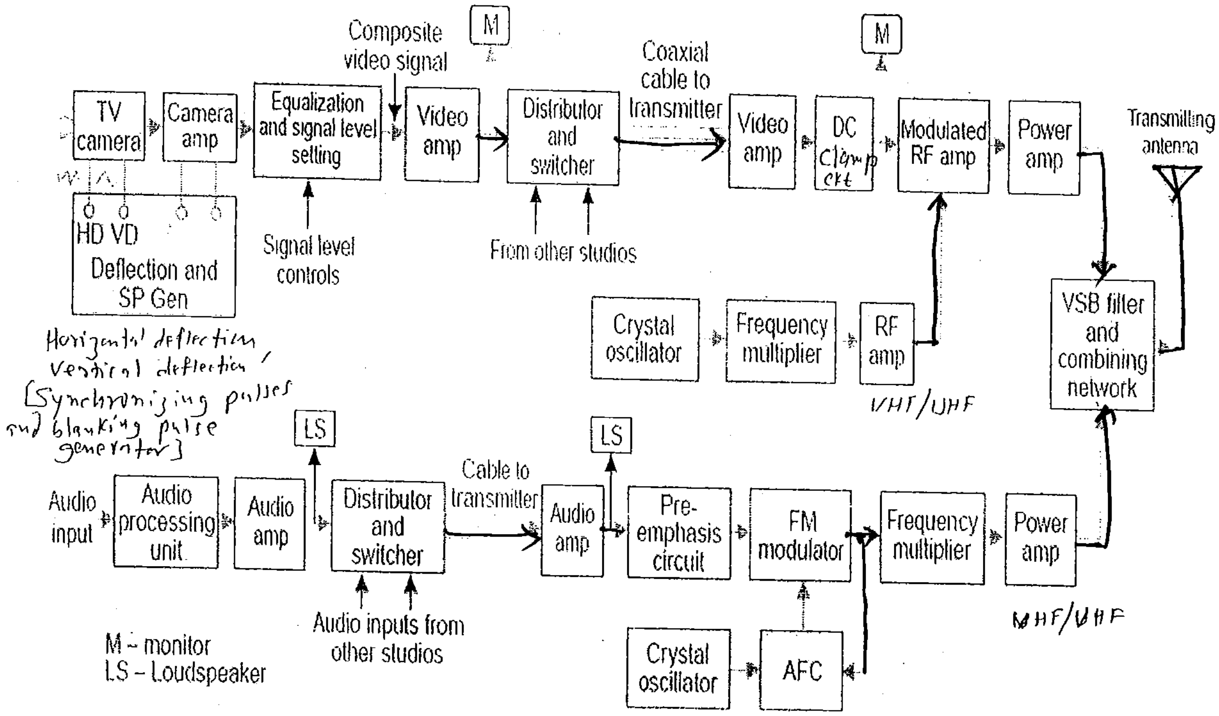


Fig. 1.16 : TV (Monochrome)

Block Diagram of a Television Transmitter



Colour TV transmitter:

The simplified block diagram of colour TV Transmitter is given in the following figure(a).

The colour camera provides video signal for Red, green and blue information. The three colour signals (R, G, and B) are applied as inputs to the colour combining matrix, to produce the luminance (Y) and two chrominance (I and Q) signals as follows.

colour

intensity

$$Y = 0.30R + 0.59G + 0.11B$$

$$I = 0.60R - 0.28G - 0.32B$$

$$Q = 0.21R - 0.52G + 0.31B$$

The Luminance signal Y can be used by black and white TV set to show Black and white picture. The I and Q signals carry colour information.

[In a colour TV Camera, the picture is split into three primary colours (using colour filters) and given to three diffrent Camera tubes. These three camera tubes produces Red, Green and Blue video signals.]

The Y, I and Q outputs from colour matrix (block) are fed to their respective low pass filters. These filters attenuate the unwanted frequencies.

There are three outputs of colour subcarrier oscillator [crystal oscillator]. One output is used to synchronize the blanking and sync pulse generators. The second output is used to colour burst generator, which ensures the correct transmission of colour burst. The last output from this oscillator is fed to a 57° phase shifter, to provide necessary shift for I signal. A further 90° phase shift is produced, giving a total phase shift of of 147° for Q signal. It is note that 90° phase difference between the I and Q signals.

The I and Q balanced modulators produces a double side band (suppressed carrier) signals (DSB-SC)

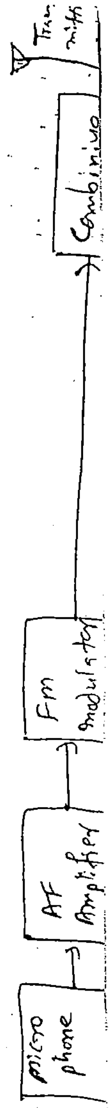
All these signals are fed to the adder, whose output i.e. ~~contains~~ composite video signal contains

(1) The Y luminance signal
(2) Synchronizing and blanking pulses.
(3) Colour burst signal
(4) I- Chroma signal
(5) Q- chroma signal

["The output of the adder is amplified and amplitude modulated]. Finally this Am signal is combined with the output of an Fm sound transmitter, whose carrier is 4.5 MHz above the picture carrier frequency

The transmitting antenna is a multiple dipole vertical arrays provide transmission in omnidirectional coverage (in all directions)

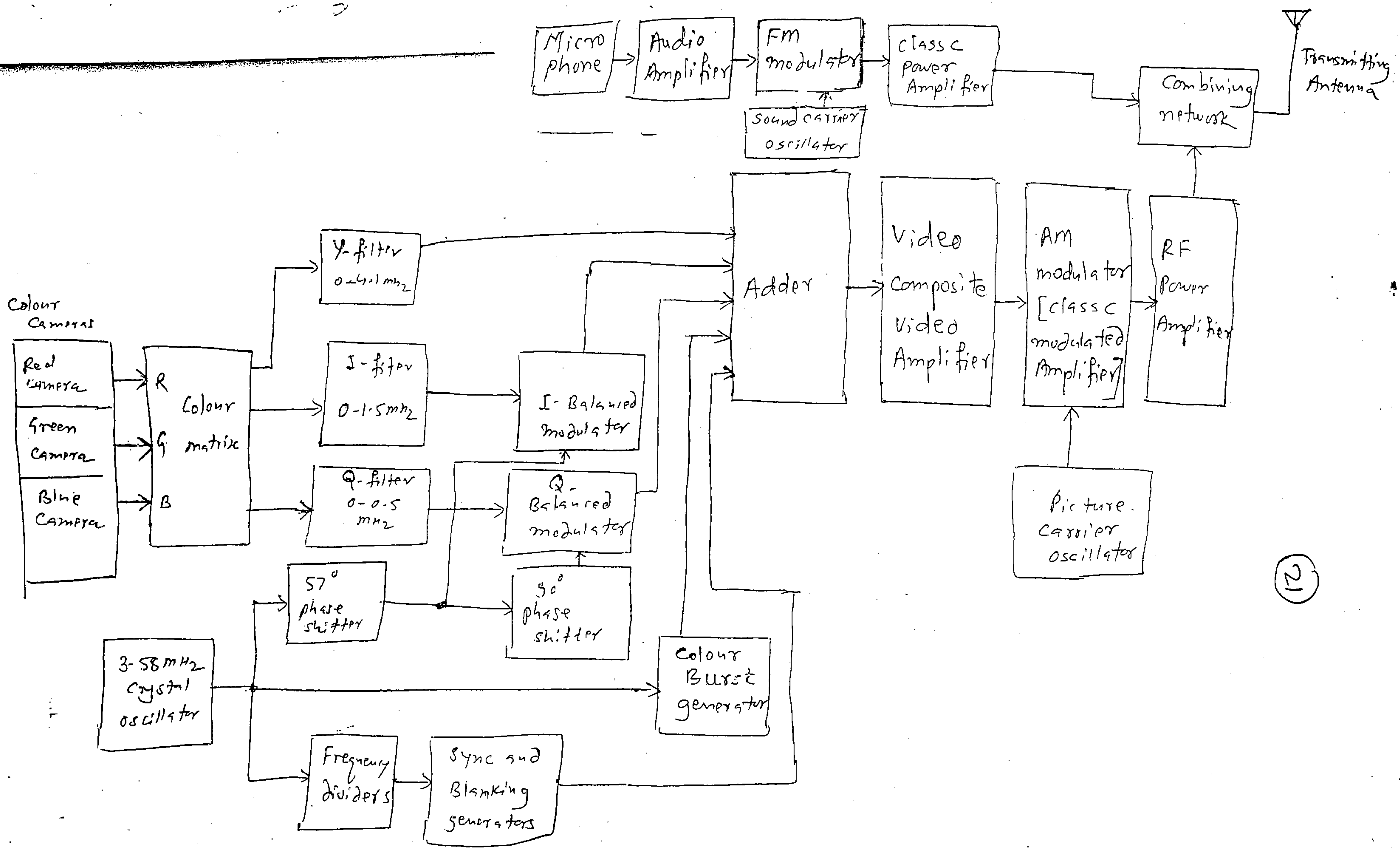
The output signal of adder i.e. composite video signal is amplified by video amplifier and then amplitude modulated in class C RF amplifier



Draw the block diagram of colour T.V. Transmitter and explain its working (6m)

what do you mean by luminance and chrominance signals?

(21)



Micro phone
Audio Amplifier
FM modulator
Sound carrier oscillator
class C power Amplifier
Combining network
Transmitting Antenna
Colour Cameras
Red camera
Green camera
Blue camera
R
G
B
Colour matrix
Y-filter 0-4.1 mhz
I-filter 0-1.5mhz
Q-filter 0-0.5 mhz
I-Balanced modulator
Q-Balanced modulator
57° phase shifter
90° phase shifter
Adder
Video Composite Video Amplifier
AM modulator [class C modulated Amplifier]
RF Power Amplifier
Picture carrier oscillator
3-58 MHz crystal oscillator
Colour Burst generator
Frequency dividers
Sync and Blanking generators

(22)

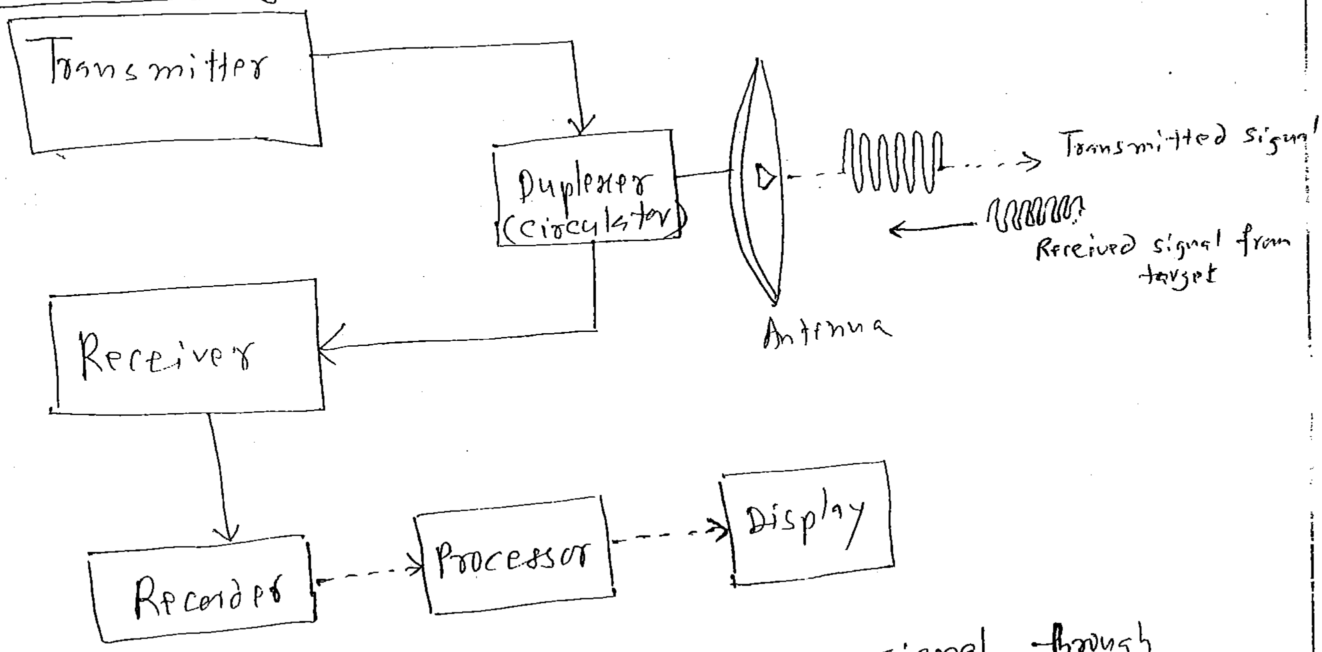
Radar :

The word Radar is stand for Radio Detection And Ranging. It is a system for detecting the Presence, position and speed of objects.

A simple Radar system has an antenna and a circuit or a computer which calculates and displays the position or speed of a target. The antenna transmit radio waves in a particular direction. The radio waves bounce off of (reflected by) objects in the path, and return to the antenna. The computer compares the reflected wave's characteristics with those of the transmitted wave to determine the target's position or speed.

Write the working of Basic Radar system.

Basic Radar system :—



The transmitter sends UHF or microwave signal through antenna. During transmission process, receiver is disconnected from antenna and transmitter to antenna by Duplexer. As soon as transmission of signal is finished, the duplexer disconnects the transmitter from antenna and connect antenna to the receiver and allow to receive reflected signal.

The receiving unit stores the received signal in memory, calculates the desired information from difference between transmitted and received signals. Now Radar system is able to show the position of or speed of target

A basic Radar system consists of

i) Transmitter
ii) Receiver
iii) switch or Circulator or Duplexer
iv) Antenna
v) processing unit and memory
vi) Display unit.

State various parts/components used in Radar system.

Write various Applications of Radar.

Applications of Radar:-

(1) Military Applications:
   (a) In missile system to guide the weapon
   (b) Identifying enemy locations in map
   (c) It is used for target detection.

(2) Air Traffic control:

The Radar has three major applications in Air Traffic control.

(a) To control air traffic near airports. The Air surveillance Radar is used to detect and display the aircraft's position in the airport terminals.

(b) To guide the aircraft to land in bad weather using precision Approach Radar.

(c) To scan the airport surface for aircraft and ground vehicle positions.

(3) Remote sensing:-

Radar can be used for observing weather or observing planetary positions and monitoring sea ice to ensure smooth route for ships.

(4) Ground Traffic control:-

Radar can also be used by traffic police to determine speed of the vehicle, controlling the movement of vehicles by giving warnings about presence of other vehicles or any other obstacles behind them.

(5) space:-

Radar has 3 major applications:

(a) To guide the space vehicle for safe landing on moon.

(b) To observe the planetary systems.

(c) To detect and track satellites.

State various types of Radar.

Types of Radar system:

(1) Continuous Wave (CW) Doppler Radar

(2) Pulsed Radar

Radar

- Continuous wave (CW) Doppler Radar
  - Simple Doppler CW Radar
  - CW Doppler Radar with IF amplifier
  - Frequency modulated (FM) cw doppler Radar
- pulsed Radar
  - pulse doppler Radar
  - Moving Target Indication (MTI) pulse doppler Radar.

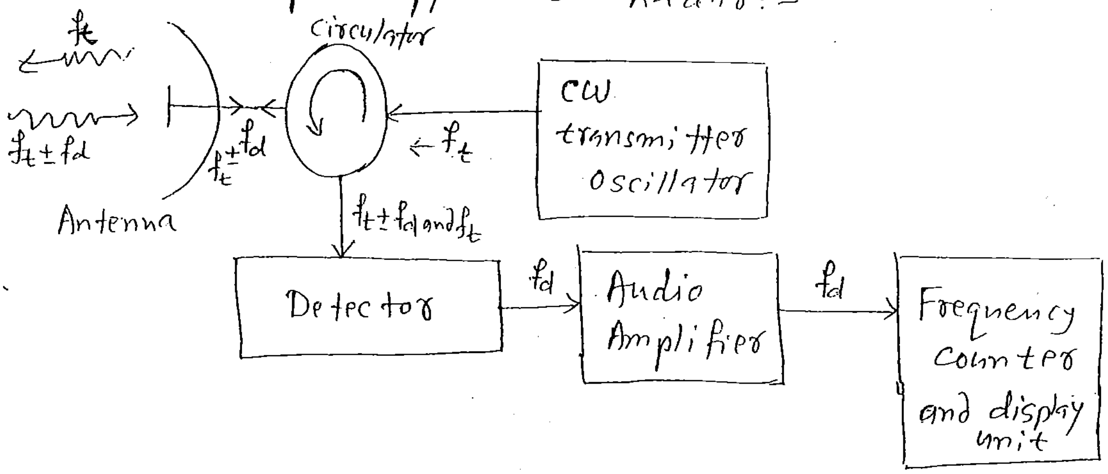
Doppler effect:

The Doppler effect (or Doppler shift) is the change in the frequency of wave for an observer, moving relative to its source. ~~There is~~ This change in frequency depends on the radial motion of source and observer.

If the source and observer are moving away from each other, then apparent frequency will decreases [i.e. Received frequency depends on the ~~distance~~ increasing or decreasing the distance between source and observer], while if they are moving towards each other the apparent frequency will increase. This Doppler effect is very useful in Radar systems.

CW Doppler Radar:- CW Doppler radar makes use of Doppler effect for target speed measurement. It transmit continuous sine wave signal rather than pulses.

(i) Simple Doppler CW Radar:-



In CW Radar, transmission is continous, hence duplexer is not required. Instead of duplexer, circulator is used to provide isolation between the transmission and receiver

The transmitted signal (f_t) is mixed with received signal (echo signal) from the target and the difference is the doppler frequency (f_d). This doppler frequency (f_d) is generally in the audio range, hence it is amplified by audio amplifier.

Then output of audio amplifier is applied to the frequency counter, whose output is displayed in terms of km/hr or miles/hr (speed), rather than actual frequency in Hertz.

The main disadvantage of this system is its low sensitivity.

Secondly diode detector not perfectly convert incoming high frequency f_t into audio output frequency, because noise is introduced. (*)

Draw the block diagram of simple CW Doppler Radar system, explain its working.

(2) CW Doppler Radar with IF amplifier:—

A small amount of the transmitter output (f_t) is mixed with the IF local oscillator signal (f_i) in Transmitter mixer, and the sum (f_t+f_i) is fed to the receiver mixer.

The receiver mixer also receives doppler shifted signal ($f_t \pm f_d$) from its antenna and produces an output difference frequency ($f_i \pm f_d$). This signal is amplified and then detected to get only doppler frequency (f_d), which is in audio range. It is amplified by audio Amplifier. (*) The sign of f_d is lost, so that so that it is not possible to tell whether the target is approaching or going away.

Extra sensitivity is provided due to too low noise of detector.

Separate antennas are used for transmission and reception for better isolation between transmitter and receiver.

(27)

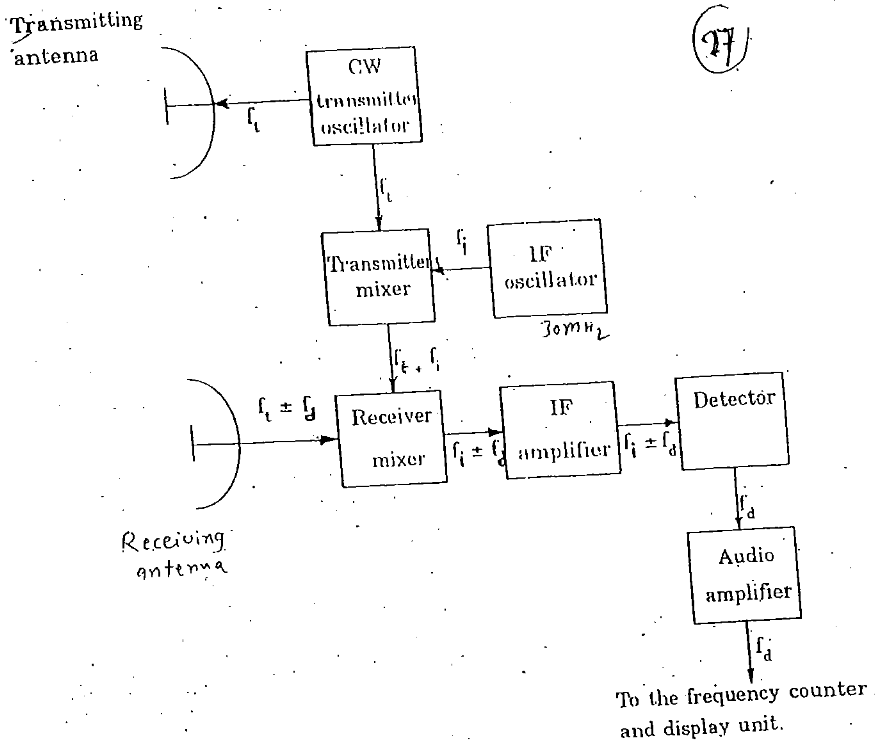


Fig. 1.17.: CW Doppler radar with IF amplifier.

separate antenna is that, there is no need for a small amount of transmitter output to leak into the receiver mixer as there was in simple doppler CW radar. To the contrary, such leakage is highly undesirable, becuase it brings with its hum & noise from the transmitter and thus degrade the receiver performance. The CW doppler radar transmits a power only upto 100 watt and is very low as compare to the pulsed radar.

Because of

not perfectly suitable to provide

Advantages of CW Doppler Radar:

i) CW radar gives accurate measurement of relative velocity of the target, using low transmitted power.

ii) Power consumption is very low, unlike a pulsed radar, because of low transmitted power.

iii) CW doppler radar receiver is working all the time and pulse loss does not occur.

iv) Simple circuitry and smaller size than that of pulsed radar system.

v) Stationary target can not affect results as it will not yield doppler frequency.

vi) A larger range of target velocities can be measured quickly and accurately

i) Write advantages of CW Doppler Radar.

ii) mention the various drawbacks of CW Doppler Radar.

iii) List the applications of CW Doppler Radar.

DISADVANTAGES OF CW DOPPLER RADAR:

i) It is limited in the maximum power, it transmits, and this places a limit on its maximum range.

ii) If a number of simulteneous targets are present, the doppler frequency will be incorrect.

iii) Doppler radar can not indicate the range of the target, it can show only its velocity, because the transmitted signal is unmodulated. The receiver can not sense which particular cycle of oscillations is being received at the moment, and therefore can not tell how long ago this particular cycle was transmitted, so that the range can not be measured.

iv) CW doppler radar can not distinguish the direction of the movement of the target.

APPLICATIONS OF CW DOPPLER RADAR:

i) CW doppler radar is used in air-craft navigation for speed measurement.

ii) It is used in rate-of-climb meter for vertical take off planes.

iii) It is used in radar speed meters used by police.

list the applications of Doppler radars.

(3) Frequency modulated (CW) dopper Radar.
(FMCW)

<u>Continous wave (CW) Radar:-</u>

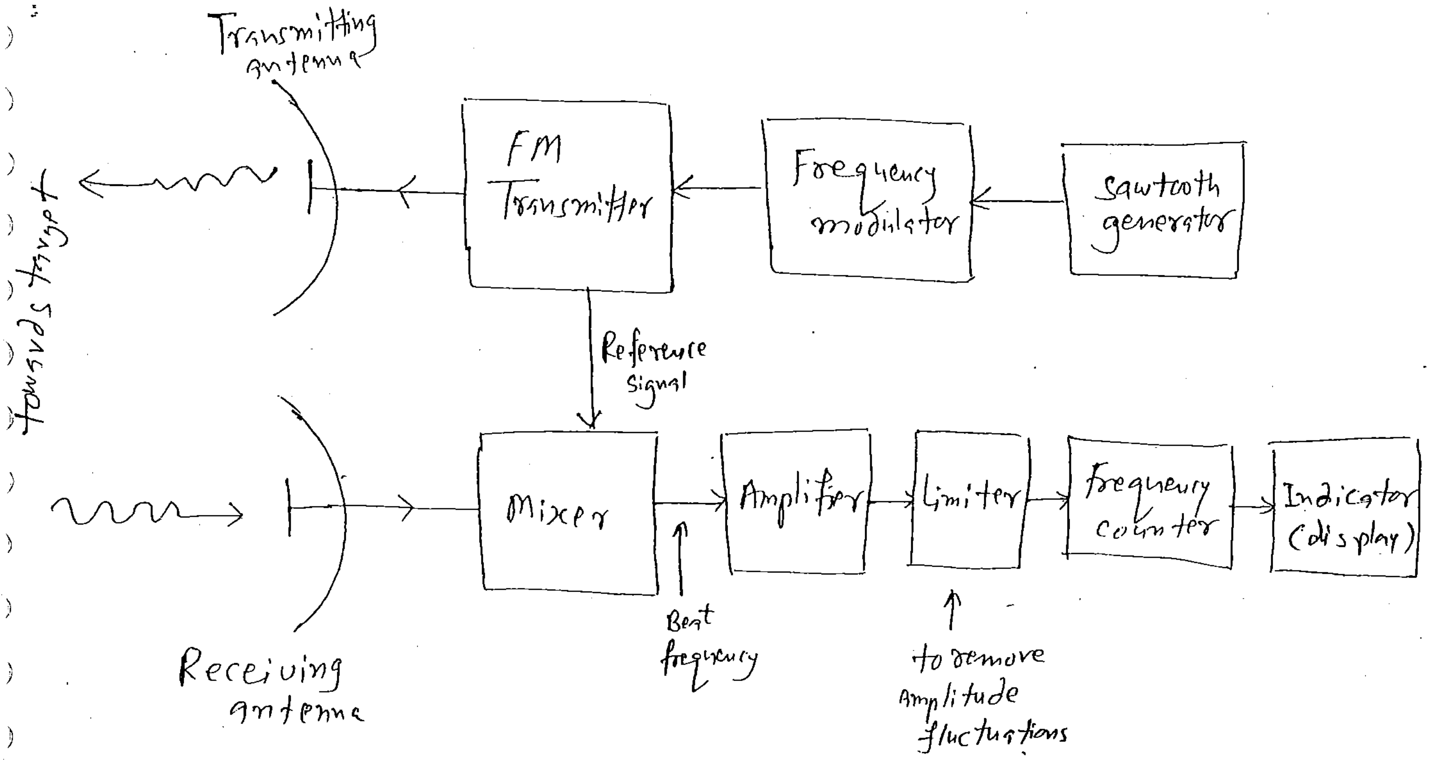
CW Radar transmit a high frequency sine wave signal continuously. The echo signal (Reflected signal) is received from target and processed, hence no range measurement. Difficult to avoid Tx ~~and~~ to Rx feed through, even with separate antennas.

Frequency modulated Continuous wave (FMCW) Radar.

The primary advantage of FMCW Radar is that, it can provide ranging data for targets relatively close to the antenna.

Features: (1) Ability to measure very small ranges to the target.
(2) Ability to measure simultaneously the target range and its relative velocity. (3) Ability to detect stationary and moving objects.
(3) Very high accuracy of range measurement.
(4) Safety from the absence of pulse radiation with high peak power.

Following figure shows block diagram of FMCW Radar



This Radar system transmitting a Continous carrier modulated signal (Freq-modulated carrier signal); The CW Carrier is modulated by a periodic signal such as sine or sawtooth.

Modulation is very important, since this add the ranging capability to FMCW Radar, compared to unmodulated CW Radar.

Explain working of FMCW Radar system with the help of block diagram. write the importance of Frequency modulation in case of FMCW Radar

In FMCM Radar system, measure the frequency difference between transmitted and received frequencies (Δf). This difference is in frequencies of directly proportional to the time delay Δt. and which is proportional to the distance of target.

The frequency modulated transmitting signal (as a refrence signal or local oscillator signal) and Received signal from receiving antenna are mixed (heterodyned) to create beat frequency, which is proportional to distance of targ.

The limiter is used to ~~remove~~ remove any change in the Amplitude of mixer output.

Frequency counter and indicator are used to display the distance in km or speed in km/hr.

CW Doppler Radar :-

In this, Radar system uses doppler frequency shift to detect moving targets. Unlike a pulse radar, a CW radar transmits while it receives (uses single antenna) i.e. simultaneous transmission and reception.

The transmitter generates a continuous (unmodulated) sinusoidal oscillator at frequency f_t, which is then radiated by the antenna. On reflection by a moving target, the transmitted signal is shifted by the doppler effect by an amount $\pm f_d$. The plus sign applies when the distance between radar and target is decreasing, thus the ~~at~~ echo signal from a closing ~~targl~~ target has a larger frequency than that of transmitted. The minus sign applies when the distance is increasing (target going away)

The detector or mixer takes the echo signal at frequency $f_t \pm f_d$ along with transmitter leakage signal f_t.

