

communication system :

It is the process of transferring the information from one place to another place or the exchange of information b/w two points.

The major elements of communication systems are transmitter of information, channel (or) medium of communication & receiver of information.

Examples :

Television, internet, E-mails, Video calls, online chat, mobiles, public switched telephone network (PSTN), satellite communication, radar communication, optical fibre communication etc.

Applications of communication system :

1. Wireless system using RF & microwave
2. wireless LAN
3. Mobile phones
4. Antenna & transmitter test.
5. Base station monitoring
6. satellite communication
7. Bus testing & cable testing.

Advantages of communication system :

1. It has less time taken to transmit the information.
2. High speed
3. It is more efficient
4. It allows remote communication.

Disadvantages :

1. privacy issues
2. Deteriorating languages
3. Noise may occur
4. Distortion may occur.

amp of amplitude modulat

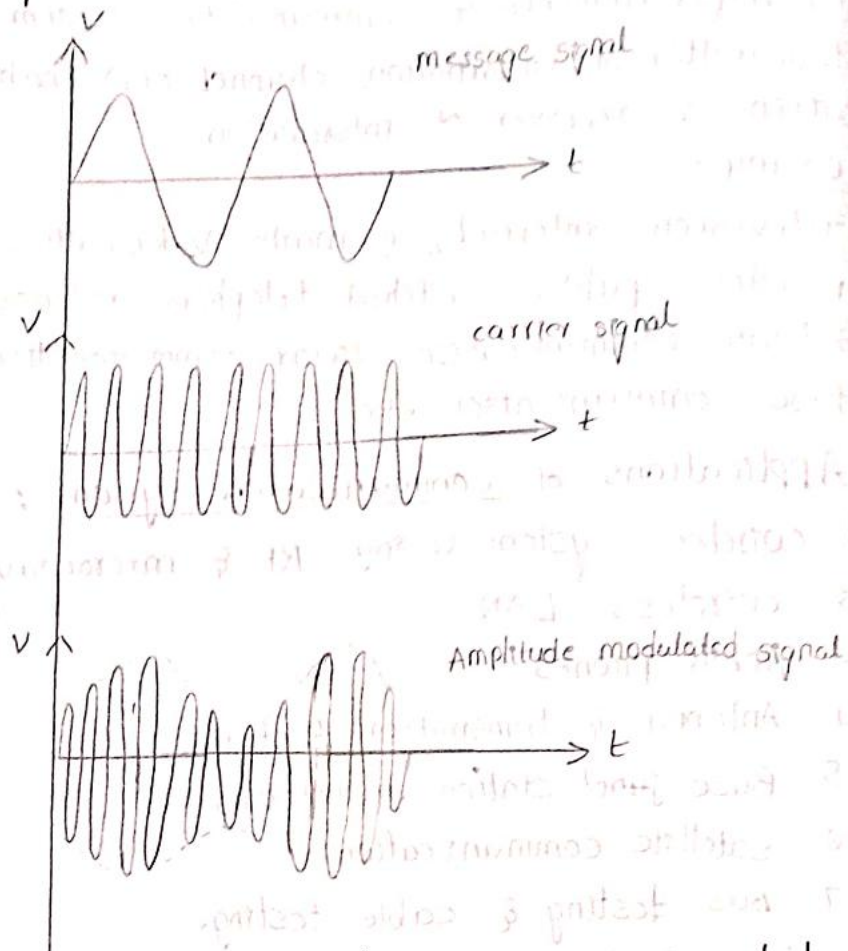
is amplitude of carrier signal + total message signal.

$$m(t) + c(t) = A$$

Amplitude modulation :

The process of changing amplitude of high frequency carrier signal in accordance with the instantaneous amplitude of message signal is called "amplitude modulation" (AM).

→ In AM the amplitude of the carrier changes but the frequency & phase remains constant.



Time domain equation for single tone carrier modulation for an AM signal.

Instantaneous Voltage of carrier signal is given by,

$$V_c = V_c \sin \omega_c t \rightarrow \textcircled{1}$$

Instantaneous Voltage of message signal is given by,

$$V_m = V_m \sin \omega_m t \rightarrow \textcircled{2}$$

where, V_m = Instantaneous voltage of message signal.

V_m = Max amplitude of message signal.

Amplitude of AM wave is given by,

$$A = V_c + V_m$$

$$A = V_c + V_m \sin \omega_m t$$

$$A = V_c + m V_c \sin \omega_m t \quad \left(\because m = \frac{V_m}{V_c} \right) \quad (V_m = m V_c)$$

The instantaneous voltage of resulting AM wave is

$$V = A \sin \omega_c t \rightarrow (3)$$

sub eq (3) in eq (4) we get,

$$V = (V_c + m V_c \sin \omega_m t) \sin \omega_c t$$

$$V = V_c \sin \omega_c t + m V_c \sin \omega_m t \cdot \sin \omega_c t \rightarrow (5)$$

The above equation may be expanded by means of trigonometric formula.

$$\sin A \sin B = \frac{1}{2} [\cos(A-B) - \cos(A+B)]$$

$$\Rightarrow \sin \omega_c t \cdot \sin \omega_m t = \frac{1}{2} [\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t]$$

equation (5) can be written as,

$$\Rightarrow V = V_c \sin \omega_c t + \frac{m V_c}{2} [\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t]$$

single tone expression for AM signal in frequency domain, :

The expression for AM wave in time domain is given by,

$$V = V_c \sin \omega_c t + \frac{m V_c}{2} [\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t]$$

$$\text{w.k.T, } \omega = 2\pi f$$

$$\Rightarrow V = V_c \sin 2\pi f_c t + \frac{m V_c}{2} [\cos 2\pi(f_c - f_m)t - \cos 2\pi(f_c + f_m)t]$$

In the above equation,

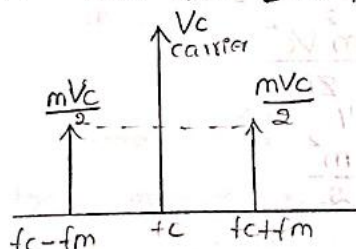
' f_c ' is the carrier frequency and its amplitude is ' V_c '.