Simple Pulse doppler Radar :

(1) It is used to obtain both target velocity and target range data. (2) This radar can detect very fast moving objects at long distances, therefore it is used in many applications both civilian and military

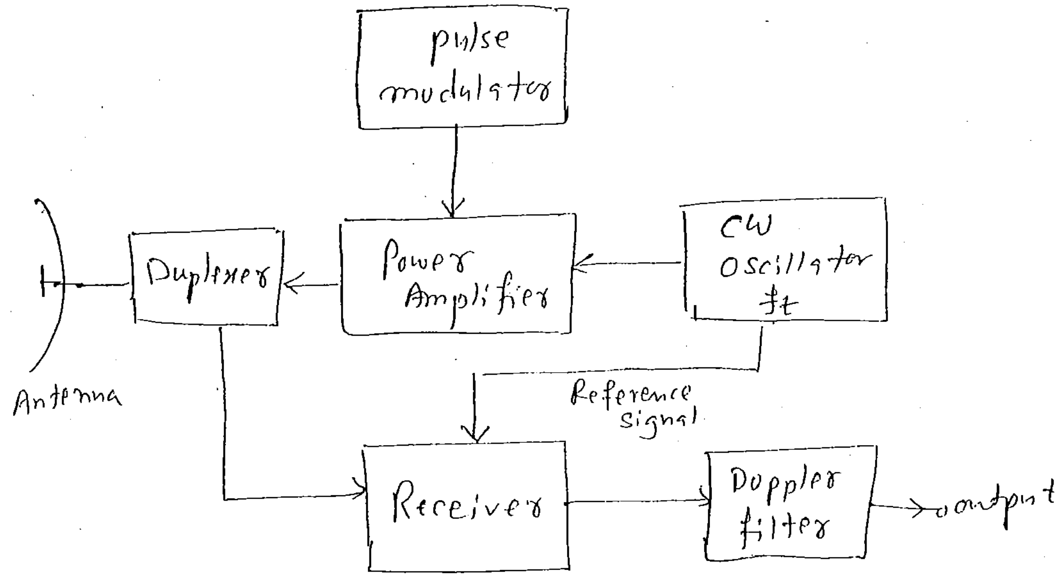
The pulse Doppler Radar transmits high power high frequency pulses towards the target

A ~~pt~~ pulse Doppler radar is a system that determines the range to a target using pulse timing techniques and uses the Doppler effect of returned signal to determine the target velocity

State the advantages of pulse Doppler Radar.

Draw the block diagram of simple pulse Doppler Radar and explain its working.

Following figure shows block diagram of simple pulse Doppler Radar:



The output of a stable CW oscillator is amplified by a high power Amplifier. The Amplifier is turned ON and off (modulated) by pulse modulator, to generate a series of high power pulses.

The received echo signal is mixed with the output of CW oscillator, which acts as a coherent reference to allow recognition (identification) of any change in the received echo signal frequency.

By coherent is meant that the phase of the ~~transmitted~~ pulse is preserved in the reference signal (i.e. ~~both~~ both have same phase)

The change in frequency is detected (recognized) by the doppler filter.

The Doppler frequency (frequency shift ~~on~~ by reflected signal from target) is given as

$$\text{Doppler frequency} = \left(\frac{2 \times \text{Transmit frequency} \times \text{Range velocity}}{c}\right)$$

(3) The pulse Doppler radars are very important for finding small moving targets hidden by heavily cluttered environment.

c = velocity of light

(34) write any one Radar to eliminate the clutter

Moving-Target Indication (MTI) (Pulse Doppler Radar):

It is possible to remove from the radar-display, the majority of clutter, that is; echoes corresponding to stationary targets, showing only the moving targets. This is always required. One of the methods of eliminating clutter is the use of MTI, which uses the Doppler effect in its operation.

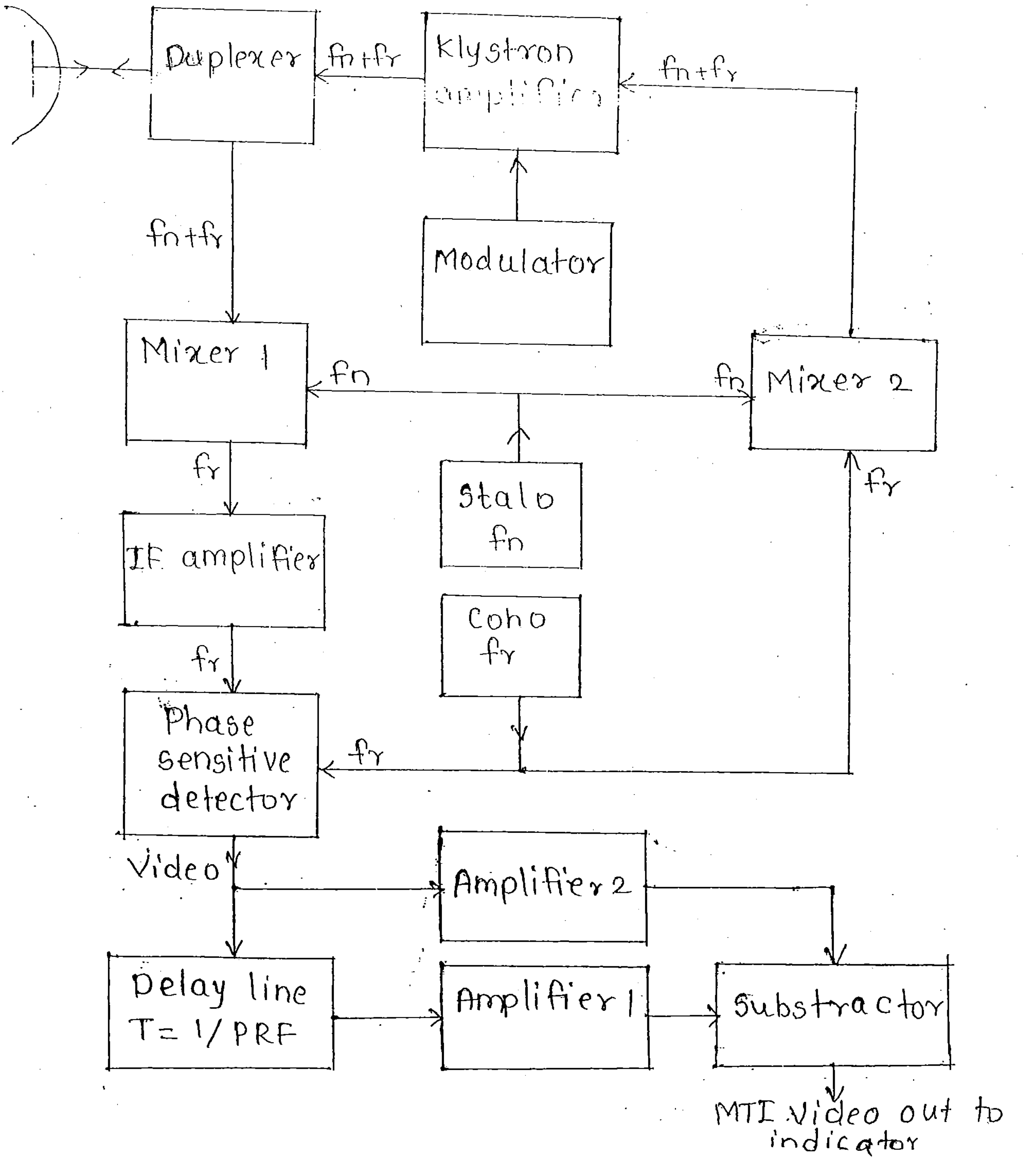
The purpose of MTI radar is to reject the signals from the fixed (stationary) unwanted targets (clutter) such as buildings, hills, trees, sea, rain etc and retain for detection or display the signals from moving targets such as aircrafts.

The MTI radar utilizes the doppler shift of the reflected signal by the moving target to differenciate moving targets from the fixed targets. Consider a radar which transmit a pulse of RF energy that is reflected by both, a building (fixed target) and an airoplane (moving target). The reflection from the building occurs exactly in same amount of time but the reflection from the moving aircraft occurs in less time. Because aircraft has moved closer to radar. [or ~~aircraft~~ reflection takes more time, when aircraft moved away from the Radar]

Block diagram of MTI Radar:-

Basically, the moving-target indicator system compares a set of received ~~tar~~ echoes. Those echoes whose phase has remained constant are then cancelled out. This is for echoes due to stationary objects, but those due to moving targets do show a phase change; they are thus not cancelled.

In this way clutter is removed and it makes it much easier to determine, which targets are moving and easy for operator to display ~~it~~ them. It also allows the detection of moving targets whose echoes are hundreds of times smaller than those of nearby stationary targets. MTI can be used with a radar using a power oscillator (magnetron) output. Following figure shows block diagram of MTI radar.



1) The transmitted frequency $f_n + f_r$ from the MTI radar system is the sume of the two output frequencies of two oscillators.
f_n = frequency of stable local oscillator (stalo).
f_r = frequency of coherant oscillator (coho).
The sum of these frequencies is performed by the mixer2.

2) The operating frequency of coherant oscillator f_r is equal to the intermediate frequency.

3) Mixer 1 and mixer 2 are identical and both have same local oscillator (stalo).

4) The reflected signal ($f_n + f_r$) throughly duplexer is mixed with the local oscillator frequency f_n in mixer 1. The output signal of the mixer 1 having frequency f_r is (intermediate frequency) which is amplified by the IF amplifier.

5) The coherant oscillator (coho) is used for two purposes –
a) For generation of the RF signals and
b) Reference signal for the phase sensitive detector.

6) The output of IF amplifier and the reference signal of coho are fed to the phase sensitive detector which is very similar to phase discriminator. Since the output of this detector is phase sensitive and the output is for all fixed and moving target. The phase difference between the transmitted and received signal will be constant for fixed target whereas it will vary for ~~co~~ moving target.

7) The output of phase sensitive detector i.e. video signals after amplification given to the

substractor circuit which gives the signal for moving target only.

8) And finally output of substractor is given to the display.

State an advantage of MTI Radar.

Draw the block diagram of MTI Radar and explain its working.

: frequency.

gun, the

cted radio

frequency

frequency

quickly a

If

used inside

must also be

example

and gun

20km/hour

r. If the

ct is not

car, then

Examples of Radar, which uses doppler effects are

(i) Pulsed Doppler Raddar — MTI Doppler Radar / pulsed Doppler Radar

(ii) simple Continuous wave (CW) Radar

(iii) Frequency modulated continuous wave (CFM-CW) Radar

(iv) CW Doppler radar with IF amplifier

(v) Speed Gun

Speed gun: The police uses this radar to measure vehicle speed. A speed gun radar uses the Doppler effect to determine the speed of a moving object. The concept of measuring vehicle speed with speed gun radar, is very simple.

A basic speed gun is a radio transmitter and radio receiver combined into one unit.

A radio transmitter gives electro-magnetic energy (~~RF radiation~~ Radio waves) with certain frequency into space, towards moving object (vehicle). The reflected Radio waves from moving object, have frequency which is different from orginal Radio waves.

Thus with the help of change in frequency the radar can calculate, how far away the object is and can also be used to measure the speed of an object. [due to Doppler shift].

When the radar gun and the vehicle are both have same speed, The echo (reflected radio wave signal) have same frequency as the original transmitted signal. But when the car is moving away from the radar gun, the wave length of reflected radio wave signal is increases hence decreases its frequency. (i.e. wave stretching)

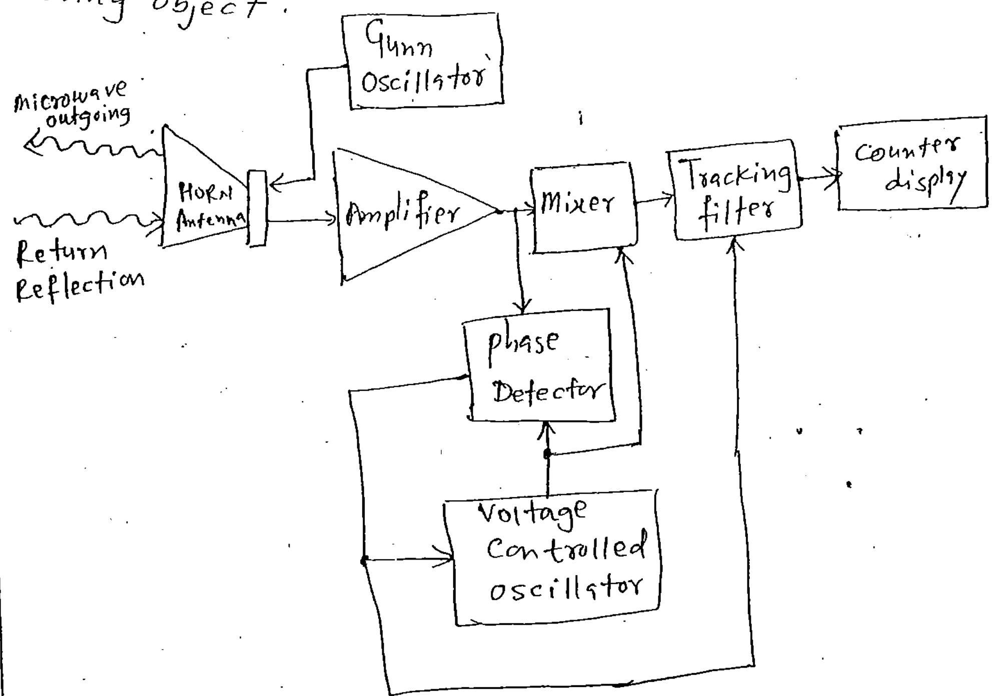
If the car is moving ~~towards~~ the radar gun, the wave get sequeezed. i.e. wave length of reflected radio wave signal is decreases, hence increases its frequency.

(pic. 8) Base on how much the frequency changes, a radar gun can calculate how quickly a car is moving towards it or away from it. ~~If~~

If the radar gun is used inside a moving police car, then its own speed must also be considered in calculation of speed of target car. For example if the police car is going 50 km per hour and gun detects that the target is moving away at 20km/hour i.e. target must be driving at 70 km/hour. If the radar gun determines that the ~~tget~~ target is not moving ~~toward~~ or away from the police car, then the target is driving at exactly 50 km/hour.

Recently police have used new speed detector, that uses light instead of radio waves. More recently police started using newer laser guns, but they soon proved too unreliable and had limitations and are influences by heat, headlights and other conditions.

Following figure shows the block diagram of radar speed gun. ~~is shown~~ A radar gun uses the Doppler effect to determine the speed of a moving object.



It's output consists of a steady, unmodulated carrier signal (microwave signal). This signal travels in a sharp beam towards a target whose speed is being ~~measured~~ monitored.

The target can be any object, such as a speeding baseball or a speeding motorist. The signal is reflected by the target back toward ~~their~~ radar gun.

Because of Doppler effect, the frequency of the reflected signal is different ~~from the~~ than the original transmitted signal. (If object is travelling towards the radar speed gun, the frequency of reflected signal is increases, while if object is moving away from radar speed gun, the frequency of reflected signal is decreases).

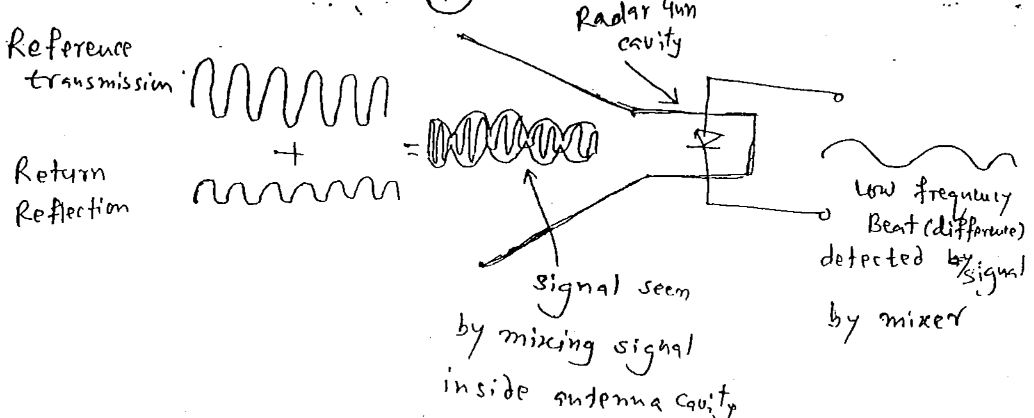
In the radar gun, the return signal (reflected signal) is mixed with a sample of the original transmitted signal to produce a difference frequency. This difference frequency is analysed and converted to produce a direct readout of speed.

The heart of a radar speed gun is a Gunn diode oscillator. This oscillator can generate microwave signal of about 100 mw for transmission. ~~With~~ This ~~signal~~ output signal ~~and along~~ is transmitted through a highly directional transmitting horn antenna. The range of radar speed gun is about 1 Km.

In the block diagram, mixer is used to provide difference frequency (using heterodyne action) The other parts of radar speed gun are phase detector, voltage controlled oscillator (VCO), tracking filter and final stage is counter display to display speed of object. The mixer have two inputs, (i) reflected signal (ii) From voltage controlled oscillator.

The tracking filter gives the signal for best frequency (doppler freq.)

(42)



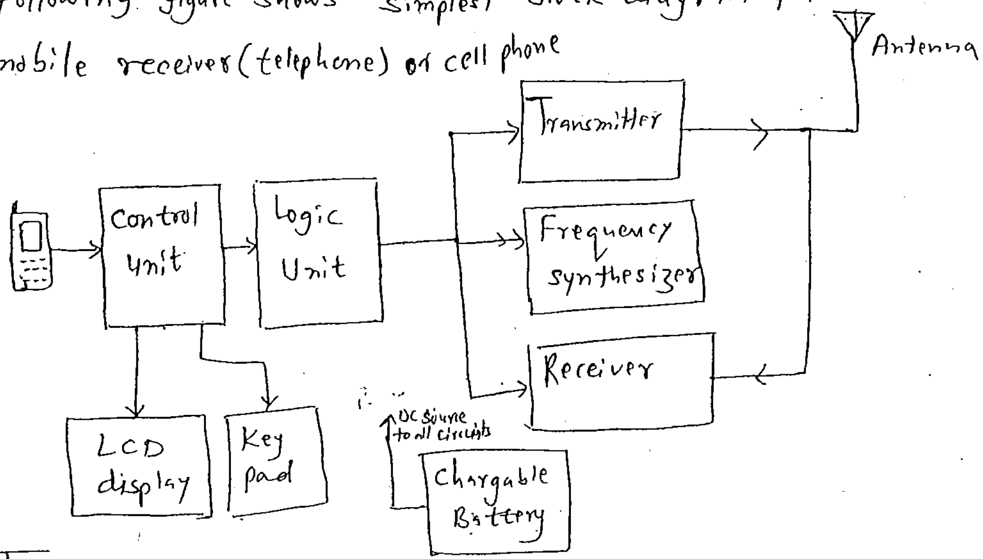
When the transmitted and Reflected signals are mixed in the antenna's horn cavity, the resulting low frequency beat can be used to determine the speed of the reflecting object.

Explain the working of speed Radar gun with suitable block diagram.

(43)

Cellular mobile telephone (receiver) : [cell phone]

Following figure shows simplest block diagram of a cellular mobile receiver (telephone) or cell phone



The main sections of mobile phone are

(1) Antenna
(2) Transmitter
(3) Receiver
(4) Frequency synthesizer
(5) Control Unit and logic unit
(6) Key pad and LCD display
(7) Rechargeable Battery

Explain working of cellular mobile phone with the help of simplest block diagram (4m)

State main blocks of cell phone (2m)

The detailed block diagram of mobile phone is given as follows:

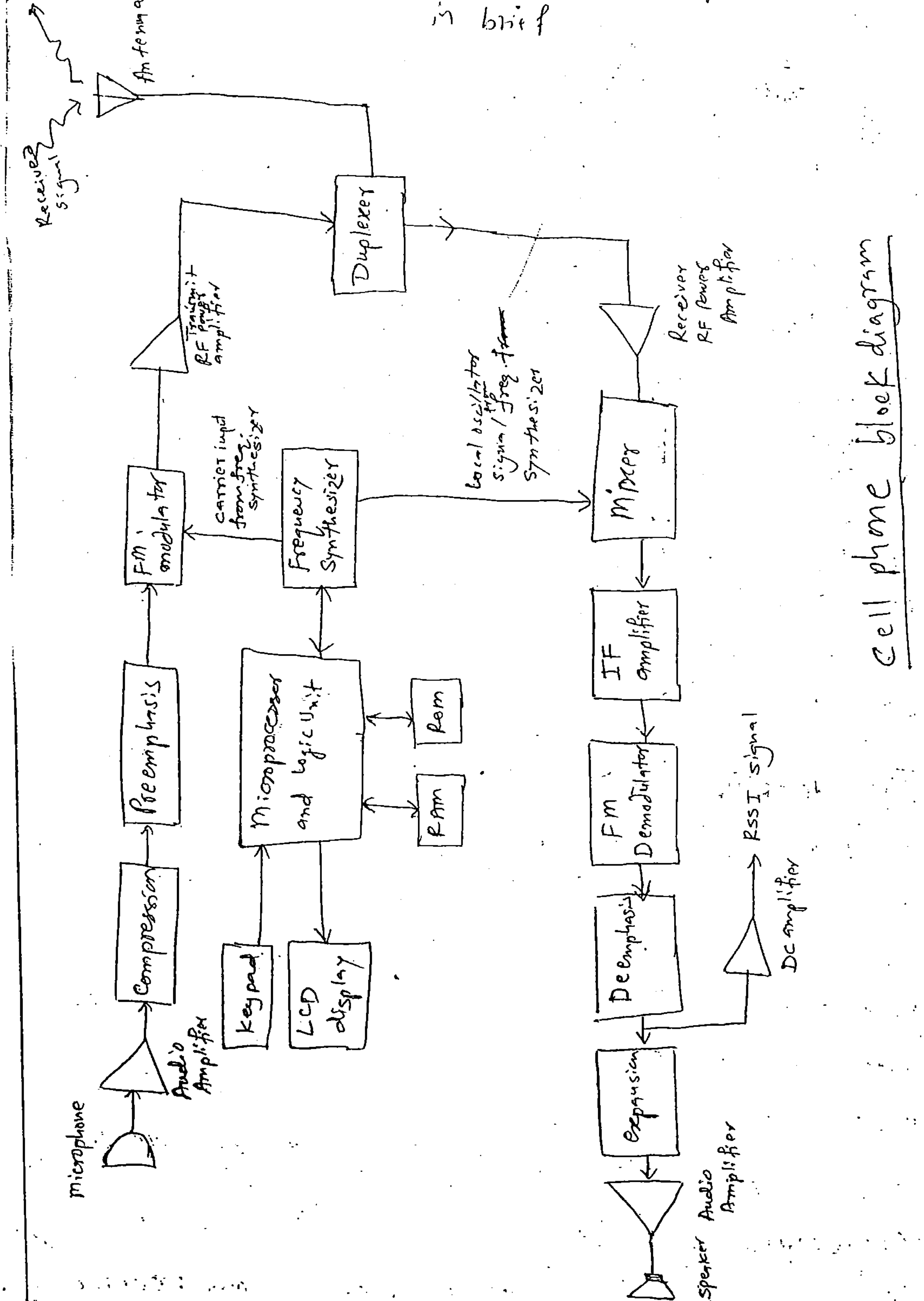
Since the mobile phone is a full duplex transceiver, the transmitter and Receiver are operated simultaneously with a single antenna. The mobile phone antenna is a quarter wave monopole antenna and half wave dipole antenna.

A duplexer is used to separate the transmitting and Receiving signals. The frequency separation of 45 MHz between transmit and receive

Write the role of Duplexer in mobile telephone (1m) frequencies is made by frequency synthesizer to minimise the interference.

Draw the block diagram of mobile phone (receiver) and explain its working in brief.

Mobile phone and explain itsw in brief



Cell phone block diagram

The switching of channels and power levels by the remote control base station are controlled through microprocessor and logic Circuit. Also due to microprocessor and digital circuits, (Such as memory) additional functions with handset are available to the user. These functions are generally timings of calls, storing messages, locking a phone, storing list of called numbers, remainders etc.

The transmitter in a cell phone handset is a low power FM transmitter operating in the frequency range of 825 MHz to 845 MHz. There are 666 transmitt channels spaced 30 KHz apart. The carrier signal is provided by a frequency synthesizer, which is frequency modulated by the voice signal. The Pre-emphasis is used to improve Signal to noise (S/N) ratio. The data (voice) is compressed before modulation.

Finally the FM signal is amplified by class C RF power amplifier and then transmitted by antenna, through Duplexer.

Duplexer Circuit allows the transmitter and Receiver to share the same antenna.

The FM signal is transmitted ~~towards~~ the cell station (or cell site).

The FM signal is received by receiver section of mobile phone from cell station. The receiver frequency range is 870.03 MHz to 889.98 MHz. There are 666 receive channels, spaced by 30 KHz bandwidth.

The mixer used to convert incoming FM signal to IF signal of 82.2 MHz. for mixer the local oscillator signal is obtained by frequency synthesizer.

After Demodulation and de-emphasis the signal is expanded to the original form and signal is given to speaker

The logic control unit includes a microprocessor (CPU) with RAM and ROM memories. This unit interprets signals from MTSO (Mobile Telephone switching office) and cell site. It also generate control signals for the Transmitter and Receiver. [Additional functions of this logic control unit are]

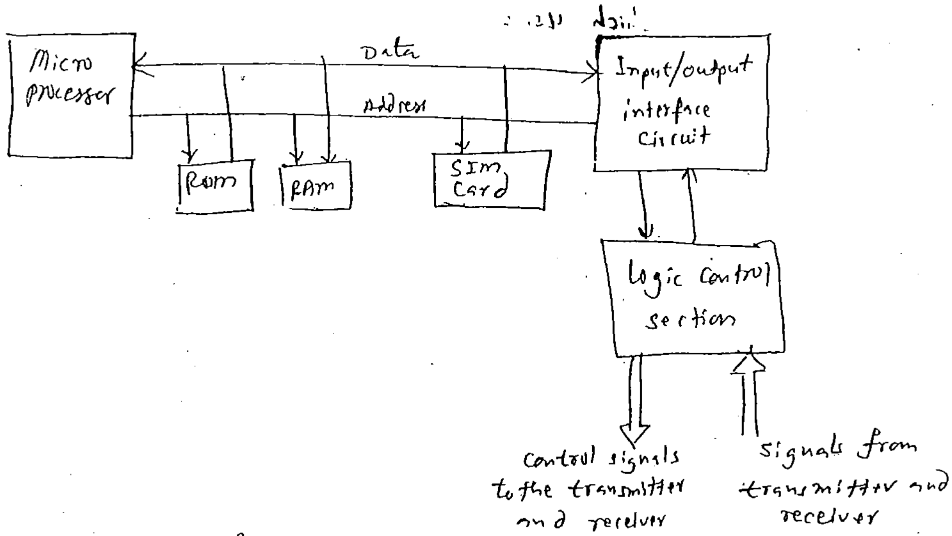
A Keypad is used along microprocessor to enter phone numbers and messages.

A LCD display is used to display phone numbers and messages.

The demodulated output is also rectified into dc signal whose amplitude is proportional to the strength of received signal. This is called the received signal strength indicator (RSSI) signal, which is sent back to cell site, so that the MTSO can monitor the received signal from the cell and make decision about switching to another cell.

Following figure shows the block diagram of logic control circuit in a mobile handset.

3



(i) This section includes a microprocessor with both RAM and ROM memories. This section interpretes signals from MTSO and cell site. It also generates control signals for the transmitter and receiver.

The SIM (Single Inline memory module) card is also included in this section. It allows the mobile telephone to identify itself when a call is initiated or when demanded by the MTSO.

GSM subscribers are provided with (Global system for mobile Telecommunication) SIM card with its unique identification at the very start of the service.

The SIM card contains an integrated circuit chip with a microprocessor, RAM and ROM. The SIM card contains all subscriber specific data.

Low Noise Amplifier (LNA):

Features:

(i) Extremely Low Noise and High Gain.

(ii) Wide dynamic frequency Range (operating freq. range 500KHz to 50GHz)

(iii) Wide operational voltage range (voltage ranged from 2V to 10V)

(iv) Highly Reliable (v) operating temperature from $-30c^{0}$ to $+50c^{0}$

Applications:

(i) Ultra Low noise applications.

(ii) LNA for cellular Base stations.

(iii) High linearity applications.

(iv) High performance Receivers

(v) millitary Radio

(vi) RF-Pre-amplification

(vi) Low level signal amplification

(vii) Test and measurement instruments

(viii) RF repeater

ix) Used in Transponders of satellite.

cellular handsets.

GPS Receivers

ISM Radius

Cordless phones

satellite communication

~~Write~~ wireless LANs

(i) Write the features of LNA. (2m)

(ii) Give the applications of LNA. (2m)

The Low Noise Amplifier is used as the front end amplifier of Receiver so that the receiver sensitivity is improved by low noise figure and high gain in the first stage.

Low Noise Amplifier (LNA) is an electronic amplifier used to amplify very weak signals (for example signal from Antenna), having lot of noise.

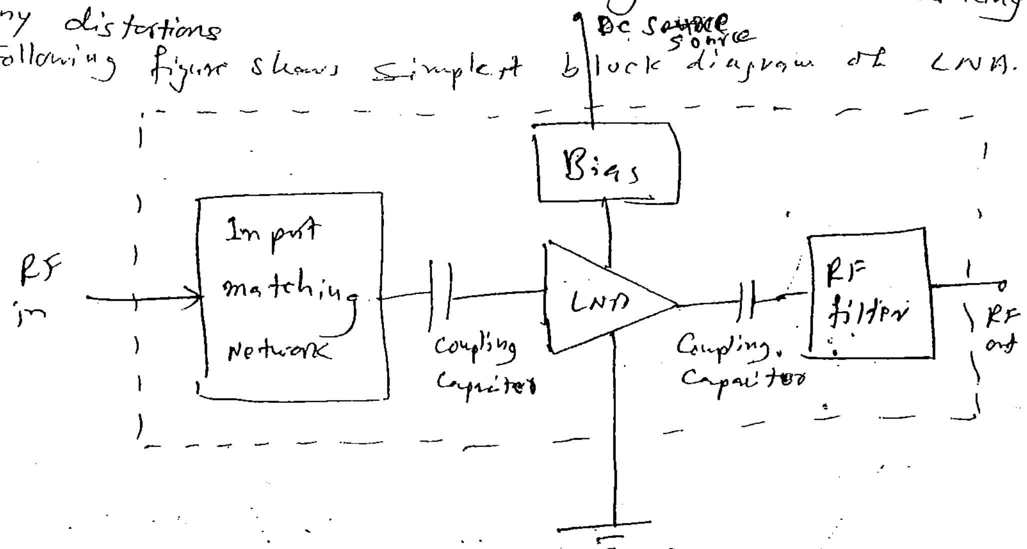
An LNA combines a low noise figure, reasonable gain, and stability without any oscillations (noise) over entire useful frequency range.

The Low Noise Amplifier (LNA) operates in class A.

The LNA play an important role in the receiver designs. Its main function is to amplify extremely low signals without adding noise, thus preserving the required Signal to Noise Ratio (S/N) of system at extremely low power levels. (at extremely low input signals)

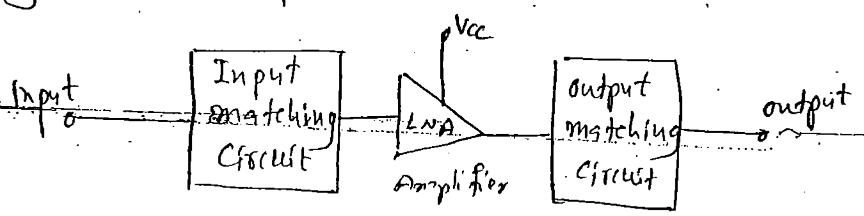
Additionally for large signal levels the LNA amplifies the received signal without introducing any distortions.

Following figure shows simplest block diagram of LNA.



An LNA have features: a low noise figure, reasonable gain and stability without oscillations over entire useful frequency range. The low noise Amplifier (LNA) always operates in class A [class A amplifier is characterised by quotient point more or less at the center of DC load line]

Following figure shows simplified block diagram of LNA



Designing of LNA is very difficult, because of its simultaneous requirement for high gain, low ~~nove~~ noise figure, good input and output matching and stability; And Linearity

Why designing of LNA is very difficult?

Low-noise amplifier

What is low noise Amplifier ?

Low-noise amplifier (LNA) is an electronic amplifier used to amplify possibly very weak signals (for example, captured by an antenna). It is usually located very close to the detection device to reduce losses in the feedline. This active antenna arrangement is frequently used in microwave systems like GPS, because coaxial cable feedline is very lossy at microwave frequencies, e.g. a loss of 10% coming from few meters of cable would cause a 10% degradation of the signal-to-noise ratio (SNR).

An LNA is a key component which is placed at the front-end of a radio receiver circuit. Per Friis' formula, the overall noise figure (NF) of the receiver's front-end is dominated by the first few stages (or even the first stage only). Using an LNA, the effect of noise from subsequent stages of the receive chain is reduced by the gain of the LNA, while the noise of the LNA itself is injected directly into the received signal. Thus, it is necessary for an LNA to boost the desired signal power while adding as little noise and distortion as possible, so that the retrieval of this signal is possible in the later stages in the system. A good LNA has a low NF (like 1 dB), a large enough gain (like 20 dB) and should have large enough intermodulation and compression point (IP3 and P1dB). Further criteria are operating bandwidth, gain flatness, stability and input and output voltage standing wave ratio (VSWR).

For low noise, the amplifier needs to have a high amplification in its first stage. Therefore JFETs and HEMTs are often used. They are driven in a high-current regime, which is not energy-efficient, but reduces the relative amount of shot noise. Input and output matching circuits for narrow-band circuits enhance the gain (*see Gain-bandwidth product*).

State important parameters in LNA design

LNA design

Low noise amplifiers are the building blocks of any communication system. The four most important parameters in LNA design are: gain, noise figure, and non-linearity and impedance matching. The design for LNA is based mainly upon the S-parameters of a transistor. The steps required in designing a LNA are as follows:

Design

There are two widely used types of devices the S-parameter and normal device. An S-parameter is a built-in device which does not require any type of external biasing because it has fixed S-parameters. Normal devices are like other transistors to which external bias can be applied. In designing a LNA, the S-parameter design is the most used.

Transducer

One of the crucial stages in designing a Low Noise Amplifier is proper selection of a transducer. The transducer selected should have a maximum gain and minimum noise figure(NF). Some examples of transistors that can be selected are- **ATF-34143** and **ATF-35143**.

Stability check

While designing any amplifier, it is important to check the stability of the device chosen, or the amplifier may function as an oscillator. For determining stability, calculate Rollet's Stability factor, (represented as variable K) using S-parameters at a given frequency. For a transistor to be stable, parameters must satisfy $K>1$ and $|\Delta|<1$.

Stability enhancement

Some of the techniques for enhancing the stability are adding a series resistance and adding a Source Inductance. In the former, a small resistance may be added in series with gate of the transistor. This technique is not used in LNA design because the resistance generates thermal noise, increasing the noise figure of the amplifier. Alternatively, an inductor may be added in series with the transistor gate. As an ideal inductor has zero resistance, it generates no thermal noise. It improves stability by reducing the gain of the amplifier by a small factor. Some of the inductors like 5.98nH and 3.1nH are used in 1st and 2nd stage respectively to improve the stability.

LNA application

State various applications of LNA.

LNA is used in various applications like ISM Radios, Cellular/PCS Handsets, GPS Receivers, Cordless Phones, Wireless LANs, Wireless Data, Automotive RKE, satellite communications, etc.

Satellite

In a satellite communications system, the ground station receiving antenna will connect to a LNA. The LNA is needed because the received signal is weak; it is usually a little above the noise floor. Satellites have limited power, so they use low power transmitters. The satellites are also distant and suffer path loss; low earth orbit satellites might be 200 km away; a geosynchronous satellite is 35 786 km away. A larger ground antenna would give a stronger signal, but making a larger antenna can be more expensive than adding a LNA. The LNA boosts the antenna signal to compensate for the feedline losses going from the (outdoor) antenna to the (indoor) receiver. In many satellite reception systems, the LNA includes a frequency block downconverter that shifts the satellite downlink frequency (e.g., 11 GHz) that would have large feedline losses to a lower frequency (e.g., 1 GHz) with lower feedline losses. The LNA with downconverter is called a low-noise block downconverter (LNB). Satellite communications are usually done in the frequency range of 100 MHz (e.g. TIROS weather satellites) to tens of GHz (e.g., satellite television).

Parameters

Write some electrical parameters of LNA.

Here some electrical parameters of LNA: Parameters of MAX 2640.

Operating supply voltage

Usually LNA require less operating voltage in the range of 2 .. 10 V. MAX 2640 operate at +2.7 .. +5.5 V.

Operating supply current

LNA require supply current in the range of mA, the supply current require for LNA is dependent on the its design and the application for which it has to be used. MAX 2640 which is used for satellite application requires a supply current of nearly 6 mA.

information signal

Chap ~~4~~: Digital communication (1)

Classification of Electronic communication systems based on nature of information signal, into two types

(1) Analog ~~digital~~ Communication systems
(2) Digital communication systems

Classify the digital Communication system.

Classify Analog communication system.

Electronic Communication systems
- Analog Communication system
  - Continuous wave systems (modulation)
    - Am
    - Fm
    - pm
  - pulse modulation
    - pAm
    - pwm
    - ppm
- Digital communication systems
  - pcm (pulse code modulation)
  - Dpcm [differential pcm]
  - Dm [Delta ~~Adtal~~ modulation)
  - ADm (Adaptive DM)

Analog communication:

The modulation systems or techniques in which one of the characteristics of carrier is changed in proportion with the instantaneous value of modulating signal is called an analog modulation system.

If the carrier is sinusoidal then its amplitude, frequency or phase is changed in accordance with the modulating signal ~~to~~ obtain Am, Fm or pm respectively. These are continuous wave modulation systems.

Analog modulation can be pulse modulation as well. Here the carrier is in the form of rectangular pulses. The amplitude, width (duration) or position of the carrier pulses is varried accordance with the modulating signal to obtain the PAm, Pwm or Ppm respectively.

Advantages of Analog communication:

(1) Transmitters and receivers are simple
(2) Low bandwidth requirement.
(3) FDM (Freq. ~~Division~~ multiplexing) can be used.

?

Write advantages and disadvantages of Analog comm. system.

Disadvantages of Analog communication:

(2)

(1) Noise affects the signal quality.

(2) Not possible to separate noise and signal.

(3) Coding is not possible.

Applications: ~~(1) Radio Broadcasting~~ (Am or Fm)

(2) TV Broadcasting

(3) Telephones.

Digital communication:

What is digital communication?

The modulation system or technique in which the transmitted signal is in the form of digital pulses of constant amplitude, constant frequency and phase is called as digital modulation system.

Examples: Pulse code modulation (pcm), Differential pcm

Delta modulation (DM)

Adaptive Delta modulation (ADM)

In pcm and Delta modulation, a train of digital pulses is transmitted by the transmitter. All pulses are of constant amplitude, width and position. The information is contained in the combination of transmitted pulses.

List advantages of Digital Communication.

Advantages of Digital communication.

(1) Digital communication has better noise immunity.

(2) Due to coding techniques, it is possible to detect and correct the errors introduced during data transmission.

(3) TDM (Time Division multiplexing) technique can be used to transmit many channels over a single common transmission channel.

(4) Due to digital nature of signal, it is possible to use the advanced data processing techniques such as digital signal processing, image processing, data compression etc.

(5) Digital communication is becoming simpler and cheaper as compared to analog communication due to availability of high speed computers and integrated circuits (ICs).

Disadvantages :

(3) State the disadvantages of Digital Communication.

(1) The bit rates of digital systems are high. Therefore they require a large channel bandwidth as compared to analog systems.

(2) System complexity is increased.

Applications :

State various applications of Digital communication.

(1) Long Distance Communication
(2) Satellite communication
(3) military communication, which uses coding.
(4) Data and computer communication
(5) Advanced Telephone systems.

Classify the pulse modulation systems :

Classification of pulse modulation

- pulse modulation
  - pulse analog modulation
    - PAM
    - PWM
    - PPM
  - pulse Digital modulation
    - PCM
    - DPCM
    - DM
    - ADM

Explain sampling process.

Sampling Process :

In the pulse modulation and digital modulation systems, the signal to be transmitted must be in discrete time format (Pulse form)

If the message signal is coming from a digital source (e.g. a digital computer) then it is in the proper form for processing by a digital communication system. But this is not always the case. The message signal can be analog in nature (e.g. speech or video signal) In such a case it has to be first converted

What is sampling process? (2m)

(4)

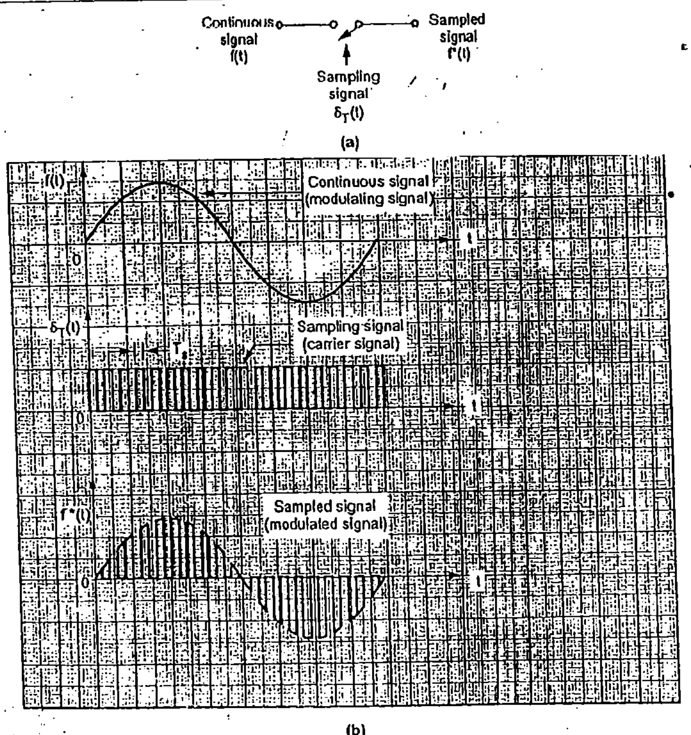
into a discrete time signal. We use the "sampling process" to do this. Thus using the sampling process we convert a continuous time signal into a discrete time ~~Signal. For~~ the sampling process, ~~to be~~ it is necessary to choose the sampling rate properly. The sampling process should satisfy the following requirements

(i) Sampled signal should represent the original signal faithfully.

(ii) One should be abled to reconstruct the original signal from its sampled version.

7.4 Sampling Process

In a sampling process a continuous time signal is converted to an equivalent discrete time signal. The Fig. 7.3 shows how this conversion can be done. As shown in the Fig. 7.3, switch position is controlled by the sampling signal. The sampling signal is a periodic train of pulses of unit amplitude and of period T_s. The time T_s is known as **sampling time** and during this time switch is closed so that sampled signal is equal to the input signal. During remaining time switch is open and no input signal appear at the output.



Note : "*" signal, f*(t) is always a sampled signal derived by sampling a continuous signal, f(t) using proper sampling, $\delta_T(t)$.

Fig. 7.3 Sampling circuit and waveforms

(5) State the sampling theorem

Explain Sampling theorem

Sampling theorem :

The sampling theorem states that, if the sampling rate (f_s) in any pulse modulation system, exceeds the maximum signal frequency (f_m), then the original signal can be reconstructed in the receiver with minimum distortion. (faithfully)

In other words if a continous [modulating or base band] signal $f(t)$ has frequency / frequencies in its spectrum with f_m is the highest frequency, then sampling frequency f_s must be at least twice the highest frequency in the modulating signal.

$$\therefore \; f_s \geq 2 f_m$$

$$f_{s_{min}} = 2 f_m = \text{Nyquist rate}$$

f_s = Sampling frequency or sampling rate

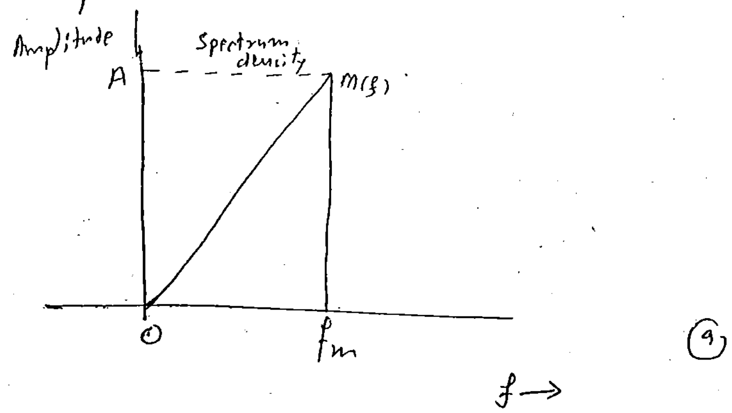
$$T_s = \frac{1}{f_s} = \text{Sampling period.}$$

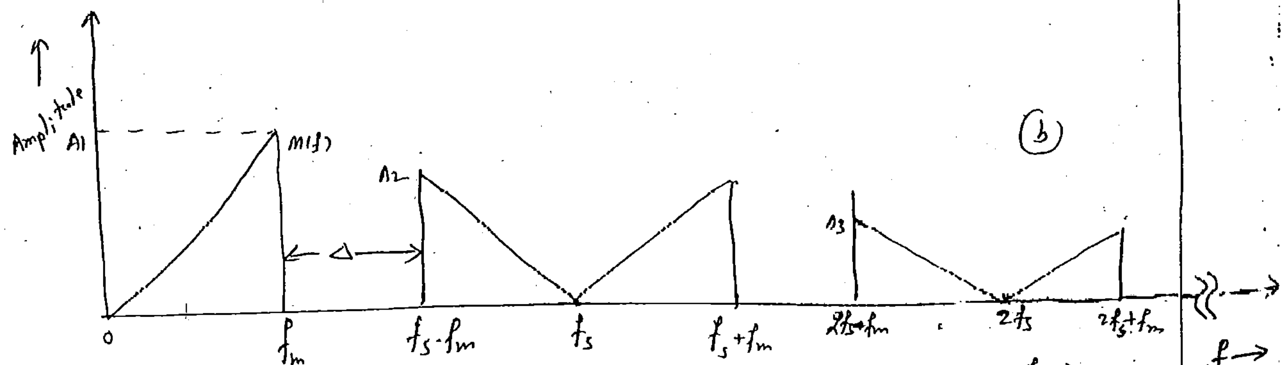
The minimum sampling frequency f_s equal to $2f_m$ cannot be achieved in practice because of the difficulty in realizing ideal filters. Therefore practically we must use the sampling frequency which is more than twice the maximum frequency in the baseband (modulating) signal. How much more is it depends upon the low pass filter characteristics and how faithfully the baseband signal (modulating signal) must be reproduced at the receiver end.

(The sampling theorem states that, if a continuous signal f(t) has frequency/frequencies in its spectrum with f_m is the highest frequency, then it is possible to convey all the information using sampled signal with $2f_m$ or more equally spaced samples per second.)

⑥

Following figure (a) shows a frequency spectrum $M(F)$ for modulating signal ~~by m(t)~~ and having highest frequency component f_m and figure (b) shows frequency spectrum for natural PAM wave.





$\boxed{f_s : \text{sampling freq.}}$ $\boxed{2f_s, 3f_s, 4f_s \text{ are its harmonics}}$

As seen from the spectrum (Double side Band with suppressed carrier (f_s)), the spectrum contains the original modulating signal with ~~freq~~ highest frequency f_m, which can be separated by a low pass filter having cut off frequency greater than f_m but less than $(f_s - f_m)$ i.e. smallest lower side band.

(Here $M(f)$ is the spectrum of the original modulating signal having f_m is its greater frequency.)

If $m(t)$ is to be recovered by low pass filtering at the receiving end, then the frequency seperation Δ between f_m and next lower order sideband $(f_s - f_m)$ must ~~be greater than~~ not be less than zero.

$$\Delta = (f_s - f_m) - f_m$$

$$\Delta = f_s - 2f_m$$

~~Therefore to have Δ greater than zero~~ need i.e.

With $\Delta \geq 0$, follows that $f_s - 2f_m \geq 0$

$$\therefore \boxed{f_s \geq 2f_m}$$

This condition is imposed on sampling frequency (f_s). i.e. the sampling frequency selected must be atleast ~~greater than~~ twice the highest ~~modulating~~ frequency of modulating (base band) signal (f_m) for effective filtering of harmonics at the receiver and reconstructing the original modulating signal without distortion.

The minimum sampling frequency required is found by ~~by~~ setting Δ equal to zero i.e.

$$\Delta = f_s - 2f_m = 0$$

$$\therefore f_s = 2f_m = \text{Nyquiest rate}$$

Thus minimum sampling frequency is twice the highest ~~modulating~~ frequency if modulating signal. This is known as sampling theorem.

(8) What is aliasing error? Explain the aliasing effect. What is the drawback of aliasing error?

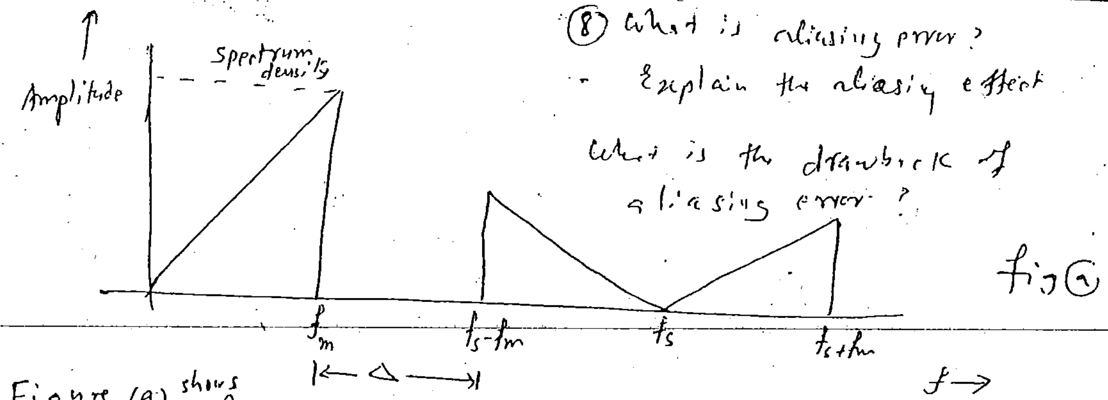
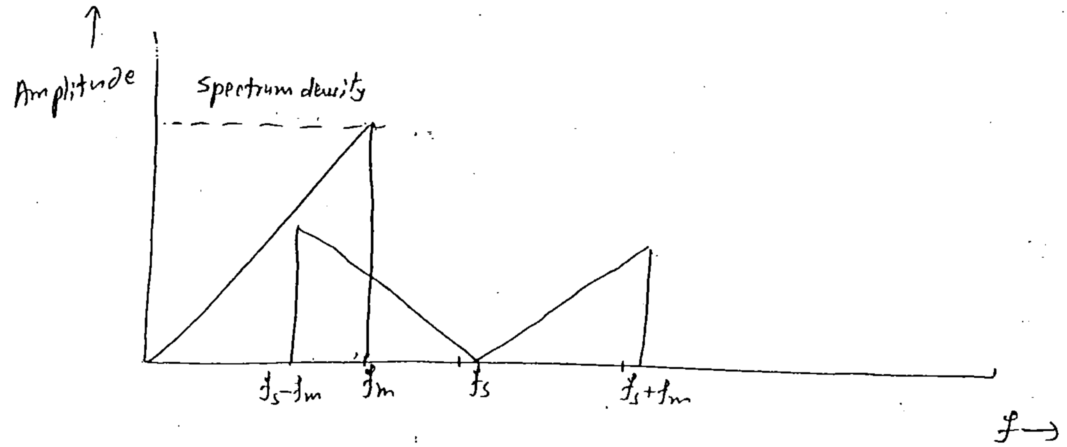


Figure (a) shows part of the spectrum for natural PAM wave showing the required separation Δ between f_m and lower side band frequency $f_s - f_m$.

When aliasing error occured:

~~Following figure (b) shows~~, If the sampling condition is not met ~~or~~ i.e. if the sampling frequency f_s is less than $2f_m$ then an error called aliasing or fold over error is observed. Parts of the spectrum overlap and once such overlap is allowed to occur, the spectrum can not longer be separated by filtering. Because of the high frequency components in DSBSC spectrum (for example $f_s - f_m$) appears in the low frequency part of the spectrum, this is called as aliasing effect. and which give rise error called as aliasing or foldover error. Following figure (b) shows aliasing effect.



(b) Aliasing effect.

Aliasing: The phenomenon of a high frequency in the spectrum of original signal (base band signal) taking on the identity of lower frequency in the spectrum of sampled signal is called as aliasing effect.

* Due to aliasing some of the information contained in the original signal is lost in the process of sampling.

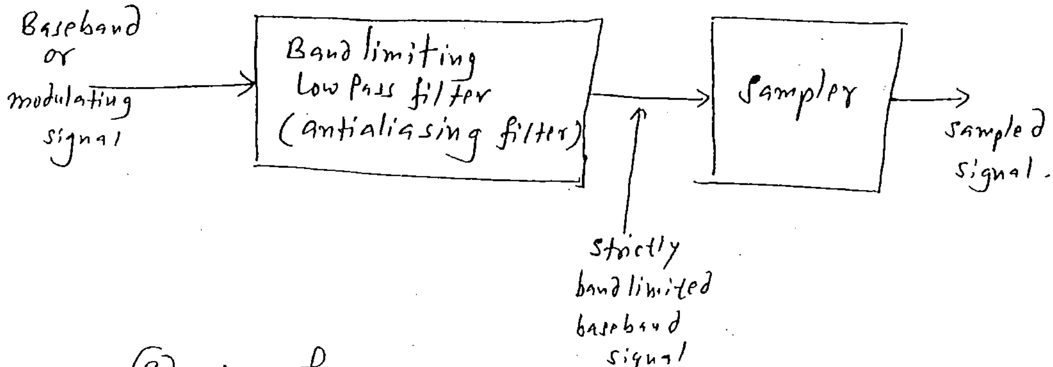
How to eliminate aliasing?

Explain how to eliminate aliasing error. (hm)

Aliasing can be completely eliminated if we take the following action:

(i) Use a band limiting low pass filter and pass the modulating (base band) signal through it before sampling as shown in the following figure.

(ii) This filter has a cutoff frequency at f_m, ~~ther~~ (maximum frequency of modulating signal), therefore it will strictly bandlimit the ~~si~~ modulating signal before sampling take place. This filter is also called as antialiasing filter.



(c) Use of a bandlimiting filter to eliminate aliasing.

(iii) The system designer must then ensure that the sampling

(18)

frequency (f_s) is atleast twice the f_m. For example: It is standard practice to use an anti-aliasing filter with a cutoff frequency of 4 KHz for digital telephony, with a corresponding sampling frequency of 8 KHz

Problem

$f_m = 4\,KHz$

$f_s = 2 f_m$

$= 2 \times 4$

$= 8\,KHz$

Pulse code modulation (PCM):

PCM is a type of digital pulse modulation like PAM, PWM or PPM but there is an important difference between them.

(pulse analog modulations)

PAM, PWM or PPM are "analog" pulse modulation systems, whereas PCM is a "digital" pulse modulation system.

That means the PCM output is in the coded digital form. It is in the form of digital pulses of constant amplitude, width and position. The information is transmitted in the form of code words. A PCM system consists of a PCM encoder (transmitter) and a PCM decoder (receiver).

(The essential operations in PCM transmitter are sampling, quantizing and encoding.)

7.5.1 Principle of operation of PCM

Similar to other pulse systems, PCM also uses the sampling process. In PCM, the peak-to-peak amplitude range which the signal to be transmitted, may have divided into a number of standard steps or levels. As these levels are transmitted in a binary code, the actual number of levels is always a power of 2. Each step is assigned a digital code number. The number of digits used in the binary code determines the total number of steps available and the accuracy of the transmitted value. For example, a 4 digit binary system provides a total of 16 steps, as shown in table 7.1 Each step is assigned a binary code.