' $f_c - f_m$ ' is the lower side band (LSB) & its amplitude is $\frac{m V_c}{2}$.

' $f_c + f_m$ ' is the upper side band (USB) & its amplitude is $\frac{m V_c}{2}$.

Bandwidth of AM signal :

The difference b/w USB and LSB frequencies is called as "bandwidth."



$$\begin{aligned}\text{Bandwidth} &= \text{USB} - \text{LSB} \\ &= (f_c + f_m) - (f_c - f_m) \\ &= f_c + f_m - f_c + f_m\end{aligned}$$

$$\boxed{\text{Bandwidth} = 2f_m}$$

Power relation in AM :

The expression for AM wave is given by,

$$V = \underbrace{V_c \sin \omega_c t}_{\text{carrier}} + \underbrace{\frac{mV_c}{2} \cos(\omega_c - \omega_m)t}_{\text{LSB}} - \underbrace{\cos(\omega_c + \omega_m)t}_{\text{USB}}$$

Total power = carrier power + LSB power + USB power

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

$$\text{WKT, power (P)} = \frac{V^2}{R}$$

$$\Rightarrow P_T = \frac{V_{\text{carrier}}^2}{R} + \frac{V_{\text{LSB}}^2}{R} + \frac{V_{\text{USB}}^2}{R}$$

$$\text{WKT, } V_{\text{rms}} = \frac{\text{max value}}{\sqrt{2}}$$

$$P_c = \frac{\left(\frac{V_c}{\sqrt{2}}\right)^2}{R} = \frac{V_c^2}{2R}$$

$$P_{\text{LSB}} = P_{\text{USB}} = \frac{\left[\frac{\frac{mV_c}{2}}{\sqrt{2}}\right]^2}{R} = \frac{\left(\frac{mV_c^2}{4}\right)}{R}$$

$$= \frac{m^2 V_c^2}{4} \times \frac{1}{2} = \frac{m^2 V_c^2}{8R}$$

$$\Rightarrow P_T = P_c + P_{\text{LSB}} + P_{\text{USB}}$$

$$= \frac{V_c^2}{2R} + \frac{m^2 V_c^2}{8R} + \frac{m^2 V_c^2}{8R}$$

$$= \frac{V_c^2}{2R} + \frac{m^2 V_c^2}{4R}$$

$$= \frac{V_c^2}{2R} + \frac{V_c^2}{2R} \cdot \frac{m^2}{2}$$

$$= P_c + P_c \cdot \frac{m^2}{2}$$

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

where, P_T = total power

P_c = carrier power

m = modulation index

* If we take critical modulation i.e., $m=1$ then total power is given by,

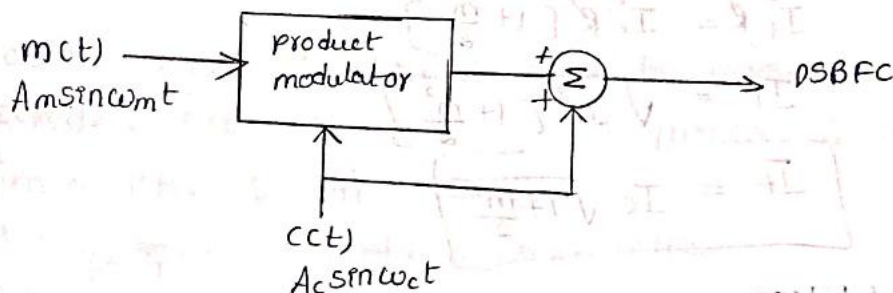
$$P_T = P_c \left[1 + \frac{m^2}{2} \right]$$

$$= P_c \left[1 + \frac{1^2}{2} \right]$$

$$= P_c \left[\frac{3}{2} \right]$$

$$P_T = 1.5 P_c$$

double side band full carrier :



Efficiency in AM :

It is defined as the ratio of power in side band to the total power.

$$\% \eta = \frac{\text{side band power}}{\text{Total power}} \times 100$$

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

$$= \frac{\frac{m^2}{2}}{1 + \frac{m^2}{2}} \times 100$$

$$= \frac{m^2}{2 + m^2} \times 100$$

$$\% \eta = \frac{m^2}{2 + m^2} \times 100$$

If $m=1$ i.e., 100% modulation

$$\% \eta = \frac{1}{2+1} \times 100 = \frac{1}{3} \times 100 = 33.33 \%$$

Voltage relation in AM :

$$\text{WKT, } P_T = P_c \left[1 + \frac{m^2}{2} \right] \rightarrow \textcircled{1}$$

$$\Rightarrow P_t = \frac{V_t^2}{R} \quad \& \quad P_c = \frac{V_c^2}{R}$$

Sub the value of P_t & P_c in eq ① we get

$$\frac{V_t^2}{R} = \frac{V_c^2}{R} \left[1 + \frac{m^2}{2} \right]$$

$$V_t = \sqrt{V_c^2 \left(1 + \frac{m^2}{2} \right)}$$

$$\boxed{V_t = V_c \sqrt{1 + \frac{m^2}{2}}}$$

current relation in AM :

Power relation in AM is given by, $P_T = P_c \left[1 + \frac{m^2}{2} \right] \rightarrow \textcircled{2}$

$$\text{WKT, } P_t = I_t^2 R \quad \& \quad P_c = I_c^2 R$$

Sub the value of P_t & P_c in eq ① we get,

$$I_t^2 R = I_c^2 R \left[1 + \frac{m^2}{2} \right]$$

$$I_t = \sqrt{I_c^2 \left[1 + \frac{m^2}{2} \right]}$$

$$\boxed{I_t = I_c \sqrt{1 + \frac{m^2}{2}}}$$

Modulation :

The process of changing some characteristics (Amplitude, phase, frequency) of the carrier signal in accordance with the instantaneous amplitude of message signal is called 'modulation'.

Need of modulation :

1. To reduce the noise
2. To increase the range of communication
3. To improve the quality of receiver
4. To avoid the interference.
5. To avoid the loss of signal
6. To multiplex the signals.

- 2) Draw the frequency spectrum of AM wave for the following data & determine the bandwidth.

$f_c = 200\text{kHz}$, $f_m = 10\text{kHz}$, $V_c = 150\text{V}$, $m = 0.45$.

$m = 0.45$

Upper side = $f_m + f_c = 200\text{kHz} + 10\text{kHz}$

$= 210\text{kHz}$

Lower side band = $f_m - f_c$

$= 200\text{kHz} - 10\text{kHz}$

$= 190\text{kHz}$

Amplitude of carrier is V_c i.e., 150V

Amplitude of side bands is $\frac{mV_c}{2} = \frac{0.45 \times 150}{2}$

$= 33.75\text{V}$

$= 33.75\text{V}$

$= 33.75\text{V}$

- 3) An audio signal of 1kHz is used to modulate a carrier of 50kHz . Determine side band frequencies & bandwidth.

$f_m = 1\text{kHz}$ & $f_c = 50\text{kHz}$

LSB = $f_c - f_m = 50\text{kHz} - 1\text{kHz}$

$= 49\text{kHz}$

USB = $f_m + f_c = 50\text{kHz} + 1\text{kHz}$

$= 51\text{kHz}$

- 4) A 20W carrier is modulated to a depth of 65% . Calculate the total power.

i) AM

ii) DSBSC

iii) SSBSC

Given, $P_c = 20\text{W}$

$m = 0.65$

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

$= 20 \left(1 + \frac{0.65^2}{2}\right)$

$P_T = 24.25\text{W}$ for AM

$$\text{DSB-SC} : P_T = P_C \frac{m^2}{2}$$

$$P_T = 20 \times \frac{0.65^2}{2}$$

$$P_T = 4.225 \text{ W}$$

$$\text{SSB-SC} :$$

$$P_T = P_C \frac{m^2}{4}$$

$$= 20 \times \frac{0.65^2}{4}$$

$$P_T = 2.1125 \text{ W}$$

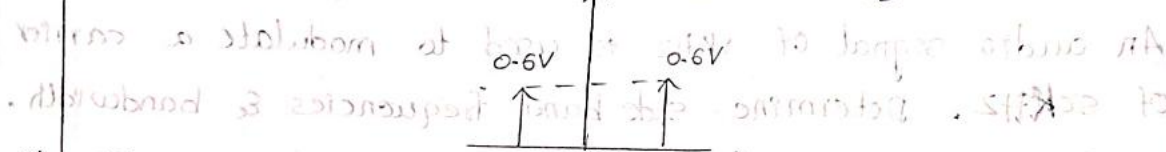
Q) A 2 MHz carrier with an amplitude of 2V peak is modulated by 2 kHz signal with $m=0.6$; sketch the frequency domain spectrum for AM?

A) $f_c = 2 \text{ MHz}$, $V_c = 2 \text{ V}$, $f_m = 2 \text{ kHz}$, $m = 0.6$

$$\text{USB} = f_c + f_m = 2 \times 10^6 + 2 \times 10^3 = 2.002 \text{ MHz}$$

$$\text{LSB} = f_c - f_m = 2 \times 10^6 - 2 \times 10^3 = 1.998 \text{ MHz}$$

Amplitude of side band is $\frac{m \times V_c}{2} = \frac{0.6 \times 2}{2} = 0.6 \text{ V}$



Q) The AM wave is $5(1 + 0.6 \sin 6280t) \sin 2\pi \times 10^4 t$

i) Find side band frequencies

ii) amplitude of each side band

iii) bandwidth.

iv) Let $R = 1 \Omega$ find the total power?

A) Given that, $5(1 + 0.6 \sin 6280t) \sin 2\pi \times 10^4 t$

comparing with $A_c [1 + m \sin \omega_m t] \sin \omega_c t$

$$A_c = 5 \text{ V}, \omega_m = 6280, m = 0.6, \omega_c = 2\pi \times 10^4$$

$$2\pi f_m = 6280$$

$$f_m = \frac{6280}{2\pi}$$

$$f_m = 999$$

$$2\pi f_c = 2\pi \times 10^4$$

$$f_c = \frac{2\pi \times 10^4}{2\pi}$$

$$f_c = 10^4$$

$$\text{USB} = f_c + f_m = 10000 + 999 = 10999$$

$$\text{LSB} = f_c - f_m = 10000 - 999 = 9001$$

Amplitude of each side band $= \frac{mV_c}{2} = \frac{0.6 \times 5}{2} = 1.5 \text{ mA}$

Bandwidth $= 2f_m = 2 \times 999 = 1998$

1) $R = 1 \Omega$

$P_T = \frac{A_c^2}{2R} = \frac{(5)^2}{2 \times 1} = \frac{25}{2} = 12.5$

- Q) A communication transmitter radiates 1.08 kW of AM signal. If the carrier power is 1 kW, find the modulation index?

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

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

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

$\sqrt{\left(\frac{P_T}{P_c} - 1 \right) \times 2} = m$

$m = \sqrt{2 \left(\frac{1.08 \times 10^3}{1 \times 10^3} - 1 \right)}$

$m = 0.4$

- Q) A carrier of power of 400 W is amplitude modulate a depth of 100%.

Find the i) total power

ii) side band power.

i) $P_T = 400 \left[1 + \frac{1}{2} \right]$

$P_T = 400 \left[1 + 0.5 \right] = 600$

ii) Side band power $= P_T - P_c = \frac{P_c m^2}{2}$

$= \frac{400 \times 1}{2} = 200$

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

- Q) A modulating ^{message} signal $10 \sin(2\pi \times 10^3 t)$ is used to modulate a carrier signal $20 \sin(2\pi \times 10^4 t)$. Determine the modulation index, % modulation, frequencies of the side band components & their amplitude. What will be the bandwidth of modulated signal & also total power radiated if $R = 1 \Omega$.