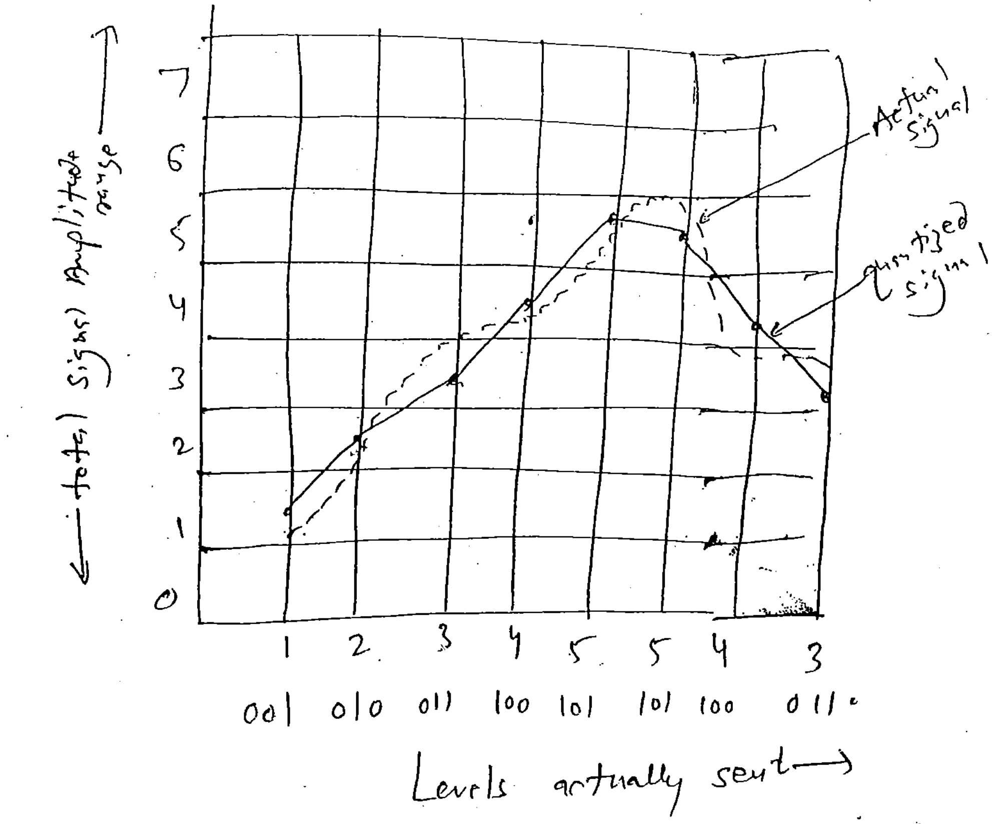
(levels)

Explain the principle of PCM.

What are the essential operations in PCM (1M)

Explain pcm technique using 8 levels.

Let us consider the principle of pcm using 8 levels.



Total peak to peak amplitude of range of modulating signal is divided into 8 standard levels. Since these levels are transmitted in binary codes, therefore actual number of levels should be power of 2. In our example no. of levels are $2^3 = 8$, but practically pcm signal use as many as 128 levels. (7 bit codes)

i.e. $2^7 = 128$ levels.

It is a cutoff spending

write advantages and disadvantages of PCM

(V2)

).2 Advantages of PCM :

(1) Very high noise immunity.

(2) Due to digital nature of the signal, repeaters can be placed between the transmitter and the receivers. The repeaters actually regenerate the received PCM signal. This is not possible in analog systems. Repeaters further reduce the effect of noise.

(3) It is possible to store the PCM signal due to its digital nature.

(4) It is possible to use various coding techniques so that only the desired person can decode the received signal. This makes the communication secure.

(5) The increased channel bandwidth requirement for PCM is balanced by the improved SNR. This is the due to the fact that PCM obeys an exponential law.

(6) There is a **uniform format** used for the transmission of different types of base band signals. Hence it is easy to integrate all these signals together and send them on the common network.

(7) It is easy to drop or reinsert the message sources in a PCM-TDM system.

).3 Disadvantages of PCM :

➤➤➤ **[Asked in Exam : Oct. 2006 !!!]**

(1) The encoding, decoding and quantizing circuitry of PCM is complex.

(2) PCM requires a large bandwidth as compared to the other systems.

20 Advantages, Disadvantages, Applications and Modifications in PCM :

(8) The PCM is considered to be the best modulation scheme to transmit the voice and video signals.

(9) All the advantages of PCM are due to the fact that it uses coded pulses for the transmission of information.

?0.1 Applications of PCM :

The of the applications of PCM are as follows :

(1) In telephony (with the advent of fibre optic cables).

(5) In the space communication, space craft transmits signals to earth. Here the transmitted power is very low (10 to 15W) and the distances are huge (a few million km). Still due to the high noise immunity, only PCM systems can be used in such applications.

state the applications of pcm technique.

It is a cutoff

expanding

Binary Code	Equivalent Amplitude	Waveform	Binary Code	Equivalent Amplitude	Waveform
0000	0		1000	8	
0001	1		1001	9	
0010	2		1010	10	
0011	3		1011	11	
0100	4		1100	12	
0101	5		1101	13	
0110	6		1110	14	
0111	7		1111	15	

Table 7.1

The sample of signal is transmitted, not as its actual amplitude, but rather as the nearest standard amplitude. This process is termed as "quantizing" the information.

Suppose at the sampling instant, the signal amplitude is 6.8 V. It is transmitted as the digit 7, because 7V is the standard amplitude nearest to 6.8V. If 16 levels [2^4] are used,4 binary digits are required corresponding to a given standard level. Then digit 7, in above example, will be sent as a series of pulses, corresponding to number 7. From the previous table, the number 7 is 0111, and could be sent as OPPP where P = Pulse and O = No pulse. Actually, it is often sent as a binary number back-to front, i.e. , as 1110, or PPPO, to make demodulation easier.

Thus, in PCM, the signal is continuously sampled quantized, coded, and transmitted, after each sample amplitude is converted to the nearest standard amplitude and into the corresponding back-to-front binary number.

7.5.2 Quantizing Noise [Quantization error] :

We have seen that in quantizing, the actual amplitude at the sampling instant is not transmitted, but, its nearest standard level is transmitted in coded form. The quantizing process then introduces a certain amount of error in the transmitted signal, which is called as quantizing error or quantizing noise. This error is small if a large number of standard levels are used for a given range of signal. A ~~5-place~~ digit code would allow 32 steps $[2^5]$ while a 7-digit code would result in 128 $[2^7]$ steps, giving better than 1% accuracy. This 7-digit code, giving 128 steps, is a standard for transmitting voice signals in telephony. However, when more number of levels are used, more bits are required to be sent per step, requiring more bandwidth, since the band width required is proportional to the number of bits per second.

What do you mean by Quantization error ? in PCM technique

To minimise

" Quantization error always use large number of standard levels for a given range of signal". Comment.

Base band

What is DPCM? Explain it with suitable wave form (14)

7.11 Differential Pulse Code Modulation

When samples of a signal are encoded by standard PCM system, the resulting encoded signal contains redundant information. This is illustrated in Fig. 7.37. Fig. 7.37 shows a continuous time signal m(t) by dotted line. This signal is sampled by flat top sampling at intervals T_s, $2T_s$, $3T_s$, ..., nT_s. The sampling frequency is selected to be higher than nyquist rate. ($f_s \geq 2f_m$) The samples are encoded by using 3 bit (8 levels) PCM. The sample is quantized to the nearest digital level as shown by small circles in the diagram, and the encoded binary value of each sample is written on the top of the samples. If we carefully observe the samples taken at $4T_s$, $5T_s$ and $6T_s$ and their encoded values, we find that the encoded values are same and thus redundant. Similarly the encoded values for samples taken as ~~$9T_s$ and 10 T_s~~ $7T_s$, $8T_s$ and $9T_s$ are also redundant.

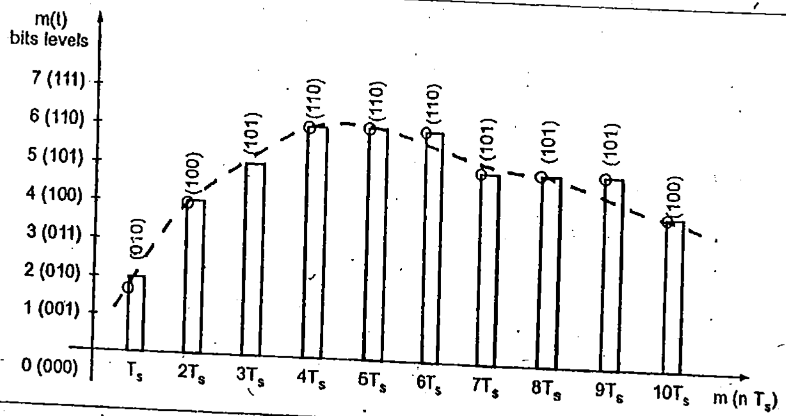


Fig. 7.37 Redundant information in PCM

If this redundancy is reduced, then overall bit rate will decrease and number of bits required to transmit signal will be reduced. As a result we can accommodate more signals on the same channel. The redundancy can be avoided by encoding difference between the samples, rather than the sample values themselves. This technique is known as **differential pulse code modulation (DPCM).**

The differential pulse code modulation works on the principle of prediction. The value of the present sample is predicted from the past samples. The prediction may not be exact but it is very close to the actual sample value. Fig. 7.38 shows the transmitter of Differential Pulse Code Modulation (DPCM) system. The sampled signal is denoted by m (n T_s) and the predicted signal is denoted by $\hat{m}$ (n T_s). The comparator finds out the difference between the actual sample value m (n T_s) and predicted sample value $\hat{m}$ (n T_s). This is called error and it is denoted by e (n T_s). It can be defined as,

$$e(nT_s) = m(nT_s) - \hat{m}(nT_s)$$

Bandwidth also reduced

Advantage of DPCM :-

As the redundant states [common states] (levels) are reduced by taking only one level from them. Thus total no. of levels required are less. Hence no. of bit required to represent the levels are reduced. Hence the Bandwidth is reduced. Hence can provide more no. of signals on the same channel.

State the advantage of DPCM.

(13) What is Delta modulation? Explain the ~~no~~ delta modulation. List the advantages of DM over PCM

Delta modulation,

In PCM system, N number of binary digits are transmitted per quantized sample. Hence the signaling rate and transmission channel bandwidth of PCM system are very large. These disadvantages can be overcome by using the delta modulation.

Delta modulation transmits only one bit per sample instead of N bits transmitted in PCM per sample. This reduces its signaling rate (Bit rate) and reduces bandwidth requirement to a great extent.

What is delta modulation?

Comment

"Delta modulation is a special case of DPCM (Differential PCM)". In delta modulation, ~~an~~ there is a single bit per sample. If the baseband (original) signal is larger than the ~~quantized~~ staircase approximation signal, then the output of DM is 0. If the baseband (original) signal is smaller than the ~~quantized~~ staircase approximation signal then the output of DM is 1.

The delta modulation is very attractive because of extremely simple coding and decoding procedures and the quantizing process is also very simple. However delta modulation cannot ~~rapidly~~ handle rapid amplitude variations and therefore quantizing noise tends to be very high.

What is the disadvantage of DM

Principle of ~~In~~ operation:

(i) In the basic or linear D.M., a staircase approximated sampled input signal is produced as shown in the following figure.

(ii) Then original signal (baseband signal) and ~~its~~ staircase approximated signals are compared to produce the difference signal.

(iii) This difference signal is quantized into only two levels ~~a~~ $\pm\delta$ corresponding to positive and negative difference respectively.

(iv) ~~That means if the approximated signal $m'(t)$~~ lies below ~~m(t)~~ ~~baseband signal m(t)~~ at the ~~sampling instant~~, ~~then~~ As shown in the figure, the D.M. output is 1 if the staircase approximated signal $m'(t)$ is increased by "δ" with respect to baseband signal $m(t)$, ~~at~~ i.e. at sampling instants 1, 2, 3, 4, 5, 6 ~~and~~ and 8.

(v) Whereas D.M. output is 0 if $m'(t)$ is decreased by "δ" i.e. at sampling instants 7, 9, 10 and 11)

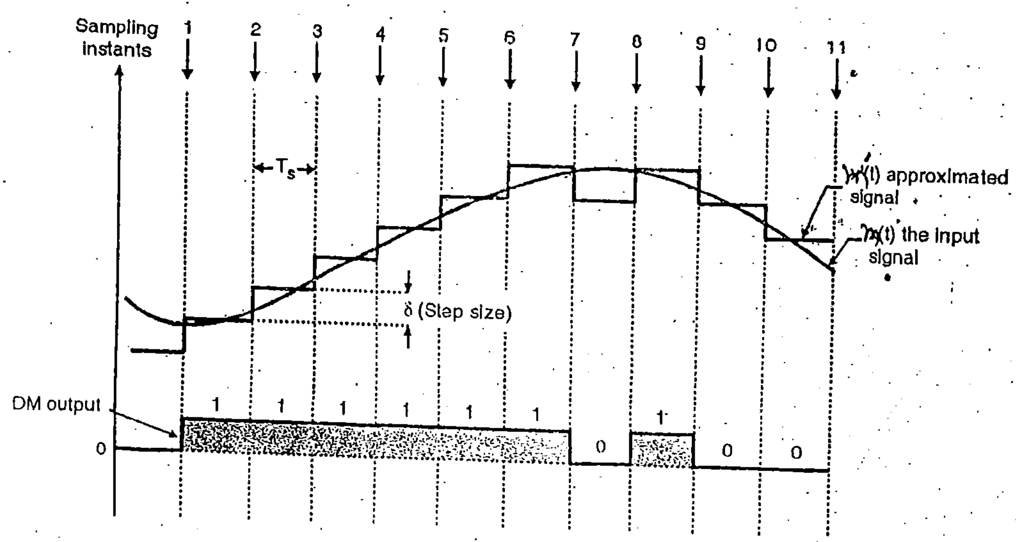


Fig. 3.13.1 : D.M. Waveforms

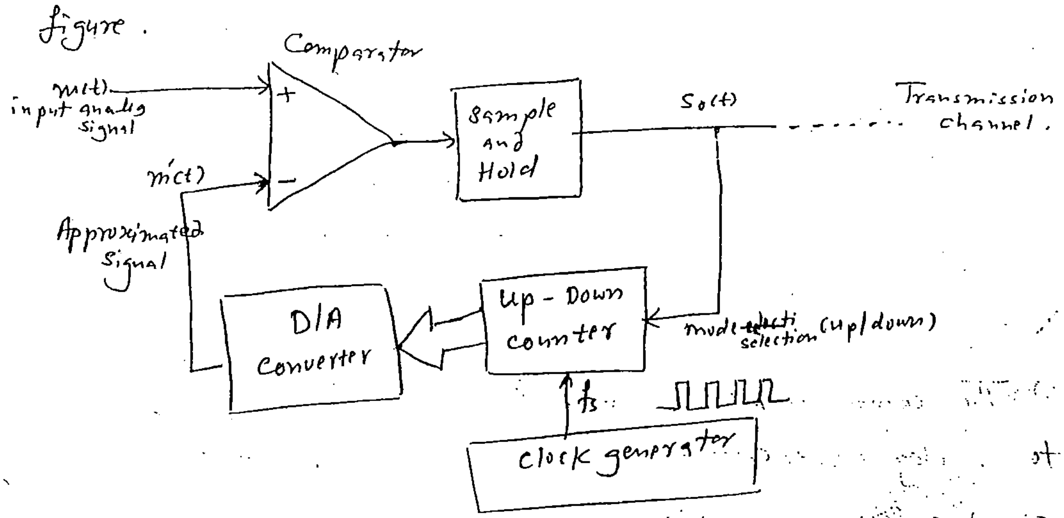
Explain the priciple of D.M with suitable wave forms.

In delta modulation, the present sample value m(t) is compared with the approximate value m'(t) and the result of this comparision is either bit 1 or bit 0 is transmitted, depends on whether the present sample value is higher than or lower than the approximate value of input signal.

Explain Delta modulator with suitable block diagram

Delta Modulator [~~Delta mo~~ or DM Transmitter]

Block diagram of delta modulator is shown in the following figure.



The operation of the circuit is given as follows:

(1) m(t) is the analog input signal and m'(t) is the quantized (approximated) version of m(t). Both these signals are applied to a comparator

(2) The comparator output goes high if m(t) > m'(t) and it goes low if m(t) < m'(t). Thus the comparator output is either 1 or 0. The sample and Hold circuit will hold this level (0 or 1)

(3) The output of sample and Hold circuit is transmitted as the output of D.M. system. Thus in DM, the information which is transmitted is only whether m(t) > m'(t) or vice versa. Also note that one bit per cycle is being sent. This will reduce the bit rate and hence the Bandwidth.

(4) The transmitted signal is also used to decide the mode of operation of an up/down counter.

Advantages of Delta modulation: Low signaling rate and low transmission channel bandwidth, because in delta modulation only one bit is transmitted per sample. The delta modulator transmitter and receiver are less complicated as compared to pcm.

(18)

The counter output increases by 1 if $S_o(t) = 1$ (Up counting) and it decremented by 1 if $S_o(t) = 0$ (down counting), at the falling edge of each clock pulse. This is shown in the following figure.

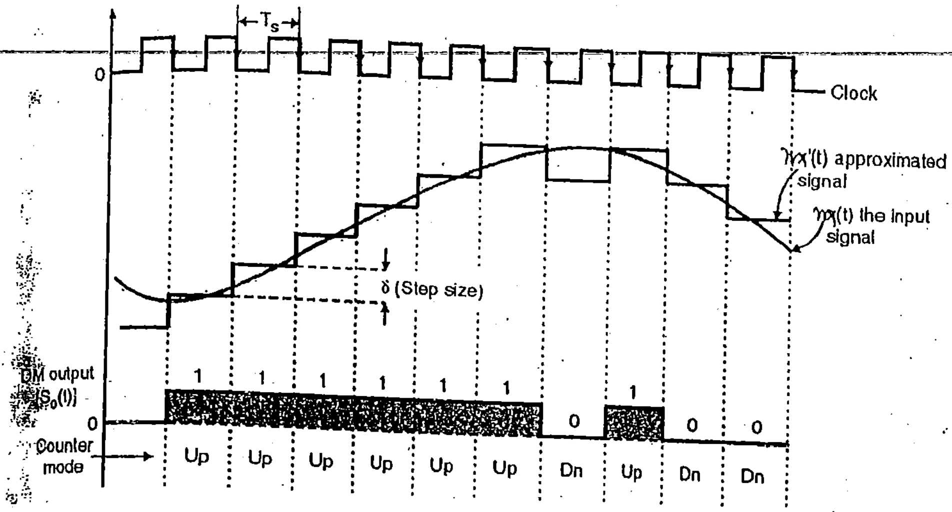


Fig. 3.13.5 : D.M. waveforms

(5) The counter output is converted into analog signal by Digital to analog (D/A) converter. Thus we get the approximated (quantized) signal $m'(t)$ at the output of D/A converter.

Features of D.M. (Advantages of D.M.) :-

(1) A one bit codeword for each output. Hence no need of framing.

(2) Simplicity of design for transmitter and receiver as compared to pcm and DPCM

(3) Reducing (low) Bit rate (signaling rate) and ~~Bandwidth~~ low transmission channel bandwidth because in delta modulation only one bit is transmitted per sample.

Applications of D.M. :-

(1) For some types of digital communications

(2) For digital voice storage.

List the applications of DM

Write the features (advantages) of DM

D.M. Bit rate (signaling rate);

$$\text{D.M. bit rate } (r) = \text{Number of bits transmitted/second}$$
$$= \text{Number of samples/second} \times \text{Number of bits/sample}$$
$$= f_s \times 1$$
$$= f_s$$

* Define Bit rate or signaling rate of D.M.

(19) ~~is transmitted~~ in case of DM. Comment.

Thus the D.m. bit rate is (1/N) times the bit rate of a PCM system, where N is the number of bits per second transmitted for PCM system. e.g. for 7 bit pcm system the bit rate is 1/7 i.e. only one bit transmitted in D.M instead of 7 bits in case of pcm

Disadvantages of Delta modulation: (or) Quantization Noise (distortions) in D.m. system: List the types of quantization error in DM system.

The DM system having two types of Quantization errors. (Disadvantages)

(1) slop overload distortion:

(2) Granular noise.

List the disadvantages of D.m. Explain slop overload distortion and Granular noise in DM system

(1) Slope overload distortion:

The slope overload distortion occurs when the rate of change of analog input signal m(t) is too high for the staircase approximated signal m'(t) to follow. If the slope of analog signal m(t) is much higher than that of m'(t) over a long ~~a long~~ duration, then m'(t) will not be able to follow m(t), at all. The difference between m(t) and m'(t) is called as the slope overload distortion. Thus the slope overload error occurs when slope of m(t) is much larger than slope of m'(t). Comment

In the following figure at points A, B and C, the D.m. system suffers from slope overload distortion.

Since the step size of ~~Delta mod~~ staircase approximated signal m'(t) remain fixed in D.m. system. Therefore in case of horizontal analog input signal m(t), the slope overload distortion is minimum and it is maximum along vertical analog input signal m(t)

The slope overload distortion can be minimised by increasing the step size δ of m'(t) ⊗. But increase in step size δ of m'(t), increases granular noise.

⊗ or by increasing the sampling frequency f_s. ~~Here~~ If f_s is increased, signaling rate and hence Bandwidth requirements will be more.

(2) Granular noise:

When the input signal m(t) having relatively constant amplitude, the stair case approximated signal m'(t)

Disadvantages of Delta modulation: Two distortions slope overload error and granular noise present. The signaling rate with no slope overload error will be much higher than that of pcm.

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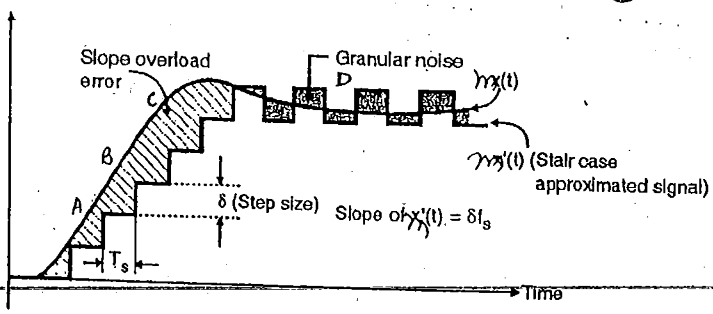


Fig. 3.13.8 : Distortions in D.M.

Q) What is adaptive Delta modulation? ?

comment

will hunt above and below m(t) as shown in above figure. This hunting produces granular noise. The granular noise occurs when the step size is ~~too~~ large compared to small variations in the input m(t) signal. This can be observed at point D. Here input m(t) is slowly ~~increasing~~ decreasing, but the staircase, ~~but~~ approximated signal waveform remains horizontal.

comment

The granular noise increases with increase in the step size. To reduce the granular noise, the step size should be as small as possible.

However this will increase the slope overload distortion.

In D.M. the step size is not variable. If it is made variable then the slope overload distortion and granular noise both can be controlled.

The Delta modulation with a variable step size of staircase signal m(t) is known as the adaptive Delta modulation (ADM).

Adaptive Delta modulation (ADM):

To overcome the quantization errors due to slope overload and

Comment

If step size in D.M is variable then slope over load distortion and granular noise both can be minimize.

distortion and granular noise, the step size of staircase approximated signal m'(t) is made adaptive to variations in the input signal m(t). Particularly in the steep segment of input signal m(t), the step size is increased. When the input m(t) is varying slowly the step size is reduced. Thus the step size is adapted as per level of input signal m(t). This method is called Adaptive Delta modulation (ADM)

Explain ADM method? List the types of ADM.

Types of ADM:

~~The typ~~ There are various types of ADM systems available depending on the type of scheme used for adjusting the step size.

In one type a discrete ~~set~~ changes in the step size is provided whereas in another type a continuous changes in step size is provided.

Comment

"In the ADM system, the step size is not constant." Rather when the slop overload ~~or~~ distortion occurs the step size becomes larger and therefore m'(t) will catch up or follow with m(t) more rapidly. [Whenever the slope of input signal is large, the step size of staircase approximated signal m'(t) is increased.]

On the other hand, when the input signal is varying slowly the step size is reduced.

Following figure shows block diagram of Adaptive Delta modulator [ADM transmitter].

If we compare this block diagram with that of the linear delta modulator, then we can find that except the up/down counter is replaced by the digital processor, and the remaining blocks are identical.

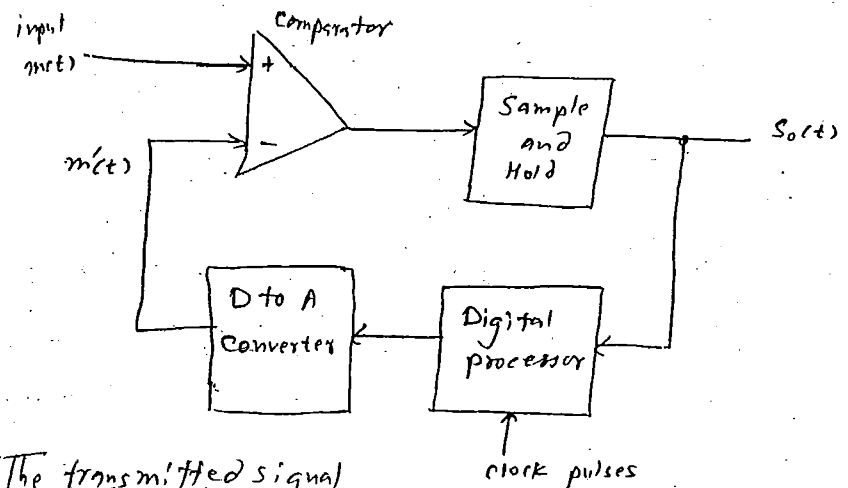
(22)

Explain ADM modulator with suitable block diagram (15) not ret ...

Operation:

(1) In response to the k^{th} clock pulse trailing edge, the processor generates a step ~~generated~~ which is equal in magnitude to the step generated in response to the previous i.e. $(k-1)^{th}$ clock edge.

(2) If the direction of both steps is same, then the processor will increase the magnitude of present step by "δ". If the directions are opposite then the processor will decrease the magnitude of the present step by "δ"



(a)

The transmitted signal of ADM is

(3) $S_o(t)$ is the output of ADM system is given by

$S_o(t) = +1$ if $m(t) > m'(t)$ just before k^{th} clock edge

and $S_o(t) = -1$ if $m(t) < m'(t)$ just before the k^{th} clock edge

Then the step size at sampling time k is given by

$$\underbrace{\delta(k)}_{\text{step size at } k^{th} \text{ clock edge}} = \underbrace{[\delta(k-1)]}_{\text{step size at } (k-1)^{th} \text{ clock edge}} \underbrace{S_o(k)}_{\text{output at } k^{th} \text{ edge}} + \underbrace{\delta}_{\text{Basic step size}} \underbrace{S_o(k-1)}_{\text{output at } (k-1)^{th} \text{ clock edge}} \quad \text{—— (1)}$$

Let us take an example:

Refer to the waveform given in fig. (b), Let us assume $k = 6$

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K^{th} clock edge

K^{th} clock edge

by

— (1)

at $(K-1)^{th}$

K edge

me $K=6$

(23)

i.e.: consider the 6^{th} clock edge

$\therefore K-1 = 5$ $\therefore \delta(K-1) = \delta(5) = \delta$

$\therefore S_o(K) = S_o(6) = +1$ $\because m(t) > m'(t)$ for $\delta(5)$ & $\delta(6)$

$S_o(K-1) = S_o(5) = +1$

substitute these in equation (1), we get

$\delta(6) = \delta + \delta = 2\delta$

so observe the fig (b) the step size at the 6^{th} clock edge is 2δ.

As shown in fig (b), due to variable step size, the slope overload error is reduced

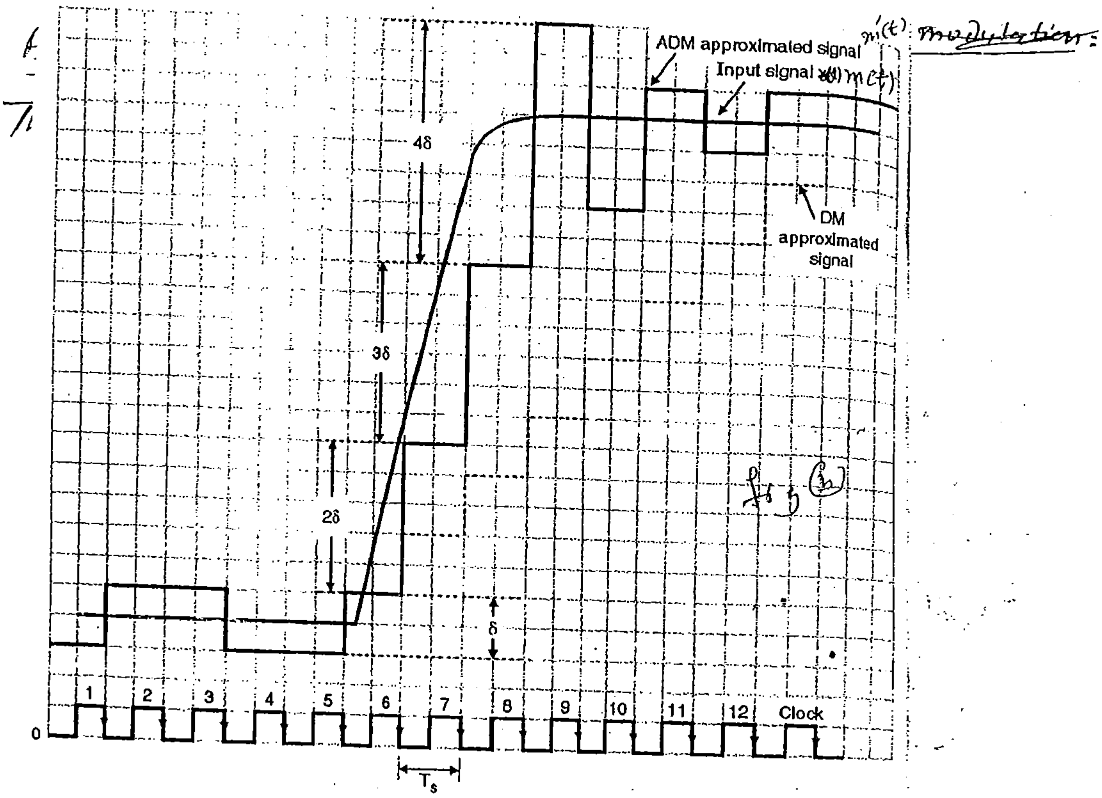

fig (b)

similarly when the input analog signal m(t) is slowly changing, then step size(δ) is reduced to minimise granular noise $(m(t) < m'(t))$

~~List~~

(28)

Advantages of Adaptive Delta modulation over Delta modulation

(1) Reduction in slope overload distortion and granular noise.

(2) Improvement in signal to noise ratio.

(3) Wide dynamic range due to variable step size.

(4) Low signaling rate [hence Low B.W.]

(5) Simplicity of implementation.

With the help of suitable block diagram of PCM transmitter Explain generation of PCM signal.

Generation ~~and Demodulation~~ of PCM [PCM Transmitter or encoder]

The basic elements for generation of pulse code modulation are given as follows (fig a). It is nothing but the block diagram of the PCM transmitter

The analog signal m(t) is passed through Bandlimiting low pass filter (antialiasing filter), which avoids aliasing. [Aliasing refers to the overlapping of parts of spectra and once such overlap is allowed to occure, the spectra can no longer be separated by filtering.]

Then band limited analog signal from the output of antialiasing filter is applied to a sample and Hold circuit, where it is sampled. The output of sample and hold circuit have flat top samples (i.e. PAM signal).

Then these flattop samples are quantized in the Quantizer. [Quantization process is the process of approximation. The quantization is used to reduce the effect of noises and ~~reduce~~ also reduce the voltage levels and hence Bandwidth.]

The combined effect of sampling and Quantization produces the quantized PAM signal at the output of Quantizer.

Comparision of Digital pulse modulation systems.

Comparision of pcm, DM, ADM and DPCM

parameter	pcm	DM	ADM	DPCM
Number of bits per sample	N can be 4, 8, 16, 32, 64 or 128	N = 1	N = 1	N is more than 1 but less than that for pcm
stepsize	Depends on the number of levels	step size is fixed	Step size is variable	step size is fixed
Distortions/errors	Quantization error	slope overload and granular noise	—	—
Signaling rate and Bandwidth	Highest	Low (if the input is slow varying)	Lowest	Less compared to pcm and more than DM
system complexity	Complex	Simple	Simple	Complex
Feedback from output	No feed back	feedback is present	feedback present	No feed back

	pcm	DM	ADM	DPCM
Noise immunity	very good	very good	very good	very good

Compare various Digital pulse modulation system (4m)

- This error should be as small as possible.
- To minimize the quantization error we need to reduce the step size "s" by increasing the number of quantization levels Q.

Why is quantization required ?

- If we do not use the quantizer block in the PCM transmitter, then we will have to convert each and every sampled value into a unique digital word.
- This will need a large number of bits per word (N). This will increase the bit rate and hence the bandwidth requirement of the channel.
- To avoid this, if we use a quantizer with only 256 quantization levels then all the sampled values will be finally approximated into only 256 distinct voltage levels.
- So we need only 8 bits per word to represent each quantized sampled value.
- Thus the number of bits per word can be reduced. This will eventually reduce the bit rate and bandwidth requirement.

2.19.5 Quantization error or quantization noise ϵ :

- The difference between the instantaneous values of the quantized signal and input signal is called as quantization error or quantization noise.

$$\epsilon = x_q(t) - x(t) \qquad \text{...(2.19.2)}$$

- The quantization error is shown by shaded portions of the waveform in Fig. 2.19.5.
- The maximum value of quantization error is $\pm s/2$ where s is step size.
- Therefore to reduce the quantization error we have to reduce the step size by increasing the number of quantization levels i.e. Q.
- The mean square value of the quantization is given by,

$$\text{Mean square value of quantization error} = \frac{s^2}{12} \qquad \text{...(2.19.3)}$$

The derivation for this expression is given later in this chapter.

- The relation between the number of quantization levels Q and the number of bits per word (N) in the transmitted signal can be found as follows :
  - Because each quantized level is to be converted into a unique N bit digital word, assuming a binary coded output signal,
  - The number of quantization levels Q = Number of combinations of bits/word.

$$\therefore \quad Q = 2N \qquad \text{...(2.19.4)}$$

  - Thus if N = 4 i.e. 4 bits per word then the number of quantization levels will be 2^4 i.e.16.

2.19.6 Companding :

- Companding is non-uniform quantization. It is required to be implemented to improve the signal to quantization noise ratio of weak signals.
- The quantization noise is given by,

$$N_q = s^2/12$$

- This shows that in the uniform quantization once the step size is fixed, the quantization noise power remains constant.

elta modulation:

lay noise.

f suitable of pcm transmitter. pcm signal.

Why quantization is required in pcm (20)

(28)

X

27

Explain quantization process in PCM generation (transmitter)

2.19.4 Quantization Process :

➤➤➤ [Asked in Exam : April 2010 !!!]

- Quantization is a process of approximation or rounding off. The sampled signal in PCM transmitted is applied to the quantizer block.
- Quantizer converts the sampled signal into an approximate quantized signal which consists of only a finite number of predecided voltage levels.
- Each sampled value at the input of the quantizer is approximated or rounded off to the nearest standard predecided voltage level.
- These standard levels are known as the "quantization levels". Refer to Fig. 2.19.5 to understand the process of quantization.

The quantization process takes place as follows :

- The input signal x (t) is assumed to have a peak to peak swing of V_L to V_H volts. This entire voltage range has been divided into "Q" equal intervals each of size "s".
- "s" is called as the step size and its value is given as,

$$\boxed{s = \frac{V_H - V_L}{Q}} \qquad \ldots(2.19.1)$$

In Fig. 2.19.5, the value of Q = 8

- At the center of these ranges, the quantization levels $q_0, q_1, \ldots q_7$ are placed. Thus the number of quantization levels is Q = 8. ~~These are also called as decision thresholds.~~
- x_q (t) represents the quantized version of x (t). We obtain x_q (t) at the output of the quantizer.
- When x (t) is in the range Δ_0, then corresponding to any value of x (t), the quantizer output will be equal to "q_0".
- Similarly for all the values of x (t) in the range Δ_1, the quantizer output is constant equal to "q_1".
- Thus in each range from Δ_0 to Δ_7, the signal x (t) is rounded off to the nearest quantization level and the quantized signal is produced.
- The quantized signal x_q (t) is thus an approximation of x (t). The difference between them is called as **quantization error or quantization noise.**

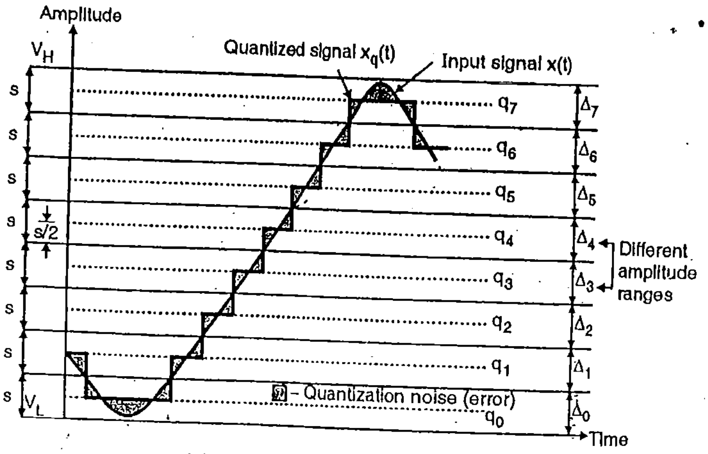

Fig. 2.19.5 : Process of quantization

This error should be as small as possible. To minimise the quantization error we should step size 's', by increasing the quantization levels 'Q'.

select

The quantized PAM pulses are applied to an encoder, which is basically an ~~A/D~~ A/D converter. Each quantized level is converted into an N bit digital word by the A/D converter.

Finally the output of encoder (A/D converter) is converted into stream of pulses (i.e. serial form) by the parallel to serial converter.

Thus at the pcm transmitter output, we get a train of digital pulses.

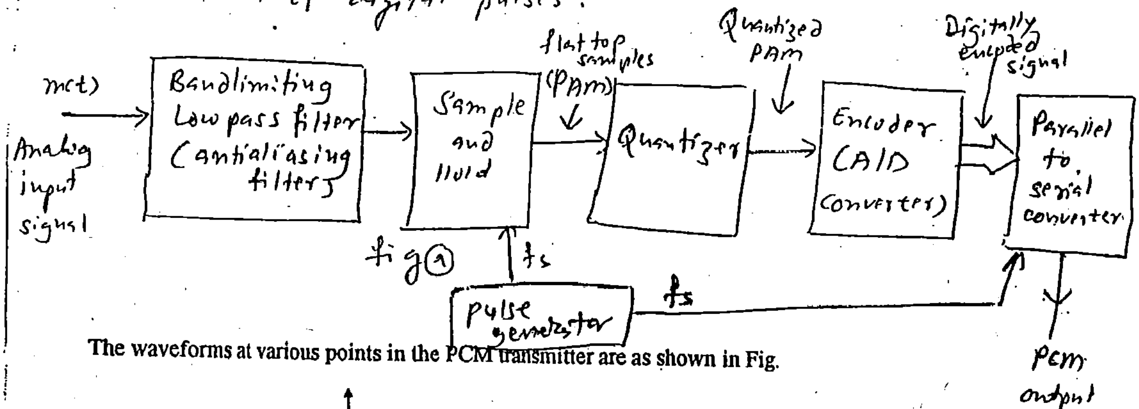

fig 1

The waveforms at various points in the PCM transmitter are as shown in Fig.

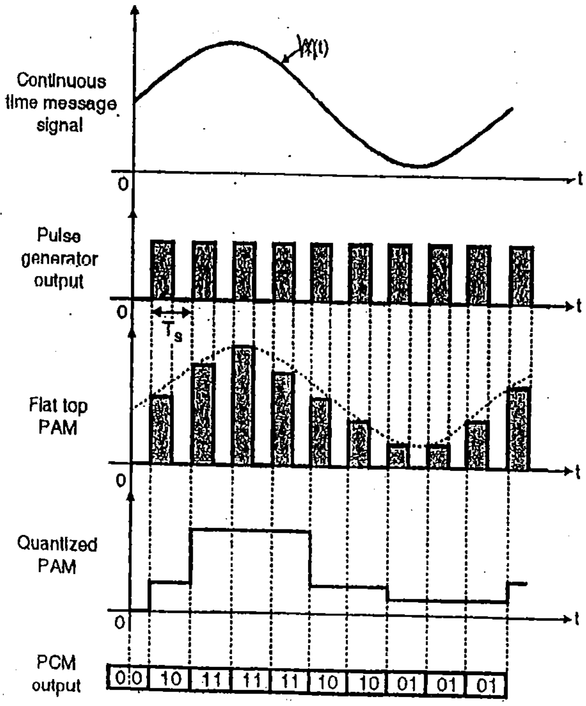

fig b

generation (transmitter)
nd Demodulation

April 2010 [II]
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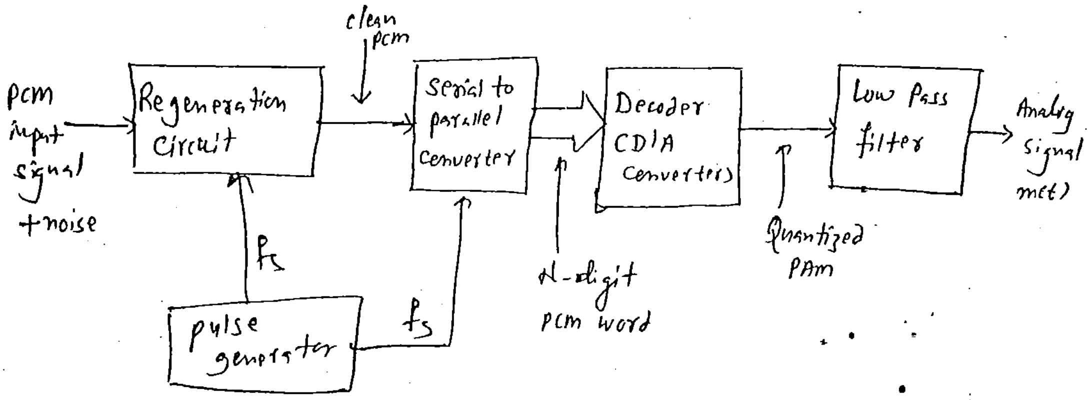
Explain the Demodulation of pcm (383) with suitable block diagram. (29)

Demodulation of PCM:

[PCM Receiver (Decoder)] Draw the block diagram of PCM Receiver and explain it. (6)

Following figure shows the block diagram of a PCM receiver (PCM Demodulator).

A PCM signal along with noise is available at the receiver input. The Regeneration circuit will separate PCM signal from noise. Then PCM signal is passed through a serial to parallel converter. The output of this block is applied to a Decoder (D/A converter). Which performs exactly the opposite operation of encoder (A/D converter) in case of PCM transmitter. Thus Quantized PAM signal is obtained at the output of D/A converter. Finally this quantized PAM signal is passed through a low pass filter to recover the original analog signal m(t).



Signaling Rate (Data Transfer Rate) and Transmission Bandwidth of PCM:

We know $Q = 2^N$ or $N = \log_2 Q$.

Where Q = Number of quantization levels.

N = Number of Bits per word [No. of pulses required in a word]

The input signal m(t) is sampled at the sampling rate f_s (sampling frequency) i.e. there are f_s number of samples per second.

Define signal rate of pcm and D.m.

Each of these samples is then converted into N bit digital word.

$\therefore$ Number of bits/sec = number of samples/sec × Number of bits/sample

$= f_s \times N$

But signaling rate is nothing but the number of bits per second.

$\therefore$ Signaling rate of pcm (r) $= N f_s$

The transmission bandwidth of pcm is equal to half the signaling rate.

$\therefore$ Transmission bandwidth of pcm $= \frac{1}{2} N \cdot f_s$.

Problem,

(1) A pcm system uses uniform quantizer followed by a 7 bit encoder. The system bit rate is 50 m bits/sec. calculate the maximum bandwidth of message signal for which this system operates satisfactorily.

Bit rate $r = 50$ m bits/sec

& $N = 7$

$\therefore$ Bit rate $r = N f_s$

$\therefore f_s = \frac{r}{N} = \frac{50 \times 10^6}{7} = 7.14$ MHz = No of samples/s

Maximum bandwidth $= f_s/2 = \frac{7.14}{2} = 3.57$ MHz

~~The bandwidth of a video signal is 4.5 MHz. This signal is to be transmitted using pcm with the number of quantization levels Q = 1024.~~
The sampling rate should be 20% higher than sampling rate

(2) If number of quantizing levels are 16, then find number of binary pulses required in one code group

$N = \log_2 Q = \log_2 16 = \log_2 2^4 = 4$

Thus 4 binary pulses are required in one code group.

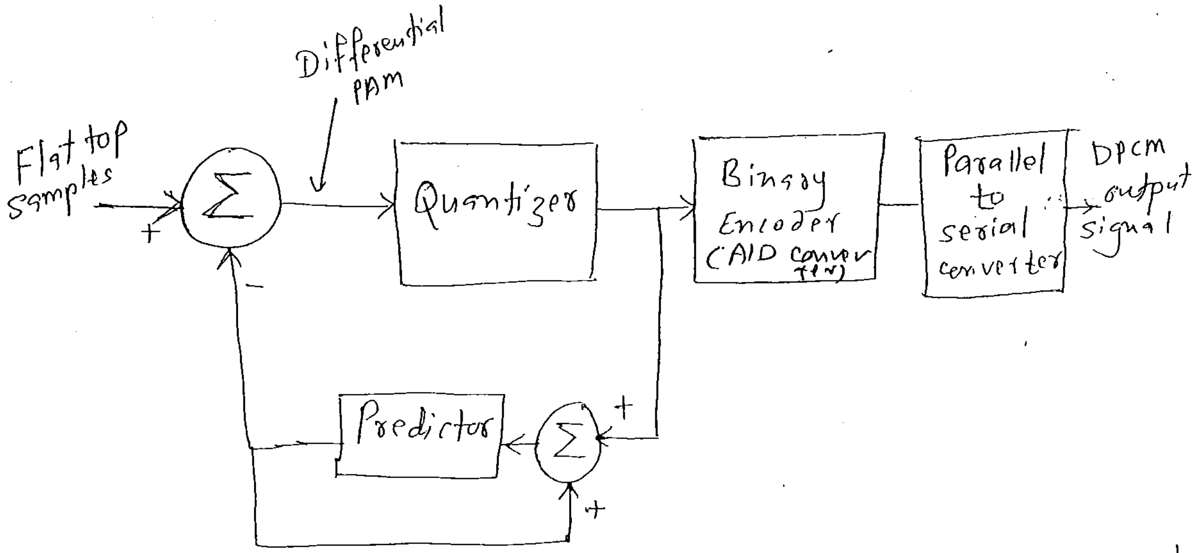
DPCM Transmitter

By removing the redundancy before encoding, we can obtained a more efficient coded signal.

Differential pulse code modulation (DPCM) is a technique in which the difference between samples, rather than the sample values themselves, is encoded in binary form.

Following figure shows a block diagram of DPCM transmitter, where it is assumed that flat top sampling has been obtained by a sample and Hold Circuit.

The input to the quantizer is the difference between the flat top samples and the feed back signal obtained from the output of quantizer (feedback through predictor block). The feed back signal is produced by the Predictor block, which is a digital type filter



The input to the predictor is the sum of its own output and recent difference level. The feed back flat top sample (from output of predictor) is subtracted from the incoming flat top sample to form the difference signal, that is quantized. This quantized PAM pulse are applied to an encoder which is basically A/D converter. Each quantized level is converted into N bit digital word by A/D converter.

Finally the output of encoder is converted into stream of pulses (i.e., serial form) by parallel to serial converter.

The predictor predict the values of future samples of input signal. This helps in reducing the redundancy.

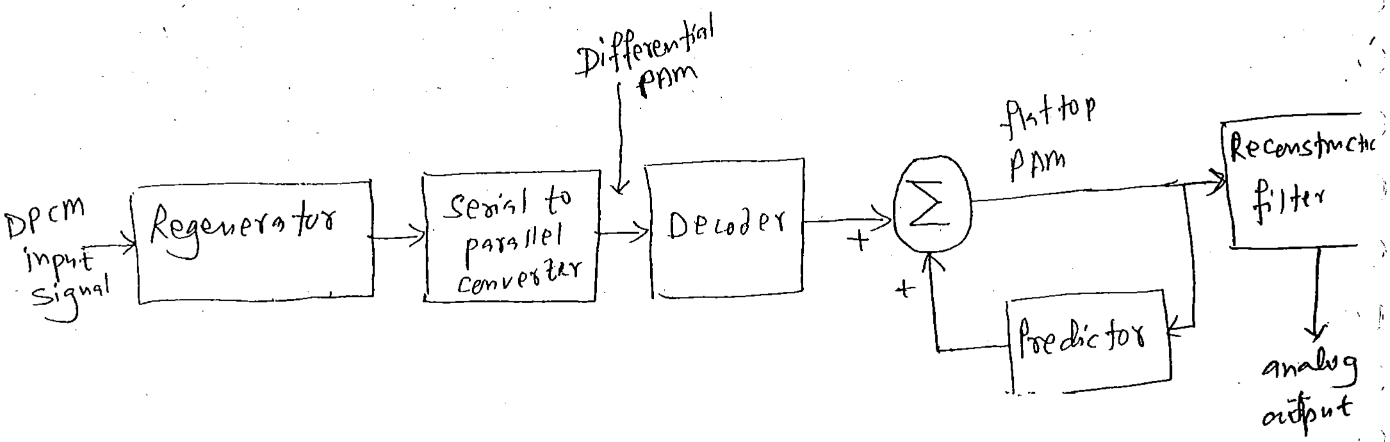
DPCM Receiver:

(32)

Following figure shows block diagram of DPCM receiver.

The regenerator block at receiver used to separate the incoming DPCM binary signal from noise. Then DPCM signal is passed through serial to parallel converter. Then incoming DPCM binary signal is decoded to form a difference PAM signal. This is fed into feed back loop similar to that used at the transmitter. The predictor & filter gives feed back flat top PAM signal, from the sum of its own output and most recent difference PAM signal from decoder.

The analog output is recovered from the flat top PAM signal by the reconstruction filter.



Draw the diagram and explain working of DPCM Transmitter/Receiver (4m)

(33) Companding: (Companded PCM):-

What is the purpose of companding? (1m)
What is companding? (1m)

Companding is non-uniform quantization. It is required to improve the signal to quantization noise ratio of weak signals.

Write short note on companding. (6m)

The quantization noise is given by

Draw compand curves for pcm signal. (2m)

$$N_q = S^2/12 \qquad S: \text{step size}.$$

This shows that in the uniform quantization, once the step size (S) is fixed, the quantization noise power remains constant. Thus for weak signals, the signal to quantization noise ratio is very poor. This will affect the quality of signal and it is not tolerable. The solution for this problem is to use companding.

Companding is the term derived from two words: Compression and expansion.

* The weak signals are amplified and strong signals are attenuated, before applying them to uniform quantizer (at transmitter side). This process is called as compression and block that provides it is called as compressor.

At the receiver end, exactly opposite process is followed, which is called expansion. The circuit used for providing expansion is called as an "expander".

* Practically it is difficult to implement the non-uniform quantization because it is not known in advance about changes in the signal levels.

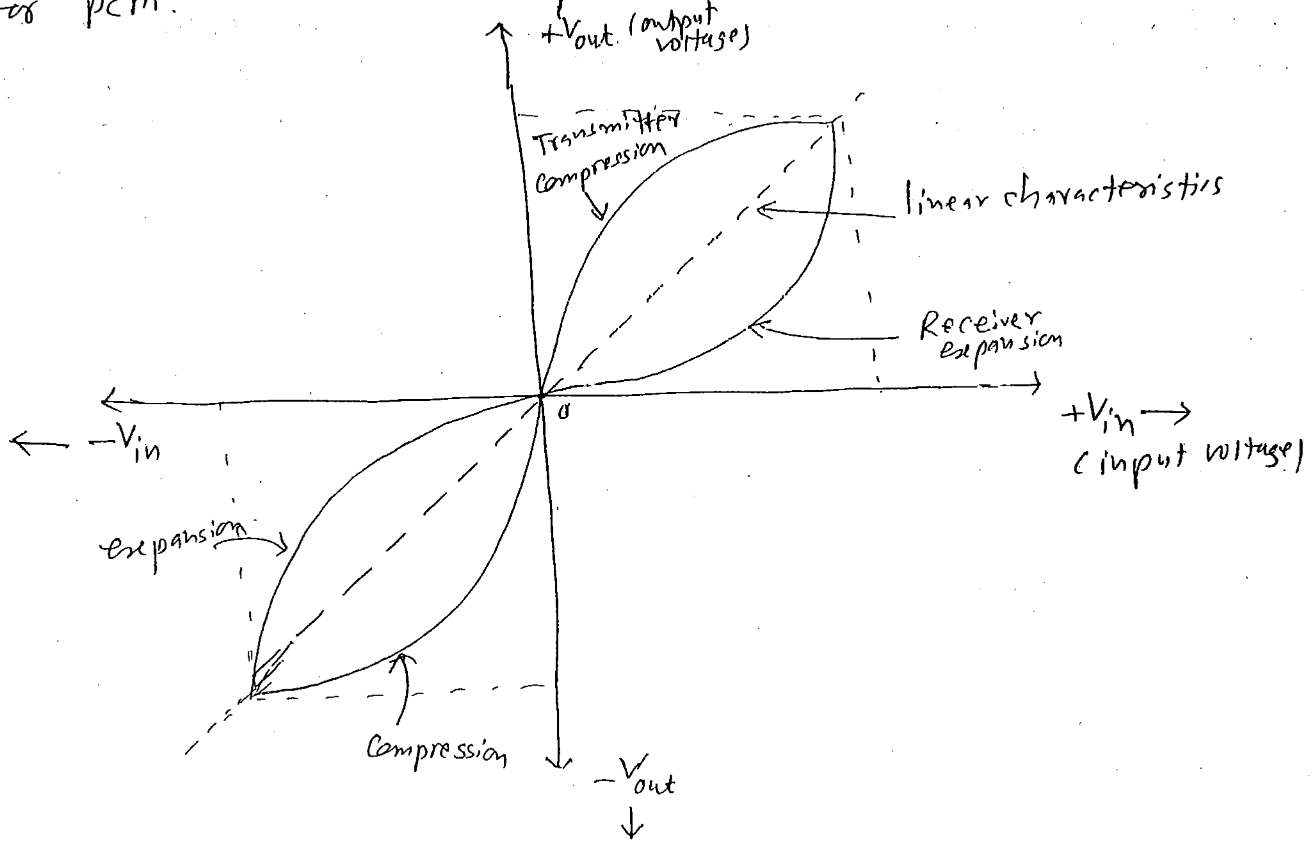
The compression of signal at the transmitter and expansion at the receiver is combined to be called as companding.

The compressor provides a higher gain to the weak signals and smaller gain to the strong input signals.

Thus weak signals are artificially boosted to improve the signal to quantization ratio. (i.e. effect of quantizing noise upon them is reduced.)

The expander ensures that all the artificially boosted signals by the compressor are brought back to their original amplitudes at the receiver.

Following figure shows companding curves for pcm.



4.9 Introduction to Multiplexing :

- Multiplexing is the process of simultaneously transmitting two or more individual signals over a single communication channel.
- Due to multiplexing it is possible to increase the number of communication channels so that more information can be transmitted.
- The typical applications of multiplexing are in telemetry and telephony or in the satellite communication.

4.10 Concept of Multiplexing :

write the applications of multiplexing,

- The concept of a simple multiplexer is illustrated in Fig. 4.10.1.
- The multiplexer receives a large number of different input signals.
- Multiplexer has only one output which is connected to the single communication channel.
- The multiplexer combines all input signals into a single composite signal and transmits it over the communication medium.
- Sometimes the composite signal is used for modulating a carrier before transmission.
- At the receiving end, of communication link, a demultiplexer is used to sort out the signals into their original form.

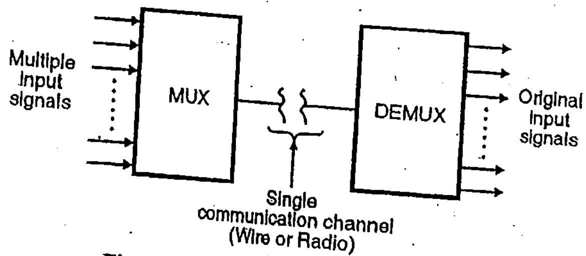


Fig. 4.10.1 : Concept of multiplexing

- The operation of demultiplexer is exactly opposite to that of a multiplexer.

4.10.1 Types of Multiplexing :

write three basic types of multiplexing.

There are three basic types of multiplexing. They are :

1. Frequency division multiplexing (FDM)
2. Time division multiplexing (TDM).
3. Wavelength division multiplexing (WDM).

4.10.2 Classification of Multiplexing :

classify the multiplexing.

- The multiplexing techniques can be broadly classified into two categories namely analog and digital.
- Analog multiplexing can be either FDM or WDM and digital multiplexing is TDM.
- Fig. 4.10.2 shows the classification of multiplexing techniques.

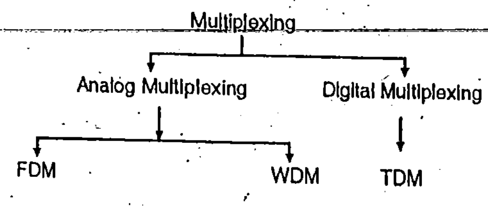


Fig. 4.10.2 : Classification of multiplexing techniques

- Generally the FDM and WDM systems are used to deal with the analog information whereas the TDM systems are used to handle the digital information.
- In FDM many signals are transmitted simultaneously where each signal occupies a different frequency slot within a common bandwidth.
- In TDM the signals are not transmitted at a time, instead they are transmitted in different time slots.