Given that,

$$A_m = 10 \sin(2\pi \times 10^3 t)$$

$$A_c = 20 \sin(2\pi \times 10^4 t)$$

$$A_c = 20V$$

$$A_m = 10V$$

$$f_m = 1 \text{ kHz}$$

$$f_c = 10 \text{ kHz}$$

$$\text{Bandwidth} = 2 \times f_m$$

$$= 2 \times 1 \text{ kHz}$$

$$= 2 \text{ kHz}$$

$$\text{i) Modulation index} = \frac{A_m}{A_c} = \frac{10}{20} = 0.5$$

$$\text{ii) \% modulation} = 0.5 \times 100 = 50\%$$

$$\text{iii) LSB} = f_c - f_m = 10 \text{ kHz} - 1 \text{ kHz} = 9 \text{ kHz}$$

$$\text{USB} = f_c + f_m = 11 \text{ kHz}$$

$$\text{iv) Amplitude of side band frequencies} = \frac{m A_c}{2} = \frac{0.5 \times 20}{2} = 5V$$

$$\text{v) } P_T = P_c \left[1 + \frac{m^2}{2} \right] \Rightarrow P_c = \frac{V_c^2}{2R} = \frac{20^2}{2 \times 1} = 200W$$

$$= 200 \left[1 + \frac{(0.5)^2}{2} \right]$$

$$= 200 \left[1 + \frac{0.25}{2} \right]$$

$$= 225W$$

Q) An AM broadcast radio transmitter radiates 10kW of power if the modulation percentage is 60. Calculate how much of the carrier power.

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

$$\frac{P_T}{\left(1 + \frac{m^2}{2} \right)} = P_c \Rightarrow \frac{10 \times 10^3}{\left(1 + \frac{(0.6)^2}{2} \right)} = \frac{8475W}{100} = 8.475 \times 10^3 W$$

- Q. The antenna current of an AM transmitter is $8A$. If only the carrier is sent, but it increases to $8.93A$ if the carrier is modulated by a single sinusoidal wave. Determine the percentage modulation. Also find the antenna current if the % of modulation changes to 0.8.

$$I_t = I_c \sqrt{1 + \frac{m^2}{2}}$$

Given that, $I_c = 8A$

$$\frac{I_t}{I_c} = \sqrt{1 + \frac{m^2}{2}}$$

$I_t = 8.93A$

$$2 \times \left(\frac{I_t}{I_c} - 1 \right) = \sqrt{m^2}$$

$$2 \times \left[\left(\frac{I_t}{I_c} \right)^2 - 1 \right] = m^2$$

$$\sqrt{2 \times \left(\frac{I_t}{I_c} \right)^2 - 1} = m$$

$$\sqrt{2 \times \left(\frac{8.93}{8} \right)^2 - 1} = m$$

~~m = 0.72~~ $m = 0.70$

% modulation = $0.70 \times 100 = 70\%$

⇒ If % modulation changes to 0.8

$I_t = ?$

$$I_t = I_c \sqrt{1 + \frac{m^2}{2}}$$

$$= 8 \sqrt{1 + \frac{(0.8)^2}{2}}$$

$I_t = 9.19A$

- Q. A carrier wave is represented by the expression $V_c(t) = 10 \sin \omega t$. Draw the waveform of an AM wave for $m = 0.5$

A) $m = 0.5$ that means under modulation ($V_m < V_c$)

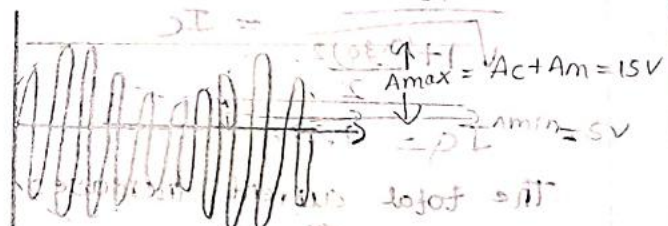
Given that, $A_c = 10V$

$m = 0.5$

where, $m = \frac{A_m}{A_c}$

$A_m = m \times A_c$
 $= 0.5 \times 10$

$A_m = 5V$



Q) Prove that in amplitude modulation, max avg power transmitted by an antenna is 1.5 times the carrier power?

A) $P_T = P_C \left[1 + \frac{m^2}{2} \right]$ we should consider the degree of modulation as critical modulation i.e., $m=1$

$$P_T = P_C \left[1 + \frac{1}{2} \right]$$

$$P_T = P_C [1 + 0.5]$$

$$P_T = P_C 1.5$$

$$\boxed{P_T = 1.5 P_C}$$

Multitone AM :

Q) The antenna current of an AM transmitter is 10A. When it is modulated to depth of 30% by an audio signal. It increases to 11A. When another signal modulates the carrier signal, what will be the modulation index due to second signal?

A) Given that, total current = 10A $\rightarrow I_{t1}$

The total current $I_{t1} = 10A$

It increases to 11A

$$I_{t2} = 11A$$

$$I_C = ?$$

$$I_{t1} = I_C \sqrt{1 + \frac{m_1^2}{2}}$$

$$\frac{I_{t1}}{\sqrt{1 + \frac{m_1^2}{2}}} = I_C$$

$$\frac{10}{\sqrt{1 + \frac{(0.3)^2}{2}}} = I_C$$

$$I_C = 9.78A$$

The total current increases to 11A

$$I_{t2} = 11A$$

$$I_t = I_c \sqrt{1 + \frac{\mu_t^2}{2}}$$

$$\sqrt{2 \left[\left(\frac{I_t}{I_c} \right)^2 - 1 \right]} = \mu_t$$

$$\mu_t = 0.72 = 72\%$$

$$\mu_2 = ?$$

$$\mu_t = \sqrt{\mu_1^2 + \mu_2^2}$$

$$(\mu_t)^2 = \mu_1^2 + \mu_2^2$$

$$\mu_t^2 - \mu_1^2 = \mu_2^2$$

$$\mu_2 = \sqrt{\mu_t^2 - \mu_1^2}$$

$$\mu_2 = 0.65 = 65\%$$

9) The antenna current of AM broadcast transmitter modulates to the depth of 40% by an audio sine wave is 11A. It increases to 12A as a result of simultaneous modulation by another audio sine wave. What is the modulation index due to the second wave?