What is FDM? Explain

4.11 Frequency Division Multiplexing (FDM) : Write a note on FDM transmitter

- The operation of FDM is based on sharing the available bandwidth of a communication channel among the signals to be transmitted.
- That means many signals are transmitted simultaneously with each signal occupying a different frequency slot within a common bandwidth.
- Each signal to be transmitted modulates a different carrier. The modulation can be AM, SSB, FM or PM.
- The modulated signals are then added together to form a composite signal which is transmitted over a single channel.
- The spectrum of composite FDM signal is shown in Fig. 4.11.1(a).
- Generally the FDM systems are used for multiplexing the analog signals.

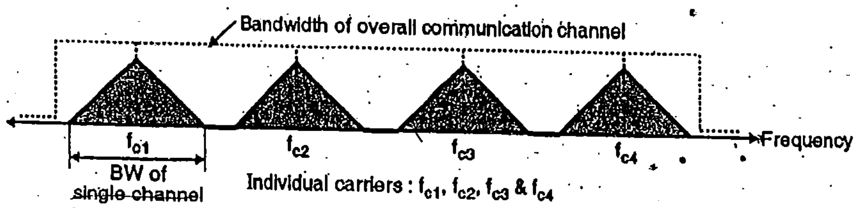


Fig. 4.11.1(a) : Spectrum of FDM signal

4.11.1 FDM Transmitter :

- Fig. 4.11.1(b) shows the block diagram of an FDM transmitter. The signals which are to be multiplexed will each modulate a separate carrier.
- The type of modulation can be AM, SSB, FM or PM.
- The modulated signals are then added together to form a complex signal which is transmitted over a single channel.

composite

37

Draw and ~~exp~~ explain block diagram of FDM Transmitter

Advantages of FM: (1) A large number of signals (channels) can be transmi...

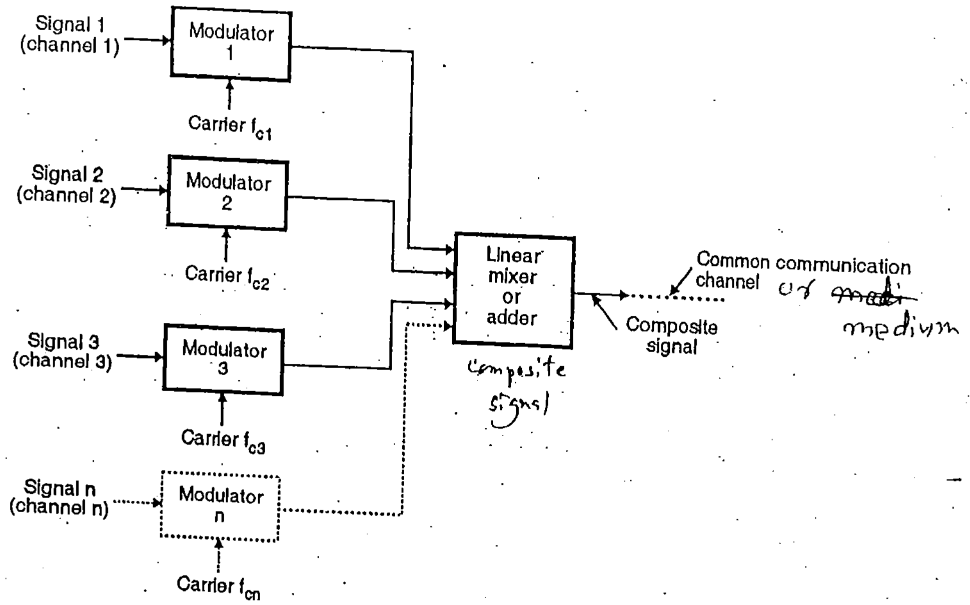


Fig. 4.11.1(b) : The FDM transmitter

Operation of the FDM transmitter

- Each signal modulates a separate carrier. The modulator outputs will contain the sidebands of the corresponding signals.
- The modulator outputs are added together in a linear mixer or adder. The linear mixer is different from the normal mixers. Here the sum and difference frequency components are not produced. But only the algebraic addition of the modulated outputs will take place.
- Different signals are thus added together in the time domain but they have a separate identity in the frequency domain. This is as shown in the Fig. 4.11.1(a).
- The composite signal at the output of mixer is transmitted over the single communication channel as shown in Fig. 4.11.1(b). This signal can be used to modulate a radio transmitter if the FDM signal is to be transmitted through air.

4.11.2 FDM Receiver :

- The block diagram of an FDM receiver is as shown in Fig. 4.11.1(c). The composite signal is applied to a group of band pass filters (BPF).
- Each BPF has a center frequency corresponding to one of the carriers.
- The BPFs have an adequate bandwidth to pass all the channel information without any distortion.
- Each filter will pass only its channel and reject all the other channels.
- The channel demodulator then removes the carrier and recovers the original signal back. (from output of low pass filter)

Fig. 4.13.1 : FDM transmitter for the basic group

Advantages of FDM: (1) A large number of signals (channels) can be transmi

FDM does not need synchronization between its transmitter and receiver for proper operation.

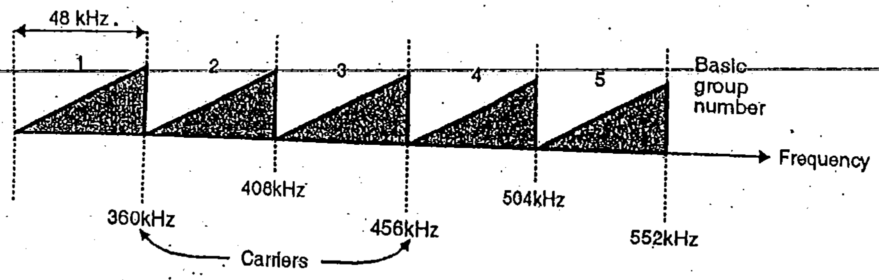


Fig. 4.12.3 : Frequency plan for a super group of FDM

4.13 Application of FDM in Telephone System :

- One of the important applications of the FDM system is the telephone system.
- Here each telephone signal is in the range of 300 to 3000 Hz. These voice channels modulate different subcarriers.
- These modulated subcarriers are then added together. As the number of telephone channels are very large, the multiplexing process is repeated at several levels.

4.13.1 FDM Telephone Transmitter :

The FDM transmitter for the "Basic Group", consisting of 12 voice channels is as shown in Fig. 4.13.1.

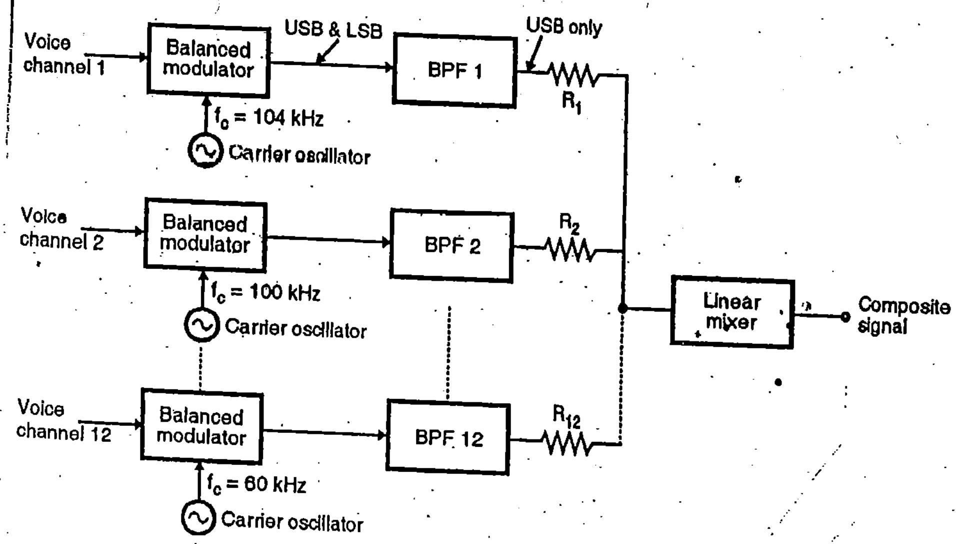


Fig. 4.13.1 : FDM transmitter for the basic group

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

4.14.2 Disadvantages of FDM :

Write the advantages and disadvantages of FDM.

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

4.14.3 Applications of FDM :

List the Applications of FDM.

Some of the important applications of FDM are :

1. Telephone systems.
2. AM (amplitude modulation) and FM (frequency modulation) radio broadcasting.
3. TV broadcasting
4. First generation of cellular phones used FDM. (5) Telemetry (6) satellite communication

Ex. 4.14.1 : Draw the FDM system to combine three voice channels. Each voice channel occupies a bandwidth of 4 kHz. The common voice channel has a bandwidth of 12 kHz from 100 kHz to 112 kHz.

Soln. : Fig. P. 4.14.1 shows the required FDM system.

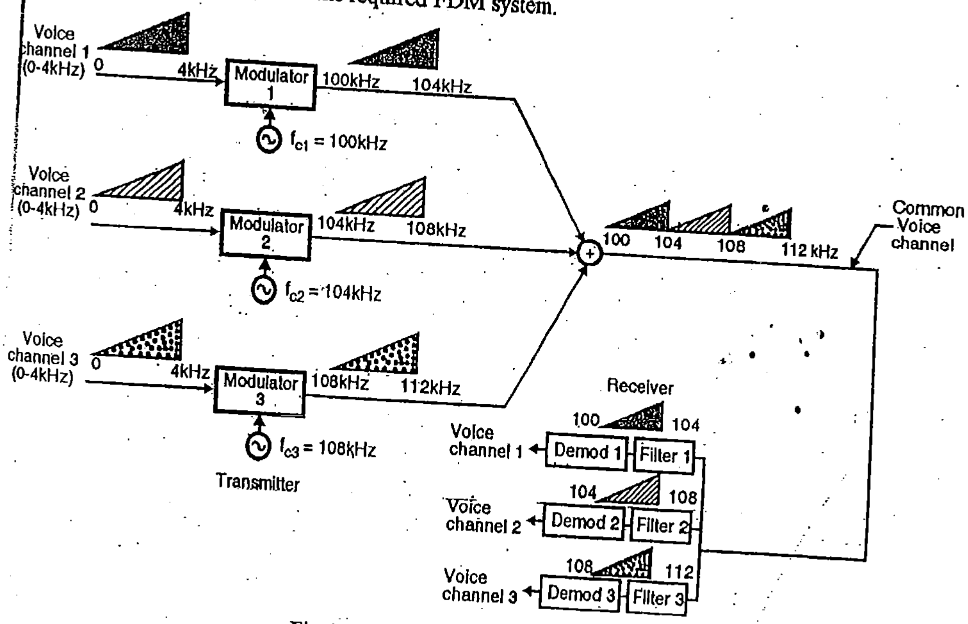


Fig. P. 4.14.1 : Required FDM system

Similar RFSK technique with

(2) In FDM, all signals are transmitted, simultaneously over the same communication medium and the signals occupy different frequency slots. But in TDM, the signals to be multiplexed are transmitted sequentially one after the other. (One by one in different time slots)

Each signal occupies a short time slot as shown in the figure 4-16-1. Thus the signals are isolated from each other in the time domain but all of them occupy the same slot in the frequency spectrum. Thus in TDM, the complete Bandwidth of communication channel is available to each signal being transmitted.

One frame: One frame corresponding to the time period required to transmit all the signals once on the transmission channel. (Shown in fig 4-16-1).

In FDM, all signals are

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4.16 ~~Synchronous~~ Time Division Multiplexing (TDM) :

- The process called multiplexing is used in order to utilize common transmission channel or medium to transmit more than one signals simultaneously.
- Two methods which are generally ~~employed~~ used are frequency division multiplexing (FDM) and time division multiplexing (TDM).
- In TDM all the signals to be transmitted are not transmitted simultaneously. Instead, they are transmitted one-by-one.
- Thus each signal will be transmitted for a very short time. One cycle or frame is said to be complete when all the signals are transmitted once on the transmission channel. The TDM principle is illustrated in Fig. 4.16.1.

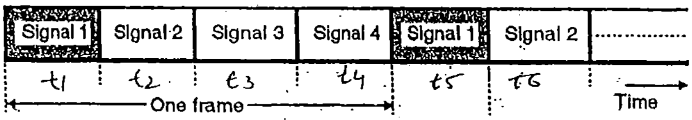

Fig. 4.16.1 : Principle of TDM

- As shown in the Fig. 4.16.1 one transmission of each channel completes one cycle of operation called as a "Frame".
- The TDM system can be used to multiplex analog or digital signals, however it is more suitable for the digital signal multiplexing.
- The concept of TDM will be more clear if you refer to Fig. 4.16.2.
- The data flow of each source (A, B or C) is divided into units (say A_1, A_2 or B_1, C_1 etc.)

  Then one unit from each source is taken and combined to form one frame. The size of each unit such as A_1, B_1 etc. can be 1 bit or several bits.

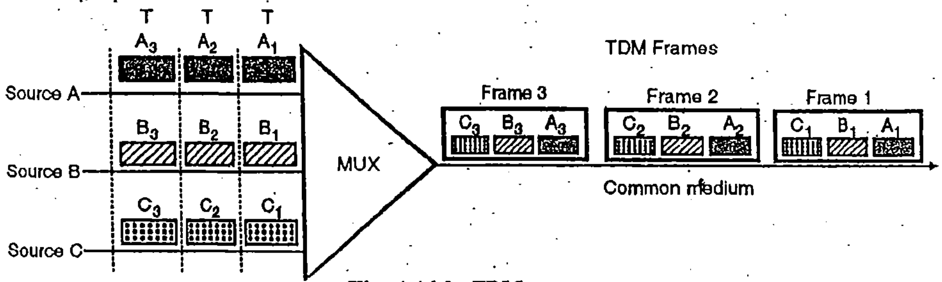

Fig. 4.16.2 : TDM system

- Fig. 4.16.3 shows the frames of TDM signal. For 3 inputs being multiplexed, a frame of TDM will consist of 3 units i.e. one unit from each source.
- Similarly for n number of inputs, each TDM frame will consist of n units.

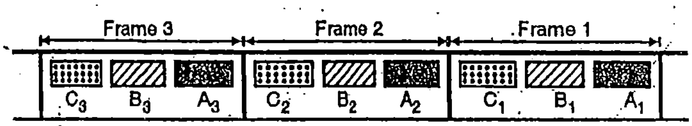

Fig. 4.16.3 : TDM frames

- The TDM signal in the form of frames is transmitted on the common communication medium.

(42)

Data rate :

Comment

- For a TDM, the data rate of the multiplexed signal is always n times the data rate of individual sources, where n is the number of sources.
- So if three sources are being multiplexed, then the data rate of the TDM signal is three times higher than the individual data rate.
- Naturally the duration of every unit (A_1 or B_1 etc.) in TDM signal is n times shorter than the unit duration before multiplexing.

4.16.1 PAM - TDM System :

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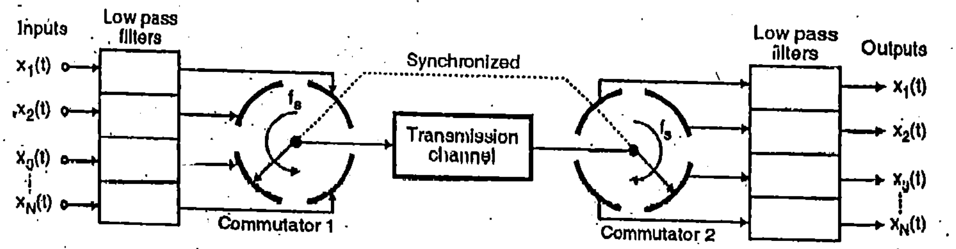


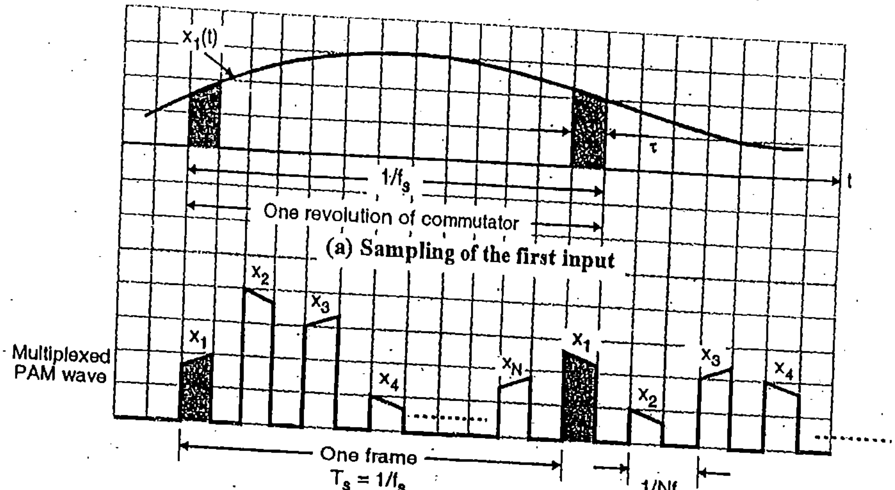
Fig. 4.16.4 : PAM/TDM system

- The TDM system which is going to be discussed now, combines the concepts of PAM and TDM both. The TDM system is as shown in Fig. 4.16.4.

The operation of the system is as follows :

- The multiplexer here is a single pole rotating switch or commutator. It can be a mechanical switch or an electronic switch. It rotates at f_s rotations per second.
- As the switch arm rotates, it is going to make contact with the position 1, 2, 3 or N for a short time. To these contacts are connected the N analog signals which are to be multiplexed.
- Thus the switch arm will connect these N input signals one by one to the communication channel.
- The waveform of a TDM signal which is being transmitted is as shown in Fig. 4.16.5. It shows that the rotary switch samples each channel during each of its rotations. Each rotation corresponds to one frame. Hence 1 frame is completed in T_s seconds where $T_s = 1/f_s$.
- At the receiver, there is one more rotating switch or commutator used for demultiplexing.
- It is important to note that this switch must rotate at the same speed as that of the commutator 1 at the transmitter and its position must be synchronized with commutator 1 in order to ensure proper demultiplexing.
- The same principle of multiplexing can be used for multiplexing more number of signals.

As discussed earlier the transmission rate of TDM is n times higher than the bit rate of each source.



(a) Sampling of the first input

(b) Multiplexed PAM signal, transmitted on the transmission channel

Fig. 4.16.5

Ex. 4.16.1 : 3 signals having a data rate of 2 kbps are grouped together by means of time division multiplexing. Each unit consists of 1 bit. Calculate :

1. The bit duration before multiplexing.
2. The transmission rate of TDM.
3. The duration of each time slot in TDM.
4. The duration of one TDM frame.

Soln. :

Step 1 : Duration of a bit before multiplexing :

- Each signal has a data rate of 2 kbps. That means 2000 bits per second.
- Hence the duration of each bit is,

$$T_b = \frac{1}{2000} = 0.5 \text{ ms}$$ ...Ans.

Step 2 : Transmission rate of TDM :

- The TDM frame is shown in Fig. P. 4.16.1.

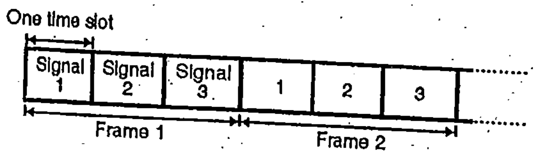


Fig. P. 4.16.1 : TDM frames

- As discussed earlier the transmission rate of TDM is n times higher than the bit rate of each source.

(44)

Transmission rate of TDM $= n \times 2000 = 3 \times 2000$

$= 6000$ bps or 6 kbps ...Ans.

Step 3 : Duration of time slot in TDM :

Duration of each time slot in TDM $= \frac{1}{6000} = 166.67$ μsec ...Ans.

Step 4 : Frame duration :

Duration of 1 frame $= n \times$ duration of one slot

$= 3 \times 166.67\ \mu s = 0.5$ ms ...Ans.

Note : The duration of a TDM frame is always equal to the duration of one unit before multiplexing.

Ex. 4.16.2 : Three channels are to be multiplexed using TDM technique. The rate of each channel is 150 bytes per second. In TDM, one byte per channel is to be multiplexed.

Calculate : 1. Frame size 2. Frame duration

3. Frame rate and 4. Bit rate of the TDM signal.

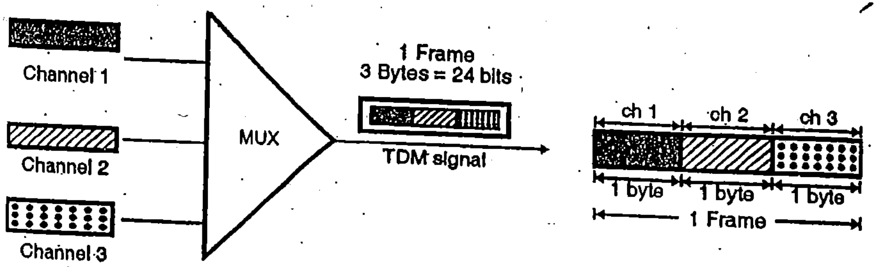
Soln. :

Given : Bit rate of each channel 150 bytes per sec, $n = 3$.

1 byte per channel to be multiplexed.

Step 1 : Draw the system block diagram :

Fig. P. 4.16.2(a) shows the block diagram of the TDM system and Fig. P. 4.16.2(b) shows one TDM frame.



(a) TDM system **(b) One frame of TDM**

Fig. P. 4.16.2

Step 2 : Frame size :

Each frame consists of one byte from each channel. So frame size is 3 bytes or 24 bits.

Step 3 : Frame duration and frame rate :

- The duration of a TDM frame is always equal to the duration of one unit before multiplexing.
- Here one unit before multiplexing is 1 byte i.e. 8 bits.

∴ Frame duration = 1 byte duration

Explain RFSK technique with

But each channel transmits at 150 bytes/sec.

$\therefore$ frame duration $= \frac{1}{150} = 6.666$ msec

frame rate $= \frac{1}{\text{frame duration}} = \frac{1}{6.666 \times 10^{-3}} = 150$ frames/sec

Bit rate of TDM signal = No. of bits per frame × No. of frames per second

$= 24 \times 150 = 3600$ bps.

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sig

(46)

State advantages and Disadvantages of TDM

150 bytes/sec

6 msec

= 150 frames/sec

bits per frame X No. of frames per second

Synchronization in digital TDM system :

- In digital TDM, the inputs are digital bit streams. All the digital pulses are of same amplitudes. So the synchronizing techniques for TDM-PAM system cannot be used here.
- If the synchronization is lost then a bit belonging to one channel may be received by a wrong channel.

Synchronization bit :

- In order to establish synchronization between the transmitter and receiver, one synchronization bit is added at the beginning of each TDM frame as shown in Fig. 4.16.8.

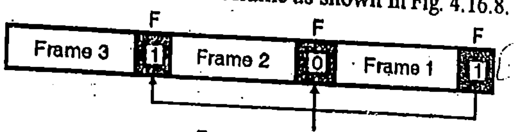


Fig. 4.16.8 : Frame synchronization in TDM

- These bits are called frame synchronizing bits or simply framing bits.
- The framing bits will follow a pattern frame to frame. For example the pattern shown in Fig. 4.16.8 is 101.
- The framing bit pattern will allow the demux to synchronize itself to the mux.

4.16.5 Advantages of TDM :

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

4.16.6 Disadvantages of TDM :

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

4.16.7 Bit Padding :

- Till now we have assumed that the data rate of all the channels is the same.
- But practically it won't be so. We will have to multiplex channels having different data rates.
- The data rates are not integer multiples of each other. But in order to multiplex them using TDM, they need to be integer multiples of each other.
- This can be achieved by a technique called **bit padding**.
- In the bit padding technique the multiplexer adds extra bits to the bit stream of a source so as to force the integer relationship between all the sources to be multiplexed.
- For example if the bit rate of one source is 3.5 times the bit rate of the other source then by using the bit padding, we can make it 4 times the bit rate of the other.
- These extra bits however do not contain any information. So they are discarded by the demultiplexer.

signals.

Ex. 4.16.7 : Sketch a channel interleaving scheme for time division multiplexing the following PAM signals : Five 4 kHz telephone channels and one 20 kHz music channel. Find the pulse repetition rate of the multiplexed signal and estimate the minimum system bandwidth required.

Soln. :

Each telephone channel of bandwidth 4 kHz must be sampled at Nyquist rate i.e. 2×4 kHz = 8 kHz. using a TDM commutator. The 20 kHz music channel must be sampled at 40 kHz (Nyquist rate) hence a separate sampler is required. The sampled signals are applied to two 4-bit A-D converters to obtain the equivalent digital signals. These signals are finally multiplexed using a multiplexer as shown in Fig. P. 4.16.7.

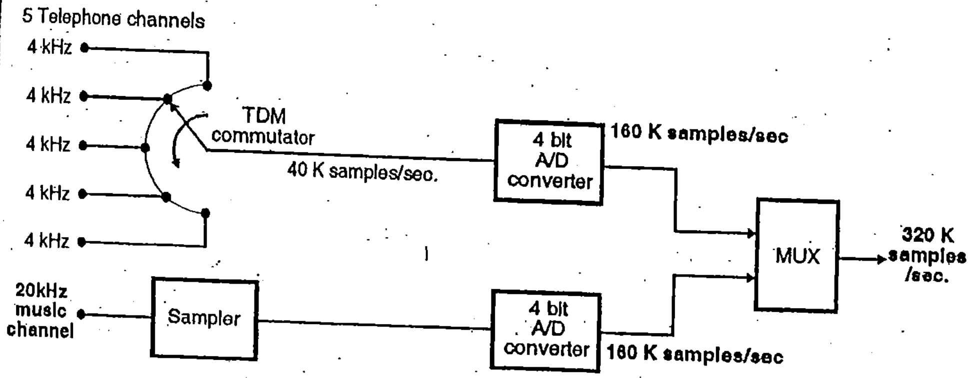


Fig. P. 4.16.7 : PAM-TDM system for Ex. 4.16.7

The TDM commutator output has a pulse repetition rate of 40K samples/sec as there are 5 channel and sampling rate is 8 kHz. Similarly the output of the separate sampler has a pulse repetition rate of 40 K samples/sec. The outputs of A-D converters have pulse repetition rates of $40 \times 4 = 160$ K samples/sec. Therefore pulse repetition rate at the output of a multiplexer is $160 + 160 = 320$ K samples/sec.

$\therefore$ Pulse repetition rate of the system = 320 kHz.

$\therefore$ Bandwidth required = bit rate = 320 kHz ....Ans.

4.16.11 Comparison of FDM and TDM Systems :

Sr. No.	FDM	TDM
1.	The signals which are to be multiplexed are added in the time domain. But they occupy different slots in the frequency domain.	The signals which are to be multiplexed can occupy the entire bandwidth but they are isolated in the time domain.
2.	FDM is usually preferred for the analog signals.	TDM is preferred for the digital signals.

~~Similar RFSK technique with~~

FDM	TDM
① In this technique, to transmit several messages (signals) on one channel, message signals are distributed in frequency spectrum such that they do not overlap [i.e. many signals are transmitted simultaneously with each signal occupying a different frequency slot within a common bandwidth.	① It is a technique for transmitting several messages (signals) on one channel by dividing time domain slots. One slot for each message. [i.e. the signals to be ~~transmitted~~ multiplexed are transmitted sequentially one after the other, in different time slots
② FDM requires modulator, filters and demodulators.	② TDM requires commutator at the transmitting end and a distributor working in perfect synchronization with commutator at the receiving end.
① In FDM all signals are transmitted simultaneously over same communication medium (channel). The signals are occupying different frequency slots	① In TDM, the signals are transmitted one by one, in different time slots.

Compare FDM and TDM. (4m)

- The codewords shown in Fig. 1.2.2 are three bit numbers. It is possible to introduce one more bit to indicate the "sign."

Error :

... quantization process the A to D conversion

Sr. No.	FDM	TDM
3.	Synchronization is not required.	Synchronization is required.
4.	The FDM requires a complex circuitry at the transmitter and receiver.	TDM circuitry is not very complex. It does not require complex circuitry
5.	FDM suffers from the problem of crosstalk due to imperfect band pass filters.	In TDM the problem of crosstalk is not severe.
6.	Due to wideband fading in the transmission medium, all the FDM channels are affected.	Due to fading only a few TDM channels will be affected.
7.	Due to slow narrowband fading taking place in the transmission channel only a single channel may be affected in FDM.	Due to slow narrowband fading all the TDM channels may get wiped out.