Given that, $P_t = 11A$, $\mu_1 = 0.40$, $P_t = 12A$, $\mu_2 = ?$

$$I_c = ?$$

$$I_t = I_c \sqrt{1 + \frac{\mu^2}{2}}$$

$$I_c = \frac{I_t}{\sqrt{1 + \frac{\mu^2}{2}}} = \frac{11}{\sqrt{1 + \frac{(0.40)^2}{2}}}$$

$$I_c = 10.58 A$$

The total current increases to 12A

$$I_t = I_c \sqrt{1 + \frac{\mu_t^2}{2}}$$

$$\mu_t = \sqrt{2 \left[\left(\frac{I_t}{I_c} \right)^2 - 1 \right]}$$

$$\mu_t = \sqrt{2 \left[\left(\frac{12}{10.58} \right)^2 - 1 \right]}$$

$$\mu_t = 0.75$$

$$\mu_2 = ?$$

$$\mu_t = \sqrt{\mu_1^2 + \mu_2^2}$$

$$\mu_2 = \sqrt{\mu_t^2 - \mu_1^2}$$

$$\mu_2 = \sqrt{(0.75)^2 - (0.40)^2}$$

$$\mu_2 = 0.63$$

- Q) A certain transmitter radiates 10 kW ^(P_c) with carrier ^(P_c) unmodulated. If we don't have modulation, there is only signal ^(P_s) & 12 kW when the carrier is sinusoidally modulated. If modulation ^{is done} power is 15 kW [↑] If modulation is not done then power is 10 kW . calculate the modulation index, If another sine wave corresponding to 50% modulation is transmitted simultaneously, determine the total radiated power?

A) $P_c = 10\text{ kW}$, $P_T = 12\text{ kW}$

$$P_T = P_c \left[1 + \frac{\mu_1^2}{2} \right]$$

$$\sqrt{2 \left(\left(\frac{P_T}{P_c} \right) - 1 \right)} = \mu_1$$

$$\sqrt{2 \left(\left(\frac{12 \times 10^3}{10 \times 10^3} \right) - 1 \right)} = \mu_1$$

$$\mu_1 = 0.81 = 81\% \quad 0.63 = 63\%$$

If another sine wave corresponding to 50% modulation is transmitted simultaneously.

$$\mu_2 = 0.50$$

$$\mu_t = \sqrt{\mu_1^2 + \mu_2^2}$$

$$\mu_t = \sqrt{(0.81)^2 + (0.50)^2} = \sqrt{0.63 + 0.50} = \sqrt{1.13} = 1.06$$

$$\mu_t = 0.80 = 80\%$$

$$P_t = P_c \sqrt{1 + \frac{\mu_t^2}{2}}$$

$$\sqrt{2 \left(\left(\frac{P_t}{P_c} \right) - 1 \right)} = \mu_t$$

$$P_t = 10 \times 10^3 \left[1 + \frac{(0.80)^2}{2} \right]$$

$$= 13.2\text{ kW}$$

- Q) An AM transmitter radiates 9 kW of power when the carrier unmodulated. & 10.125 kW when the carrier is sinusoidally modulated. Find the modulation index, % of modulation. Now if another sine wave corresponding to 40% modulation is transmitted simultaneously, then calculate the total radiated power?

$$P_c = 9 \text{ kW}, \quad P_T = 10.125 \text{ kW}$$

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

$$\mu = \sqrt{2 \left[\frac{P_T}{P_c} - 1 \right]}$$

$$\mu = \sqrt{2 \left[\left(\frac{10.125 \times 10^3}{9 \times 10^3} \right) - 1 \right]}$$

$$\mu_1 = 0.5 \rightarrow \% \text{ of modulation} = 50\%$$

$$\mu_2 = 0.40$$

$$\mu_{lt} = \sqrt{(0.5)^2 + (0.40)^2}$$

$$\mu_{lt} = 0.64$$

$$P_t = P_c \sqrt{1 + \frac{\mu_{lt}^2}{2}}$$

$$= 9 \times 10^3 \sqrt{1 + \frac{0.64^2}{2}}$$

$$P_t = 9.81 \text{ kW} \quad 10.81 \text{ kW}$$

$$\Rightarrow \% \text{ modulation} = 64\%$$

Modulation index of AM : (1)

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

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

Modulation index is the value lying b/w 0 to 1.

If $m < 1$ under modulation $V_m < V_c$

If $m = 1$ critical modulation $V_m = V_c$

If $m > 1$ over modulation $V_m > V_c$

→ Modulation index is also called as modulation factor, modulation co-efficient, or degree of modulation or depth of modulation.

calculation of modulation index :

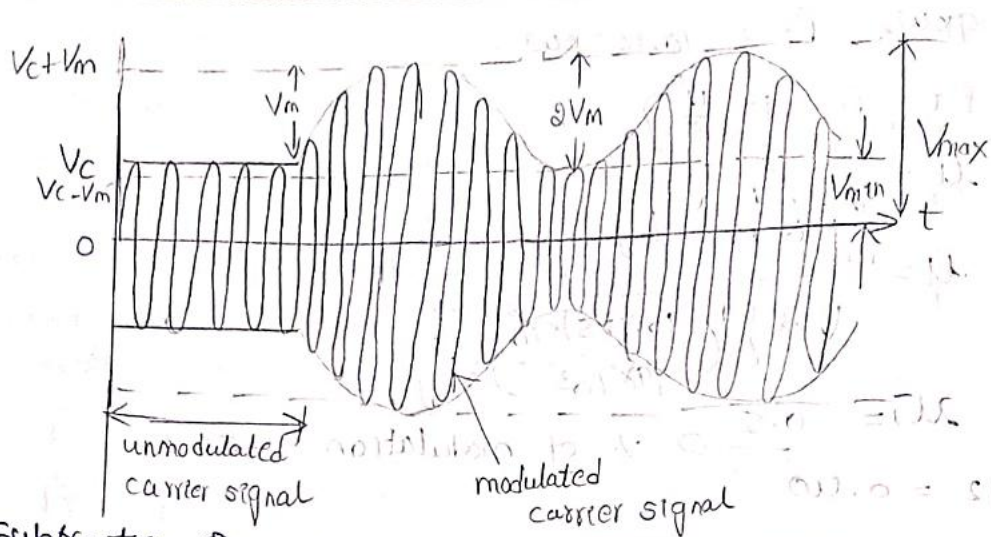
$$V_{\max} = V_c + V_m \rightarrow (1)$$

$$V_{\min} = V_c - V_m \rightarrow (2)$$

As signal is going to be increased then the power is also increases.

→ If modulation index is then power ↑

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



Subtraction ① & ② we get,

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

$$\Rightarrow V_c + V_m - V_c + V_m = 2V_m$$

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

Adding ① & ② we get,

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

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

$$\therefore m = \frac{V_m}{V_c} = \frac{V_{max} - V_{min}}{V_{max} + V_{min}}$$

$$\% \text{ of modulation} = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} \times 100$$

Effects of over modulation

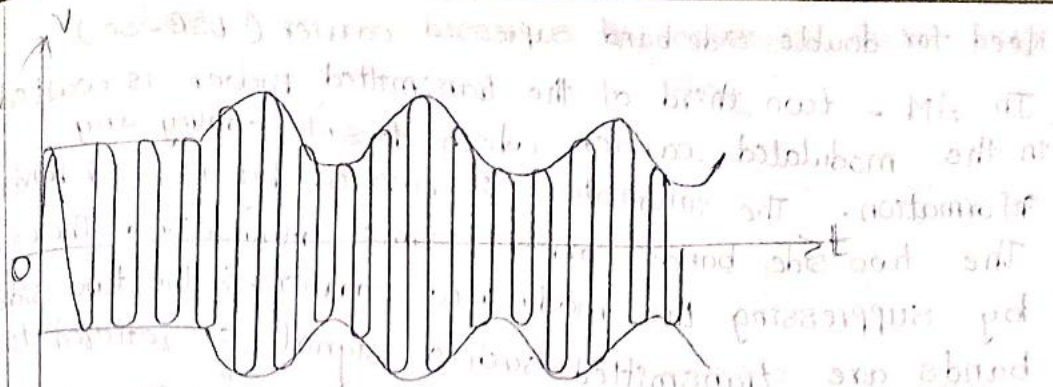
If $m < 1$ under modulation ($V_m < V_c$)

If $m = 1$ critical modulation ($V_m = V_c$)

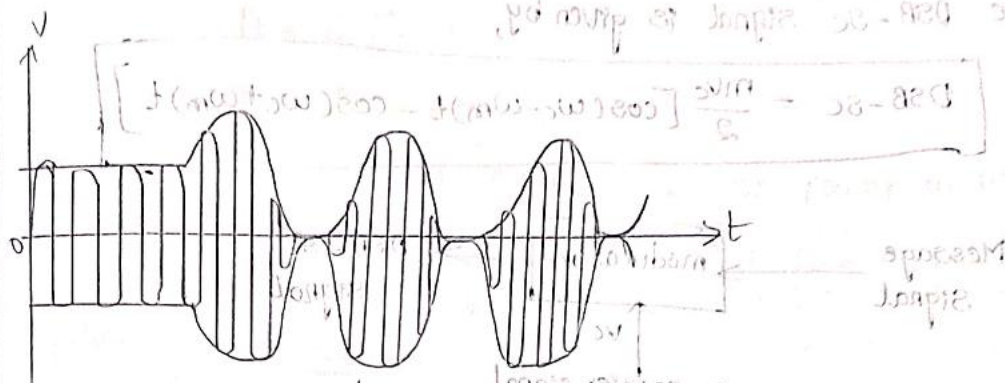
If $m > 1$ over modulation ($V_m > V_c$)

$$① \leftarrow m + V_c = \text{max}$$

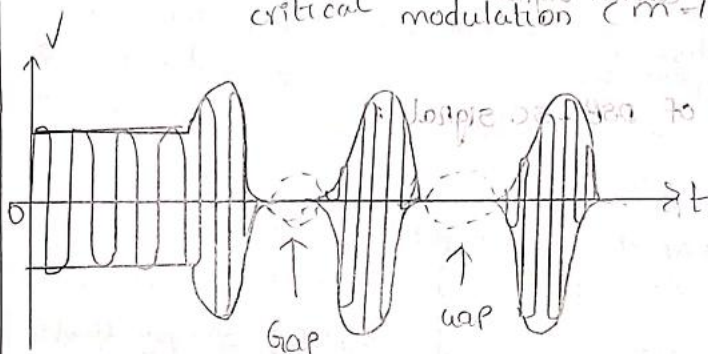
$$② \leftarrow m - V_c = \text{min}$$



under modulation ($m < 1$)



critical modulation ($m = 1$)



over modulation ($m > 1$)

If $m > 1$, the envelope overlaps then the some portion of information will be lost.

To avoid, over modulation, the modulation index $m \leq 1$ i.e. the amplitude of the message signal is always less than or equal to the amplitude of the carrier signal ($V_m \leq V_c$)

The following are the effects of over modulation:

1. Transmission quality becomes poor.
2. envelop will be clipped at the receiver.
3. envelop distortion occurs.
4. Instability in carrier frequency & Voltage.

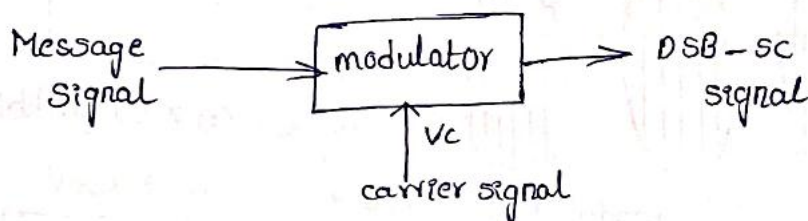
Need for double sideband suppressed carrier (DSB-SC) :

In AM - two third of the transmitted power is wasted in the modulated carrier which doesn't convey any information. The information is carried by two side bands.