4.17 TDM Hierarchy :

- The telephone companies implement TDM (time division multiplexing) through the hierarchy of digital signals. This is called as digital signal (DS) service.
- Fig. 4.17.1 shows the DS hierarchy and the bit rates supported by various levels.

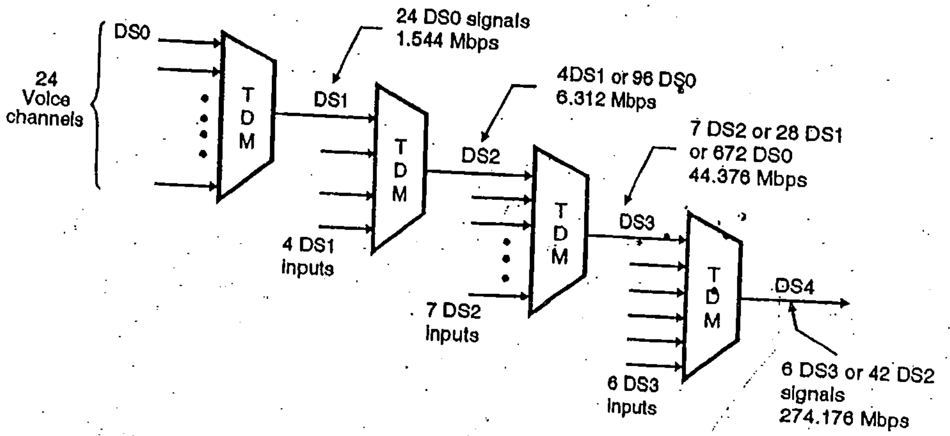


Fig. 4.17.1 : D.S. hierarchy

Receiver

Fig. 1.3.1 : Digital communication system

...ns
...a technique for
...g several messages on
...el by dividing time (signals)
...lots. One slot for
...essage. [i.e. the signals ~~transmitted~~ multiplexed
...mitted sequentially
...after the other, in
...ferent time slots

...m requires commutator,
... transmitting end and
...butor working in perfect
...nization with commutator
... receiving end.
In TDM the signals are transmitted one by one, in different time slots.

(50)

- The codewords shown in Fig. 1.2.2 are three bit numbers. It is possible to introduce one more bit to indicate the "sign."

technique for

Error :

- Due to the approximation taking place in the quantization process, the A to D conversion introduces some error in the digital signal.
- Such errors are non-reversible and it is not possible to produce an exact replica of the original analog signal at the receiver.
- It is possible to minimize these errors by selecting a proper sampling rate and, number of quantization levels.

1.2.3 Advantages of Digital Representation of a Signal :

The digital representation of a signal has following advantages :

1. Immunity to transmission noise and interference.
2. Regeneration of the coded signal along the transmission path is possible (Repeaters can be used).
3. Communication can be kept "private" and "secured" through the use of encryption.
4. The possibility of uniform format for different kinds of baseband signals.
5. It is possible to store the signal and process it further.
6. Techniques such as data compression and image enhancement can be used.

1.2.4 Disadvantages :

The advantages mentioned above are obtained at the expense of following disadvantages :

1. Transmission bandwidth is increased to a great extent.
2. System complexity is increased.

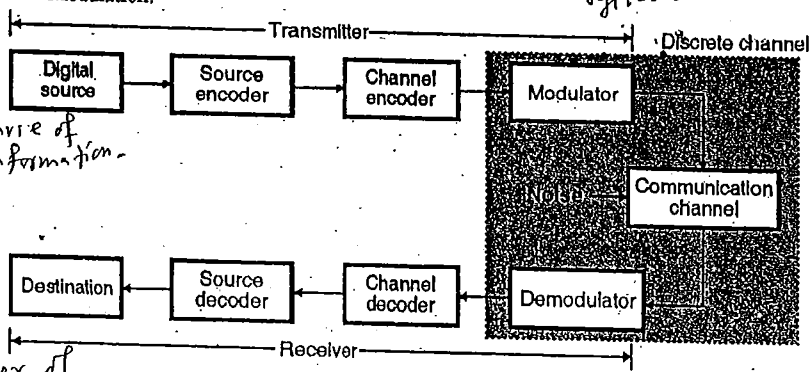
1.3 Basic Signal Processing Operations in Digital Communications :

Two of the most commonly used digital communication systems are PCM (Pulse Code Modulation) and DM (Delta Modulation).

Fig. 1.3.1 shows the block diagram of a digital communication system. In this diagram three basic signal processing operations have been included. They are :

With the help of block diagram, explain Digital communication system

- Source coding
- Channel coding and
- Modulation.



Source of information.

User of information

Fig. 1.3.1 : Digital communication system

- The source of information is assumed to be digital. If it is analog then it must be converted first to digital.

Source coding :

In source coding the encoder converts the digital signal generated at the source output into another signal in digital form. Source encoding is used to reduce or eliminate redundancy for ensuring an efficient representation of the source output. Different source coding techniques are PCM, DM, ADM etc.

- The conversion of signal from one form to the other is called as mapping. Such a mapping is usually one as to one.
- Due to elimination of redundancy the source coding provides an efficient representation of the source output.

Source decoder :

- Source decoder is at the receiver and it behaves exactly in an inverse way to the source encoder.
- It delivers the destination (user) the original digital source output.

Main advantage of using the source coding is that it reduces the bandwidth requirement.

Channel coding :

- Channel encoding is done to minimize the effect of channel noise.
- This will reduce the number of errors in the received data and will make the system more reliable. Channel coding technique introduces some redundancy.
- The channel encoder maps the incoming digital signal into a channel input.

Channel decoder :

- The channel decoder is at the receiver and it maps the channel output into a digital signal in such a way that effect of channel noise is reduced to a minimum.
- Thus channel encoder and decoder together provide a reliable communication over a noisy channel. This is achieved by introducing redundancy (parity bits) in a prescribed form, at the transmitter.
- The output of the channel encoder is a series of codewords which include the message and some parity bits. These additional parity bits introduce redundancy.
- The channel decoder converts these codewords into digital messages.

Thus in source coding the redundancy is removed whereas in channel coding the redundancy is introduced in a controlled manner.

- The source encoding alone or channel encoding alone can be performed. It is not essential to perform both but in many systems both these are performed together.
- It is possible to change the sequence in which channel encoding and source encoding are being performed.
- Channel and source encoding improve the system performance at the expense of increased circuit complexity.

Modulation :

- Modulation is used for providing an efficient transmission of the signal over the channel. The modulator can use any of the CW digital modulation techniques such as ASK (amplitude shift keyings), FSK (frequency shift keying) or PSK (phase shift keying).
- The demodulator is used for demodulation.

(52)

X

Discrete channel :

- As shown by a dotted box in ~~Fig. 1.3.1, the~~ discrete channel consists of modulator, channel and detector.
- It is called as discrete channel because its input as well as output are in the discrete form.
- In the traditional systems the modulation and coding are performed separately. But this increases the bandwidth requirement.
- Hence in some application these two operations are performed simultaneously.

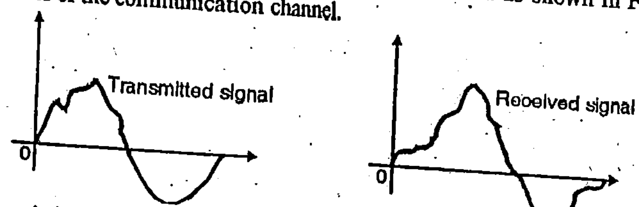
1.3.1 Why Digital Communication ?

- In this section we will compare the performance of analog communication systems with that of digital communication systems based on the following factors :

1. Effect of noise 2. Distance to be covered 3. Services provided.

1.3.2 Comparison of Analog and Digital Transmission : Effect of Noise :

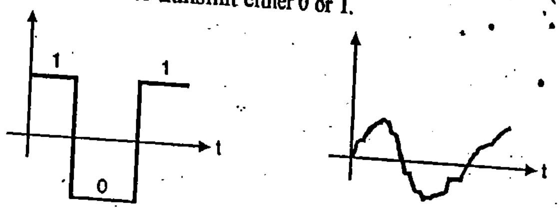
- In the analog communication, the objective is to transmit a waveform as shown in Fig. 1.3.2(a).
- But in doing so the shape of transmitted signal gets distorted as shown in Fig. 1.3.2(b) due to non-adequacies of the communication channel.



(a) Transmitted signal in analog communication (b) Received signal in the analog communication

Fig. 1.3.2

- Since all the information is contained in the shape of the waveform, it is necessary to preserve the shape of waveform.
- But the noise will distort the waveform as shown in Fig. 1.3.2(b). Therefore the received information will also be distorted.
- Now consider the transmission of digital signal as shown in Fig. 1.3.2(c). To transmit such a binary signal the objective is to transmit either 0 or 1.



(c) Transmitted signal in digital transmission (d) Received signal in digital transmission

Fig. 1.3.2

- (1) ~~Therefore digital transmission~~ Digital communication is more immune to noise as compared to the analog communication.

1.3.3 (2) Digital Communication is Suitable for Long Distances :

- The digital communication becomes cost effective over analog communication if the distance involved is large.
- Consider a long distance communication link shown in Fig. 1.3.3.
- As the signal from the source travels a distance from source, its amplitude will reduce due to attenuation. In addition to that, the signal will become increasingly distorted. The noise from various sources gets added to the signal.

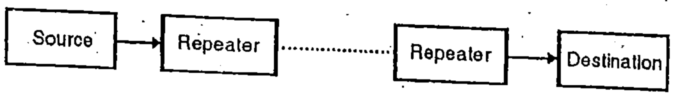


Fig. 1.3.3 : A long distance communication link

- In digital communication system repeaters are introduced as shown in Fig. 1.3.3 between the destination and source.
- The repeaters are basically regenerators of the signals. They receive the (signal + noise) at their input, separate out the signal from noise and regenerate the signal which is free from noise.
- Due to the use of repeaters the noise performance of digital communication systems is much better than that of the analog systems. Note that repeaters cannot be used for the analog communication system.
- The operation of a digital repeater becomes clear from Fig. 1.3.4.

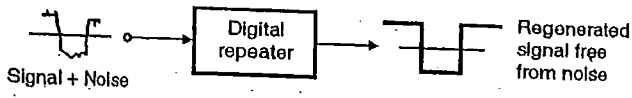


Fig. 1.3.4 : A digital repeater

1.3.4 Digital Networks can Handle Many Types of Services :

- The networks based on digital transmission are capable of handling any type of information which can be represented in the digital form.
- Hence such networks can handle many types of services.

1.3.5 Advantages of Digital Communication :

Some of the advantages of digital communication are as follows :

1. Due to the digital nature of the transmitted signal, the interference of additive noise does not introduce many errors. So digital communication has a better noise immunity.
2. Due to the channel coding techniques used in digital communication, it is possible to detect and correct the errors introduced during the data transmission.
3. Repeaters can be used between transmitter and receiver to regenerate the digital signal. This improves the noise immunity further.
4. Due to the digital nature of the signal, it is possible to use the advanced data processing techniques such as digital signal processing, image processing, data compression etc.
5. TDM (Time Division Multiplexing) technique can be used to transmit many voice channels over a single common transmission channel.

(54)

6. Digital communication is useful in military applications where only a few permitted receivers can receive the transmitted signal.
7. Digital communication is becoming simpler and cheaper as compared to the analog communication due to the invention of high speed computers and integrated circuits (ICs).

1.3.6 Disadvantages :

List the disadvantage of digital communication

Some of the important disadvantages of digital communication are :

(1) Large system Bandwidth
(2) System Synchronization

1. The bit rates of digital systems are high. Therefore they require a larger channel bandwidth as compared to analog systems.
2. Digital modulation needs synchronization in case of synchronous modulation, whereas the analog signals generally have no such requirement.

1.3.7 Comparison of Analog and Digital Modulation :

Compare analog and Digital communication systems. (2m)

Sr. No.	Analog modulation	Digital modulation
1.	Transmitted modulated signal is analog in nature.	Transmitted signal is digital i.e. train of digital pulses.
2.	Amplitude, frequency or phase variations in the transmitted signal represent the information or message.	Amplitude, width or position of the transmitted pulses is constant. The message is transmitted in the form of code words.
3.	Noise immunity is poor for AM, but improved for FM and PM.	Noise immunity is excellent.
4.	It is not possible to separate out noise and signal. Therefore repeaters can not be used.	It is possible to separate signal from noise. So repeaters can be used.
5.	Coding is not possible.	Coding techniques can be used to detect and correct the errors.
6.	Bandwidth required is lower than that for the digital modulation methods.	Due to higher bit rates, higher channel bandwidths is needed.
7.	FDM is used for multiplexing.	TDM is used for multiplexing.
8.	Not suitable for transmission of secret information in military application.	Due to coding techniques, it is suitable for military applications.
9.	Analog modulation systems are AM, FM, PM, PAM, PWM etc.	Digital modulation systems are PCM, DM, ADM, DPCM etc.

1.4 Channels for Digital Communications :

- The type of modulation and coding used in a digital communication system is decided by the channel characteristics and application areas.
- Some of the important characteristics of a channel are :
  1. Power required to achieve the desired S/N ratio.
  2. Bandwidth of the channel.
  3. Amplitude and phase response of channel.
  4. Type of channel (Linear or Nonlinear)
  5. Effects of external interference on the channel.

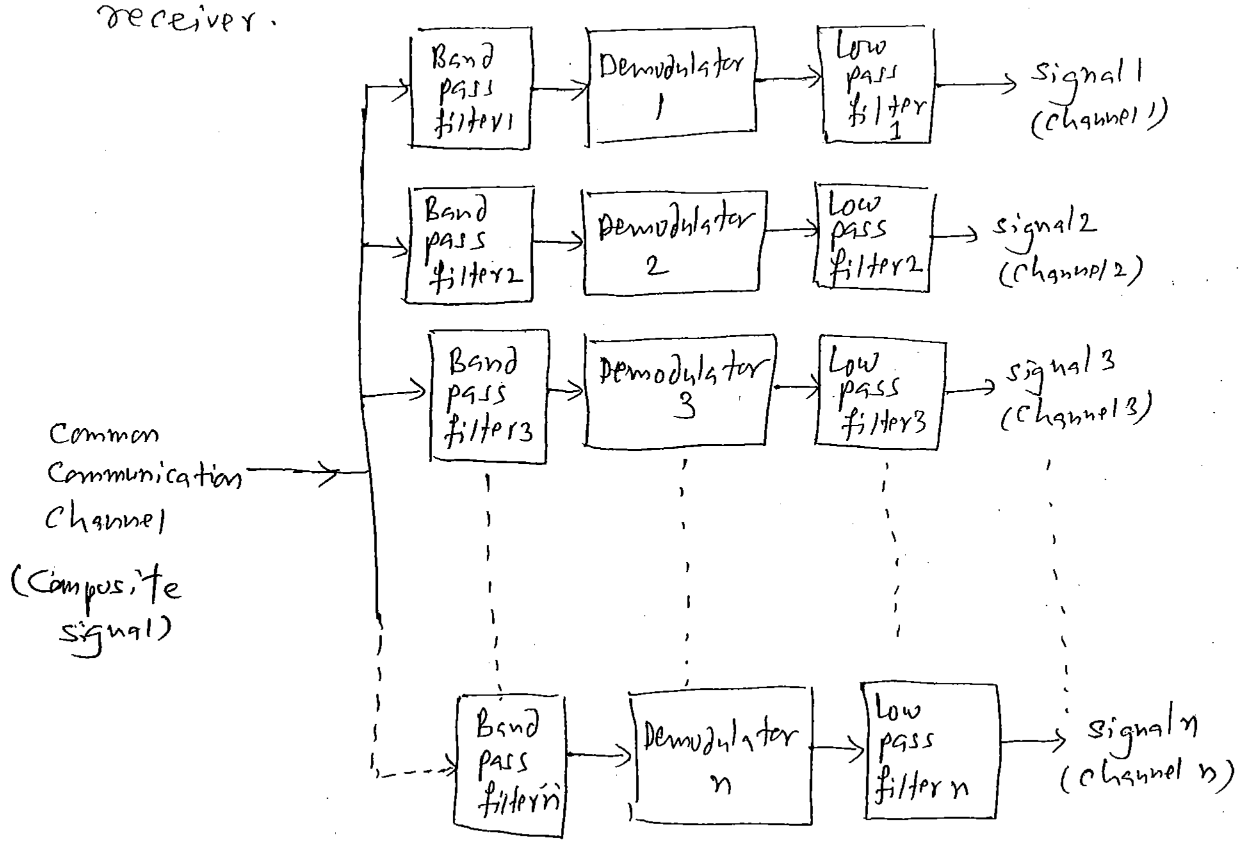
List the important characteristics of a channel.

24

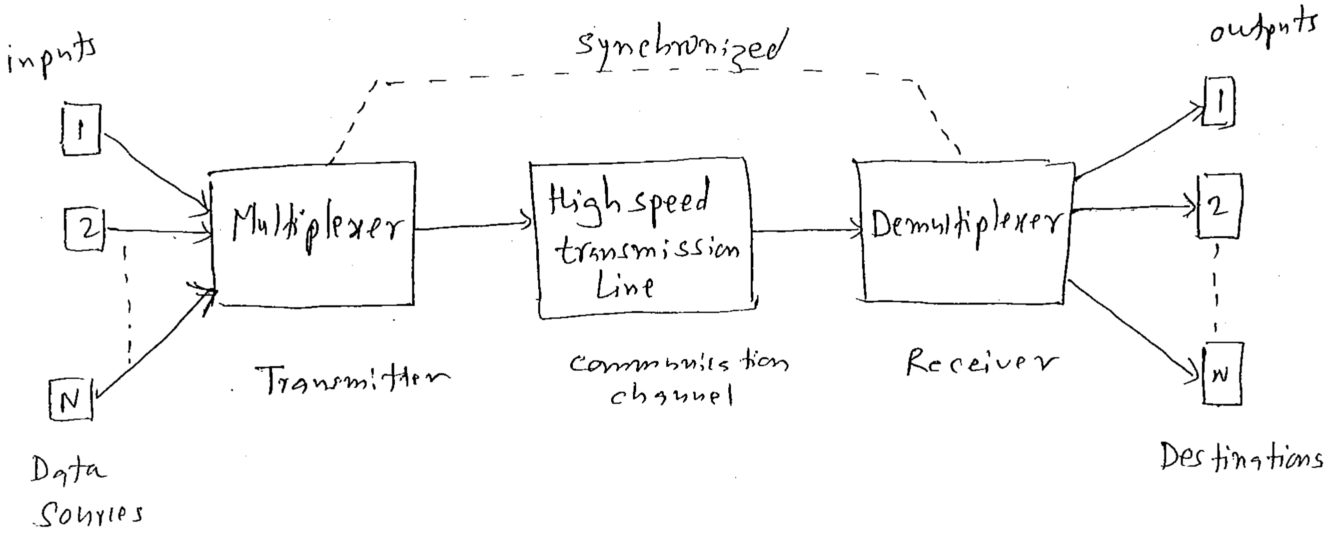
its working.

(55)

FDM receiver: Following figure the block diagram of FDM receiver.

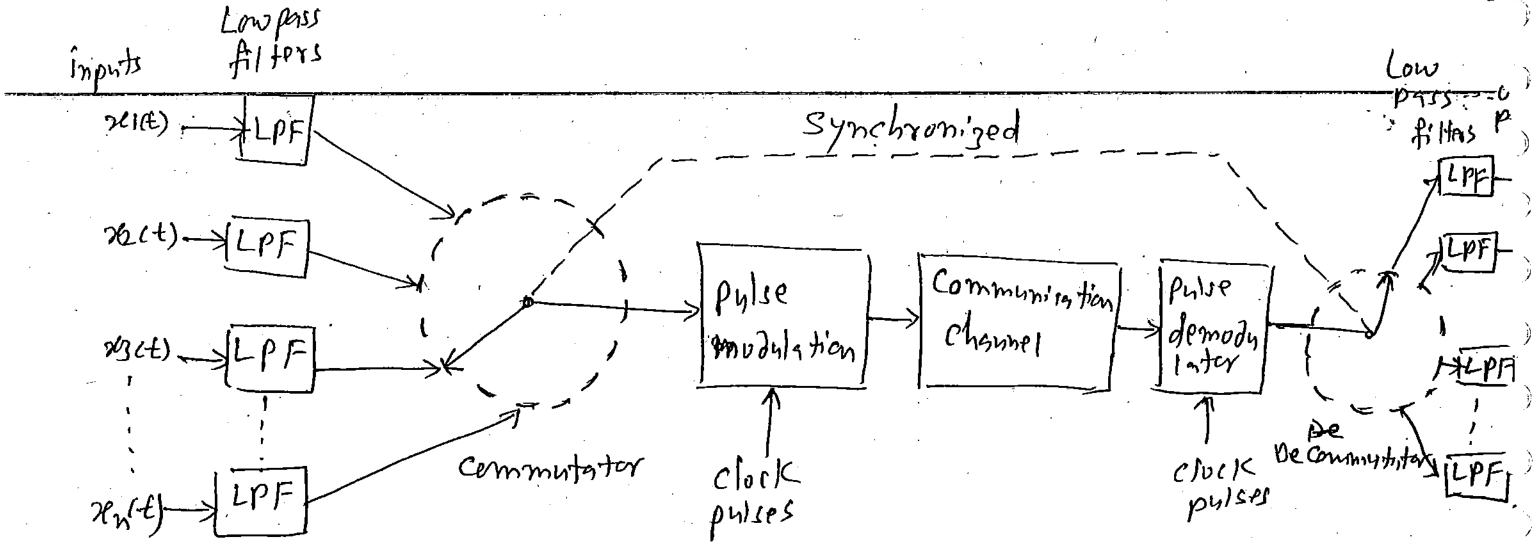


TDM: Following figure show a simple block diagram of TDM system.



(50) (56)

Following figure shows Block diagram of PAM TDM system



Block diagram of TDM system:-

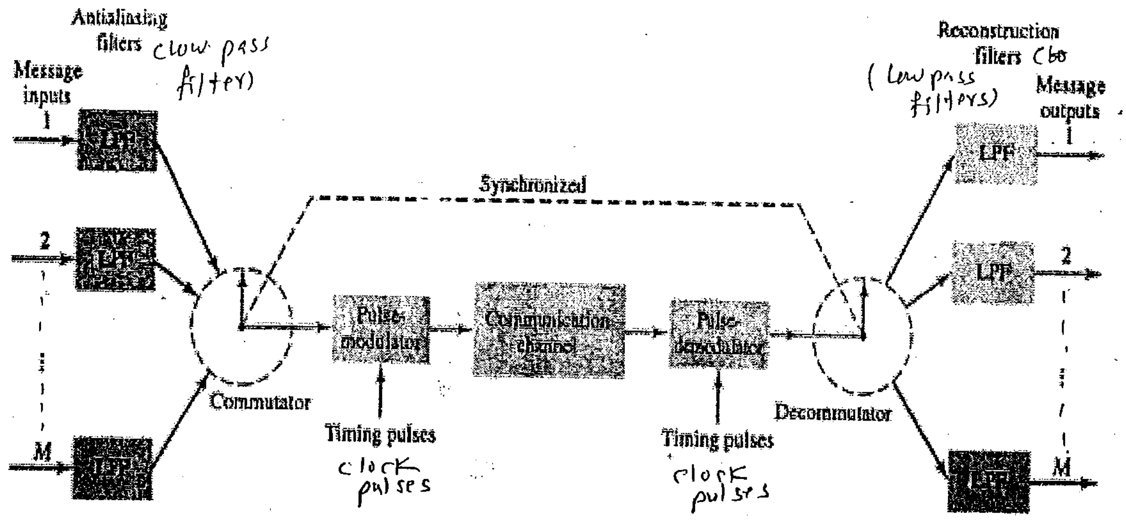
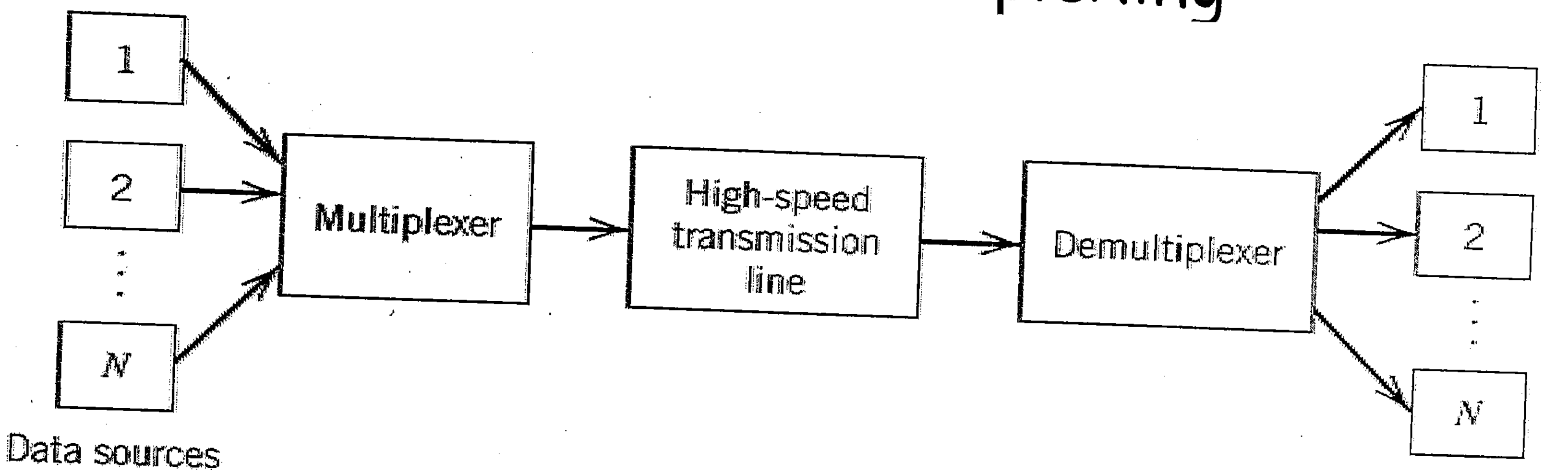


FIGURE 5.29 Block diagram of TDM system, showing the important constituents of the transmitter and receiver.

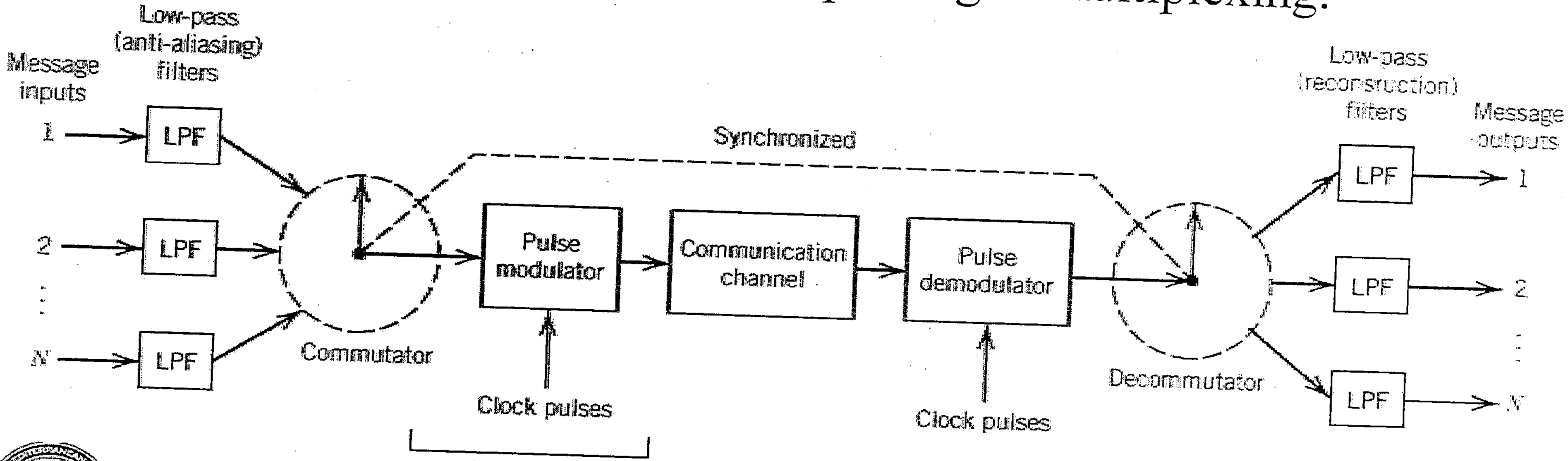
At transmitter ~~Here~~ multiplexer is a single pole rotating switch or commutator. It is a mechanical switch or an electronic switch. It is going to make contacts with the positions 1, 2, 3, ... or M for short time interval. To these contacts, M analog signals are connected, which are to be multiplexed. Thus the switch arm will connect these M input signals one by one to pulse modulator. and M pulse modulated signals are given one by one to the communication channel.