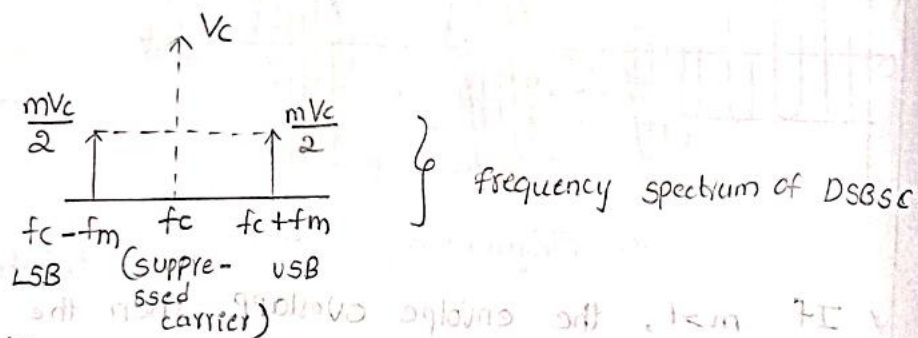
The two side bands carries same information. Therefore by suppressing the modulated carrier & the two side bands are transmitted such a signal is referred to as "Double side band suppressed carrier" (DSB-SC). Hence in DSBSC, a lot of power is saved compared to AM.

The DSB-SC signal is given by,

$$DSB-SC = \frac{mV_c}{2} [\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t]$$



Frequency spectrum of DSB-SC signal :



The bandwidth of DSB-SC is equal to $2f_m$.

$$Bw = (f_c + f_m) - (f_c - f_m)$$

$$= f_c + f_m - f_c + f_m$$

$$B.w = 2f_m$$

Power saving in DSB-SC signal :

The carrier signal is suppressed from the AM modulated wave. Therefore the two side bands are transmitted.

$$\% \text{ Power saving} = \frac{\text{carrier power}}{\text{Total power}} \times 100$$

WKT, total Power transmitted in AM wave is

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

$$\therefore \text{Power saving} = \frac{P_c}{P_c \left[1 + \frac{m^2}{2} \right]} = \frac{2}{2+m^2} \times 100$$

If $m=1$, then Power saving = $\frac{2}{2+1} \times 100 = 66.66\%$

Power derivation :

WKT, total power transmitted in AM wave is

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

$$P_t = P_c + P_c \frac{m^2}{2}$$

carrier is suppressed in DSB-SC so power in DSB-SC is given by,

$$P_t = P_c \frac{m^2}{2}$$

If $m=1$ then $P_t = 0.5 P_c$

Advantages of DSB-SC :

1. It saves power more than AM.
2. Generation of DSB-SC is simple compared to AM.
3. Low power consumption.

Disadvantages :

1. Bandwidth required for DSB-SC is same as that of AM.
2. complex detection.

Need for single side band suppressed carrier (SSB-SC) :

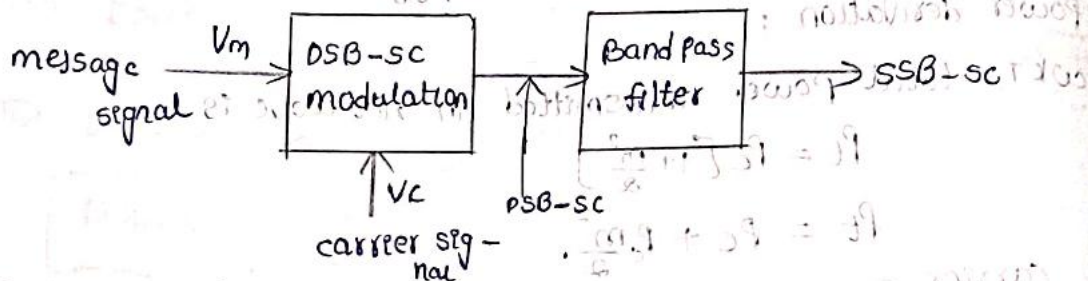
In AM, two-third of the transmitted power is wasted in the modulated carrier which does not convey any information. The information is carried by two sidebands only. Therefore by suppressing the modulated carrier & two sidebands are transmitted such a signal is referred to as "DSBSC signal". But both sidebands carry same information. That is in DSB-SC, the information is transmitted twice. Therefore one side band may be suppressed, the resulting signal is called "single side band suppressed carrier" (SSB-SC) signal. Hence in SSB-SC, a lot of power & half of the bandwidth can be saved compared to AM and DSBSC.

The DSB-SC signal is given by,

$$V_{DSB-SC} = \frac{mV_c}{2} [\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t]$$

If LSB is chosen for transmission, in case of SSB.

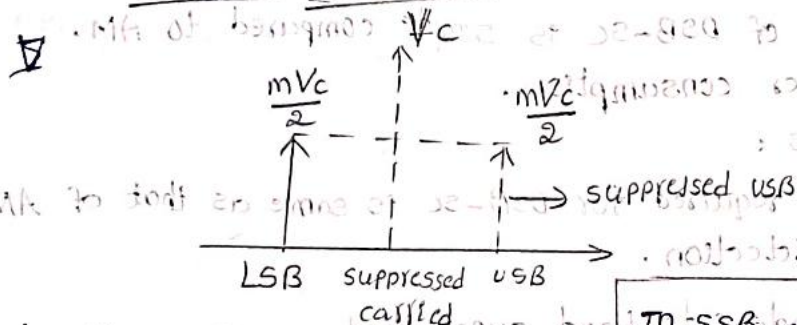
$$V_{SSB} = \frac{mV_c}{2} \cos(\omega_c - \omega_m)t$$



Generation of SSB-SC signal

For generating SSB-SC, the DSB-SC signal is passed through Band pass filter, the cutoff frequency either LSB or USB comes out of band pass filter.