At the receiver, pulse demodulator is used extract original baseband signals, one by one. At the receiver, there is one more rotating switch or decommutator used for demultiplexing. It is important to note that this decommutator must rotate at the same speed as that of Commutator at the transmitter and its position must be synchronized with the commutator, in order to ensure proper demultiplexing.

Time Division Multiplexing



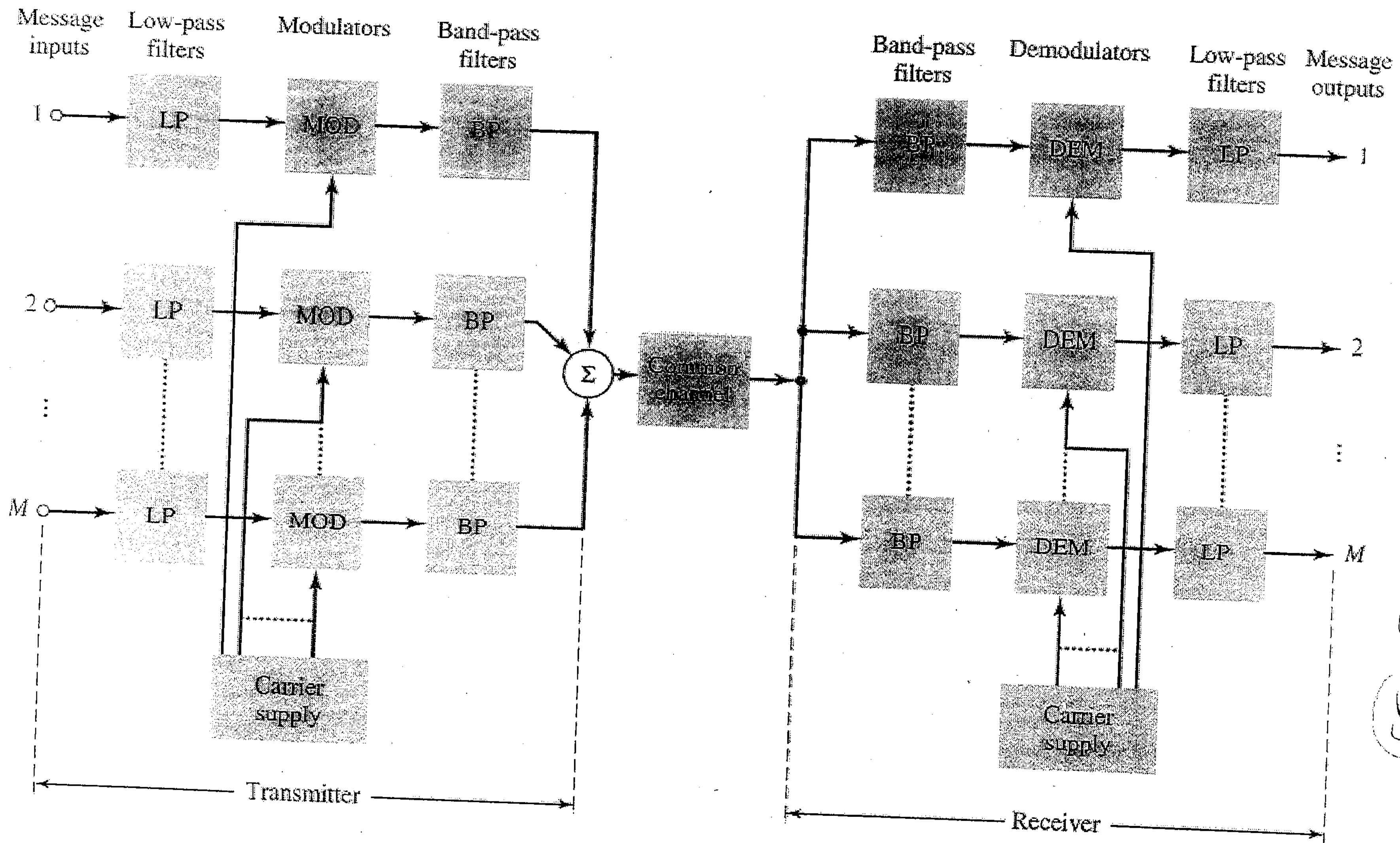
Conceptual diagram of multiplexing-demultiplexing.



PAM TDM System

Frequency Division Multiplexing (FDM)

➢ Block diagram of FDM system, showing the important constituents of the transmitter and receiver.



Draw block diagram of FDM system and explain working in detail (Ru.)

Vocoders

what is vocoder?

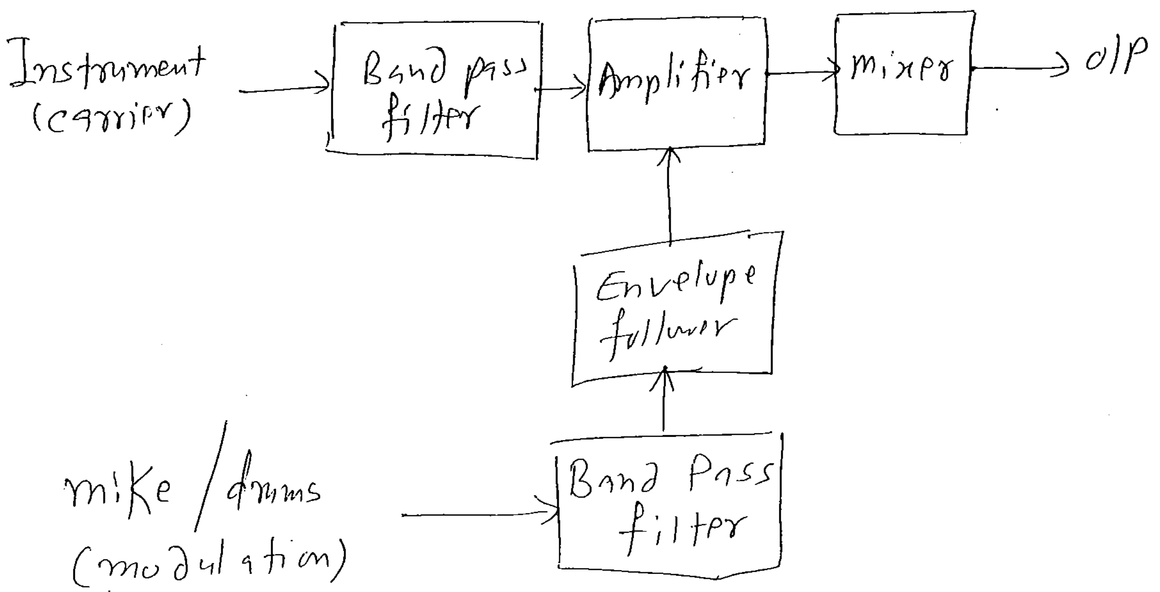
Vocoders stands for voice coders [or voice encoders] In vocoders synthetic sound is reproduced with artificial quality. Vocoders transmit signals with low bit rate, usually in the range of 1·2 to 2·4 KB.

A vocoder is an audio processor that captures the characteristics elements of an audio signal and then uses this characteristic signal to affect other audio signals.

Vocoders are used to synthesize speech.

A vocoder needs two inputs, to function properly. A carrier to vocode through and the modulator your voice. The modulator takes your voice, finds the fundamental frequencies (important bits) of it and converts them into levels of amplitude and passess through band pass filters.

A simple block diagram of a specific vocoder is given as follows:



Applications:

Write applications of vocoders. (1m)

1) Digital WLL
2) Voice Pagers
3) Digital voice Repeaters
4) Digital Trunking
5) messaging system.

Different vocoding systems

State different types of vocoding system

(1) Linear predictive coder (LPC)
(2) channel vocoder
(3) Formant vocoder
(4) Voice excited vocoder.

Explain any with block diagram any one vocoder system (4m)

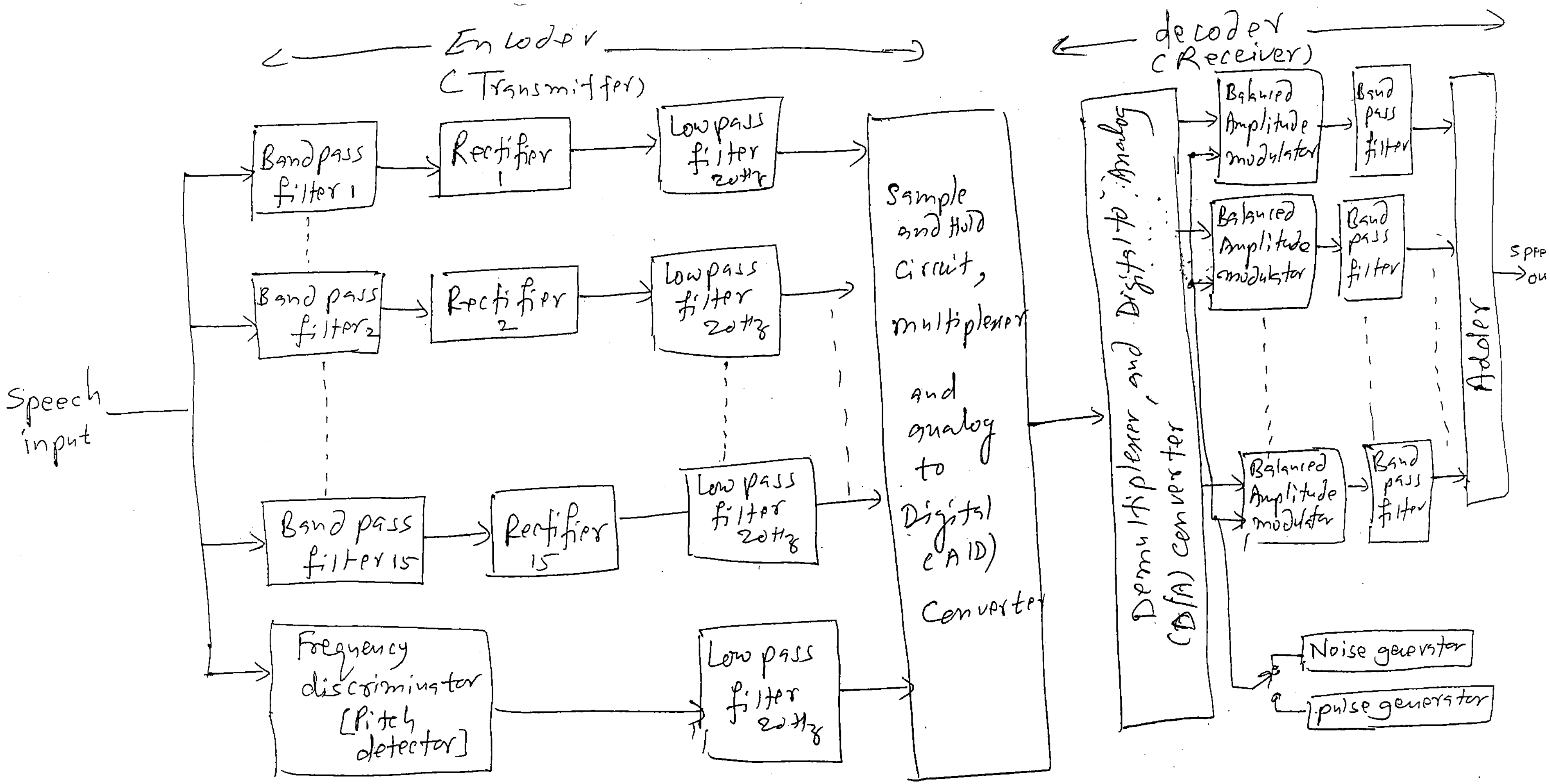
Channel Vocoder:

One of the vocoder systems is the channel vocoder. Following figure shows block diagram of channel vocoder.

In this encoding system, the spectrum of the input speech is divided into 15 frequency ranges, each of Bandwidth 200 Hz. (Using Bandpass filters).

The output of each Bandpass filter is followed by rectifier, Low pass filter of 20 Hz

The outputs of 16, 20 Hz low pass filters are sampled, multiplexed and A/D converted.


Encoder
(Transmitter)
decoder
(Receiver)
Speech input
Band pass filter 1
Band pass filter 2
Band pass filter 15
Rectifier 1
Rectifier 2
Rectifier 15
Low pass filter 20Hz
Low pass filter 20Hz
Low pass filter 20Hz
Frequency discriminator [Pitch detector]
Low pass filter 20Hz
Sample and Hold Circuit, multiplexer and analog to Digital (A/D) Converter
Demultiplexer, and Digital to Analog (D/A) converter
Balanced Amplitude modulator
Balanced Amplitude modulator
Balanced Amplitude modulator
Band pass filter
Band pass filter
Band pass filter
Noise generator
pulse generator
Adder
speech out

~~At~~ At ~~receiver~~ Transmitter (encoder) side, the input speech is additionally applied to a frequency discriminator followed by 20 Hz Low pass filter. When the signal is voiced, the output of filter provides a voltage which is proportional to the voice frequency. This frequency is the pitch of the voice. The discriminator-filter combination is a special, in that when the speech is unvoiced, the output of the filter is smaller voltage, than that for voiced speech. Using a detector, we can then determine, by noting the output of the discriminator-filter combination, whether the speech is voiced or unvoiced and if voiced, the voltage detected is determined by the pitch

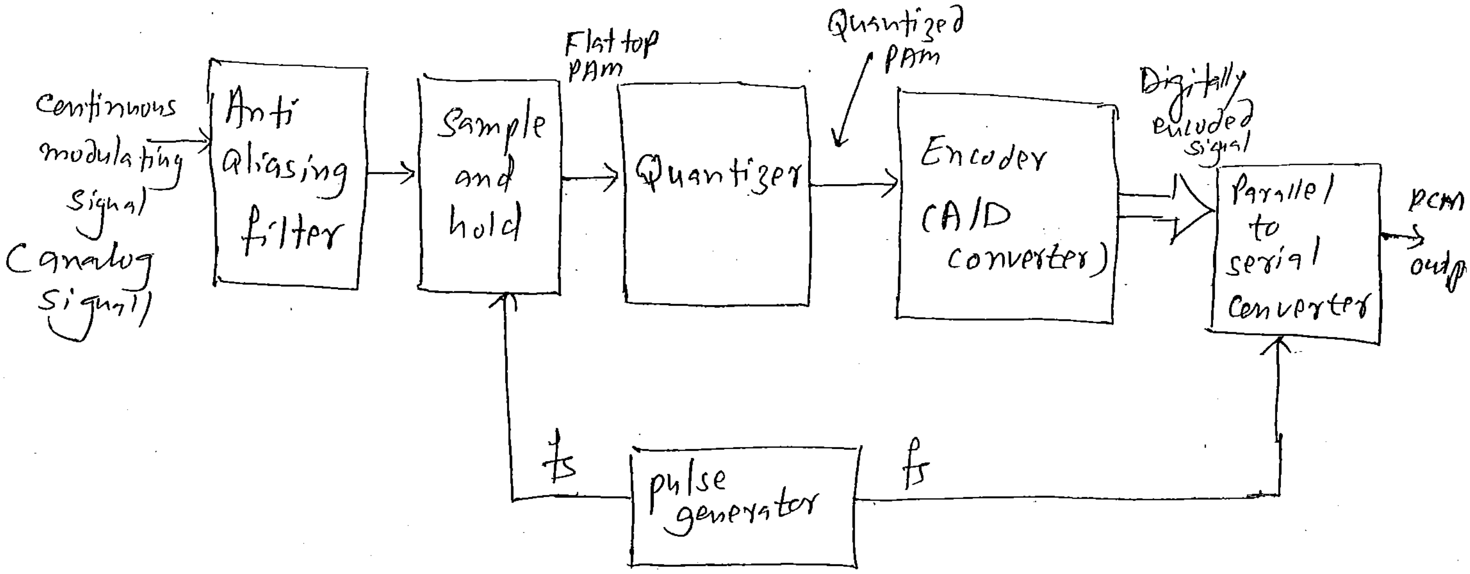
At the vocoder receiver the signal is demultiplexed and decoded i.e converted back to analog form. Corresponding to each filter-rectifier combination at the encoder, there is a balanced amplitude modulator and Bandpass filter at the decoder. The carrier input to each modulator is noise generator or pulse generator. The modulation input is the amplitude signal (of each of the 15 rectifier-filters) provided by the encoder.

A PCM system consists of a pcm encoder (transmitter) and a pcm decoder (receiver).

4 mark. Draw block diagram & explain working pcm transmitter (encoder)

The essential operations in the pcm transmitter are sampling, quantizing and ~~en~~ encodeing

Following figure shows block diagram of pcm Transmitter (encoder).



The analog signal is first passed through an antialiasing filter and then it is applied to sample and hold circuit, where it is sampled, ~~where~~ at sufficient high sampling rate. output of Sample and hold circuit is a flat topped PAM signal.

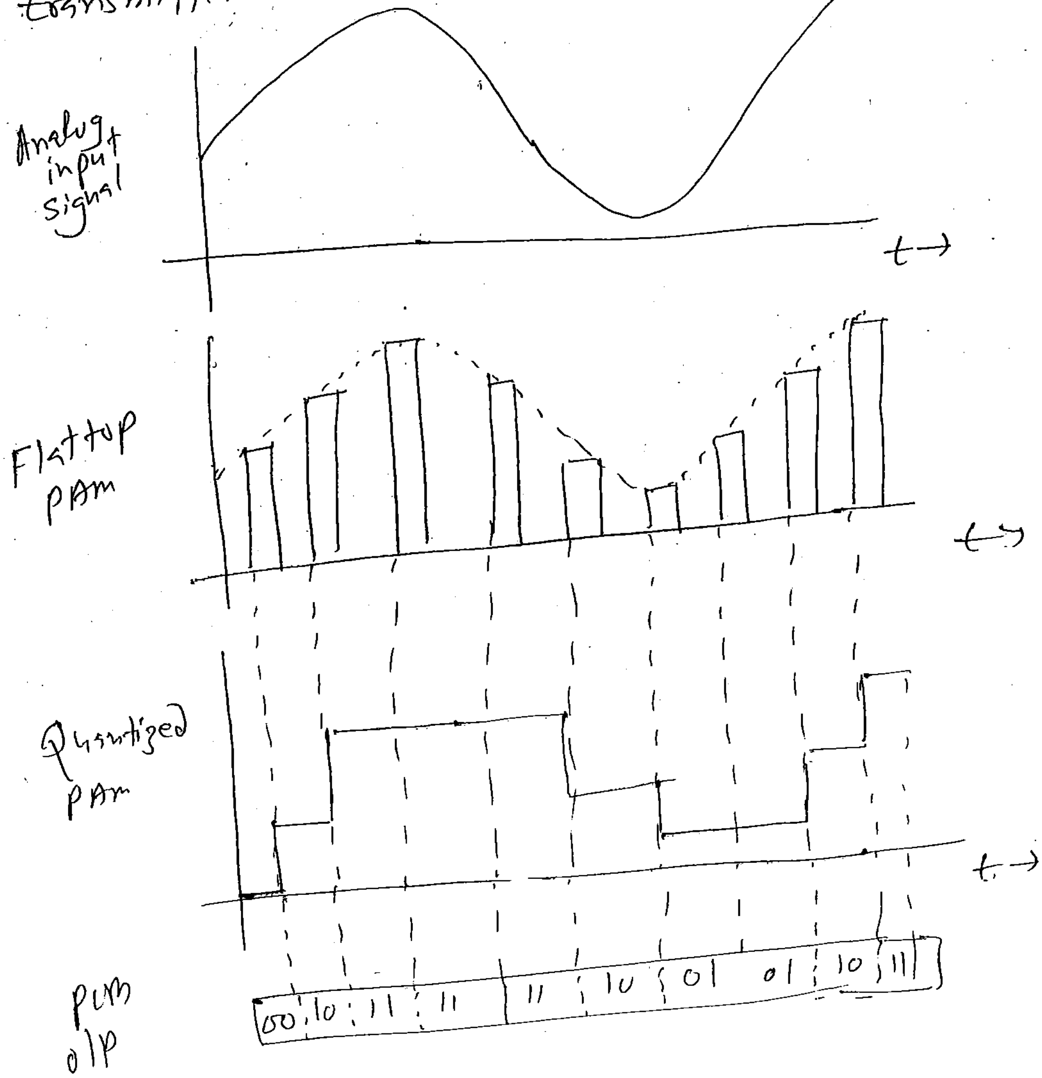
Then samples are given to ~~next~~ Quantizer ~~circu~~ circuit. The Quantizer converts the sampled signal into approximate voltage levels (i.e. coded form). The combined effect of sampling and quantization produces the quantized PAM signal at the output of quantizer.

The quantized PAM pulses are applied to encoder (which is an A/D converter). Each quantized level is converted into N bit digital word by A/D converter. The value of N can be 8, 16, 32, 64 etc

The encoder output is converted into stream of pulses by parallel to serial converter block.

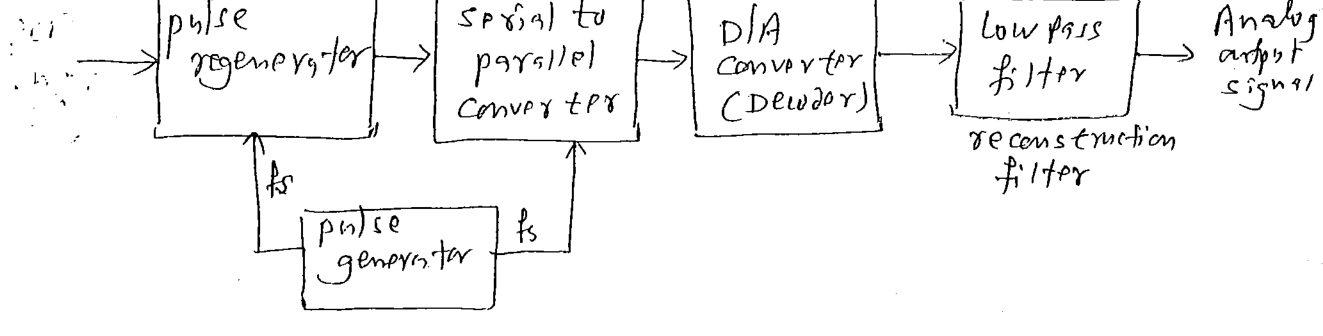
Thus at the pcm transmitter output, we get a train of digital pulses.

Following figure shows waveforms at different points of pcm transmitter



~~pulse~~ pcm Receiver (Decoder):-

Following figure shows block diagram of pcm Receiver.

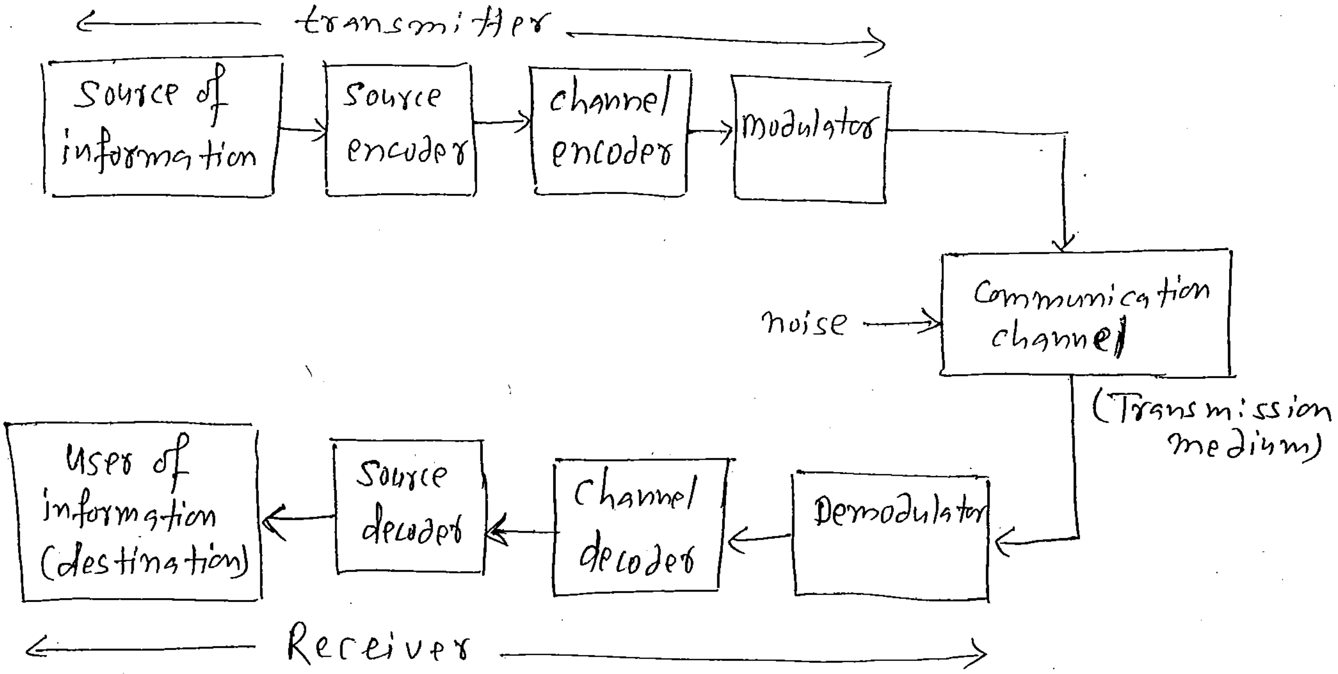


A PCM signal along with noise is available at the receiver input. The pulse regenerator circuit at the receiver ~~with~~ will separate the PCM pulses from noise. Then PCM signal passed through a serial to parallel converter. Then output of this block is applied to a Decoder. The decoder is a D/A converter, which performs exactly the opposite operation of encoder. The output of Decoder is the sequence of quantised pulses (quantised PAM signal). This quantized PAM signal is passed through a low pass filter (reconstruction filter) to recover the original analog signal.

Draw the block diagram of Digital comm. system and explain it . (6m)

(67)

Block diagram of Digital communication system:



Source of information: (a) Analog information sources
(b) Digital information sources.

The source of information is assumed to be digital. If it is analog then it must be converted first into digital.

Source Encoder:

The source Encoder converts the digital signal generated at the source output into another signal in digital form.

Source encoding is used to reduce or eliminate redundancy in the transmitting information, so that bandwidth required for transmission is minimized. Different source coding techniques are pcm, Dm, ADm etc.

channel Encoder:

channel encoding is used to minimize the effect of channel noise. This will reduce the number of errors in the received data and will make the system more reliable

Modulator:

The modulation is used to provide efficient ~~efficie~~ transmission of signal over the channel. The modulator can use any digital modular techniques such as ASK (amplitude shift keying), FSK (Frequency shift keying) or PSK (phase shift keying)

Communication channel: The communication channel or medium provides link between source (Transmitter) and destination (Receiver). The Transmission medium may be a telephone line, coaxial cable, twisted pair ~~ca~~ wire, optical fiber cable, Radio waves, microwave link or satellite link for long distance communication.

Demodulator:

The extraction of the information from modulator is obtained using Demodulator at receiver side.

Channel Decoder:

The channel Decoder at receiver recovers the the information, from the coded binary form (coded words). Error detection and possible correction is also performed by channel decoder.

Source decoder:

Source decoder is at the receiver and it operates exactly in ~~an~~ opposite way to the source encoder. It provides original digital source information to the destination (user)

Advantages of Digital communication:

(1) Digital Communication has better noise immunity. The effect of distortion, noise and interference is less in a digital communication system.

(2) Digital Circuits are more reliable and cheaper compared to analog Circuits.

(3) Signal processing functions like encryption, compression, can be used to maintain the secrecy of the information.

(4) Due to digital nature of the signal, it is possible to use the advanced data processing techniques such as digital signal processing, image processing, data compression etc.

(4) Due to channel coding techniques used for digital communication, it is possible to detect and correct the errors introduced during the data transmission.

(5) Repeaters can be used between transmitter and receiver, to ~~generate the~~ transmit the digital signal for a long distance.

(6) TDM (Time Division Multiplexing) technique can be used to transmitt many digital signals (channels) over a single Common transmission channel.

(7) Digital communication becomes simpler and cheaper as compared to analog communication, due to availability of high speed microprocessors, VLSI chips etc.

(8) Error detecting and Error Correcting codes improve the system performance by reducing the errors.

Dis advantages of Digital Communication:

(1) Large channel Bandwidth:
The Bit rates of digital systems are high, therefore they require a large bandwidth to communicate ~~the same information in a digital format as compared~~ to analog ~~format~~ system.

(2) System Complexity is increased.

(3) System synchronization is required for digital communication, whereas analog communication have no such requirement

Write advantages and disadvantages of Digital comm-system. (4m)