Frequency spectrum of SSB-SC



power in SSB:

WKT, total power transmitted in AM wave is,

$$P_t = P_c \left[1 + \frac{m^2}{2} \right] = P_c + P_c \cdot \frac{m^2}{2}$$

In SSB, one side band & carrier is suppressed. So the total power is given by,

$$P_t = P_c \cdot \frac{m^2}{4}$$

If $m=1$ then $P_t = 0.25 P_c$

Bandwidth:

$$BW = f_c - (f_c - f_m) = f_c - f_c + f_m$$

$$BW = f_m$$

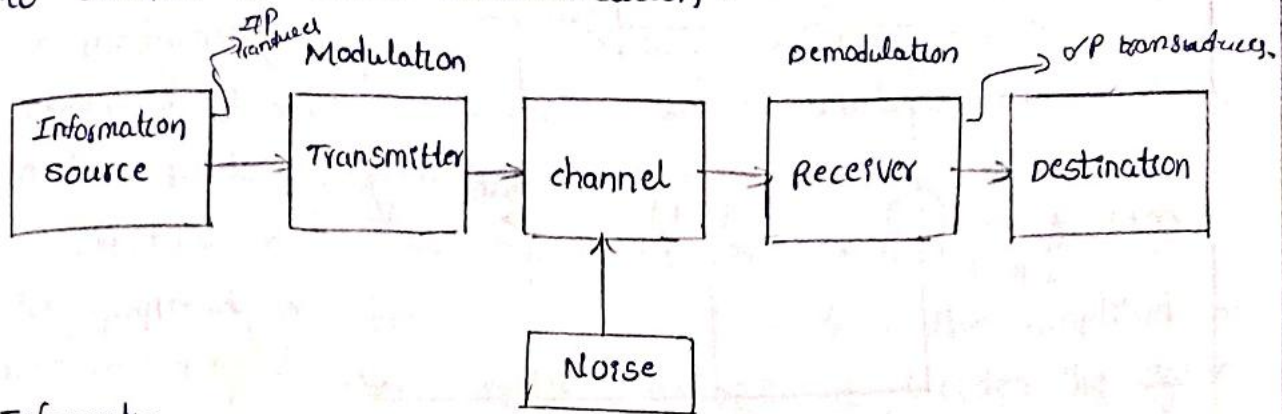
Advantages of SSB-SC:

1. More transmitted power is saved than AM & DSB-SC.
2. Requirement of bandwidth is less.
3. Less noise.

Disadvantages:

1. The generation & reception of SSB signal is complex process.
2. Since carrier is absent the SSB transmitter & receiver need to have excellent frequency stability.

communication system : The transfer of information from one place to another is called communication.



Information source : The message produced by information source is not electrical in nature, but it may be a Voice signal, a picture signal etc.

I/P transducer : The I/P transducer converts the message signal into time varying electrical signals. This signal is called baseband signal.

Transmitter : The transmitter converts the message signal into a suitable form for propagating over channel. This is achieved by modulating the carrier signal which may be an electromagnetic wave.

channel : The channel provides a physical connection between the transmitter & receiver and is used to send the signals from transmitter to receiver.

Noise : The unwanted form of electric energy is known as "noise." and it enters into the ~~channel~~ communication channel system through the channel.

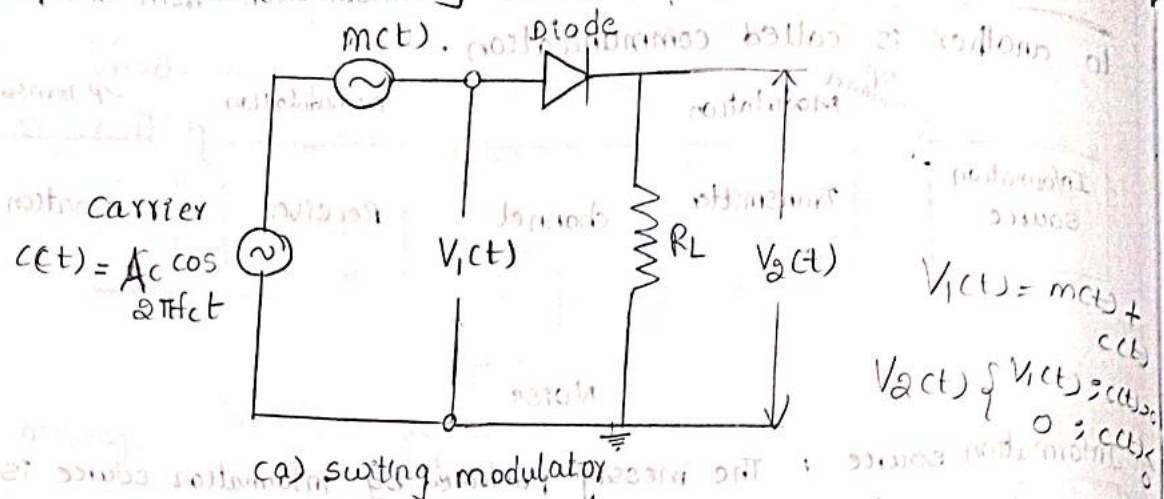
Receiver : The main function of the receiver is used to receive the signals from the transmitter through the channel.

O/P transducer : The O/P transducer converts the electrical signals into message signal.

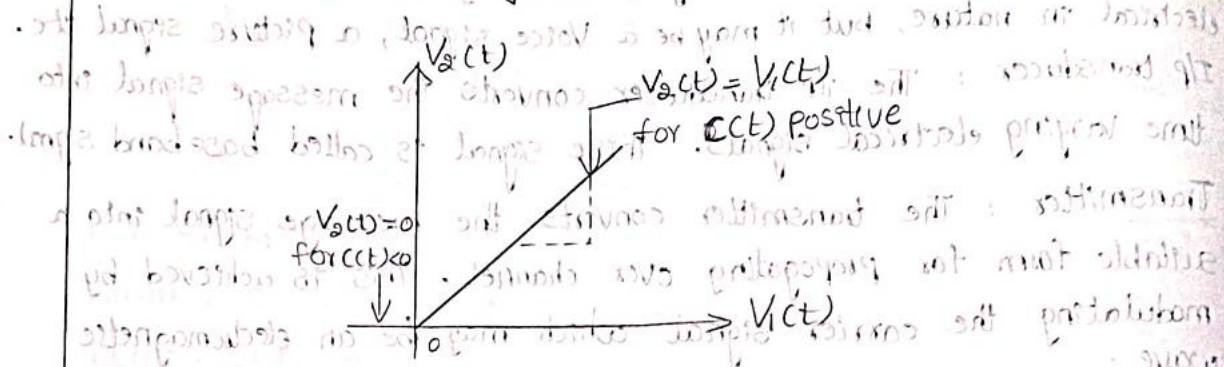
Destination : Destination is the final stage which is used to convert an electrical signal into its original form.

→ The O/P of the receiver (Destination) may be loud speaker, Video display unit, TV picture unit, computer etc.

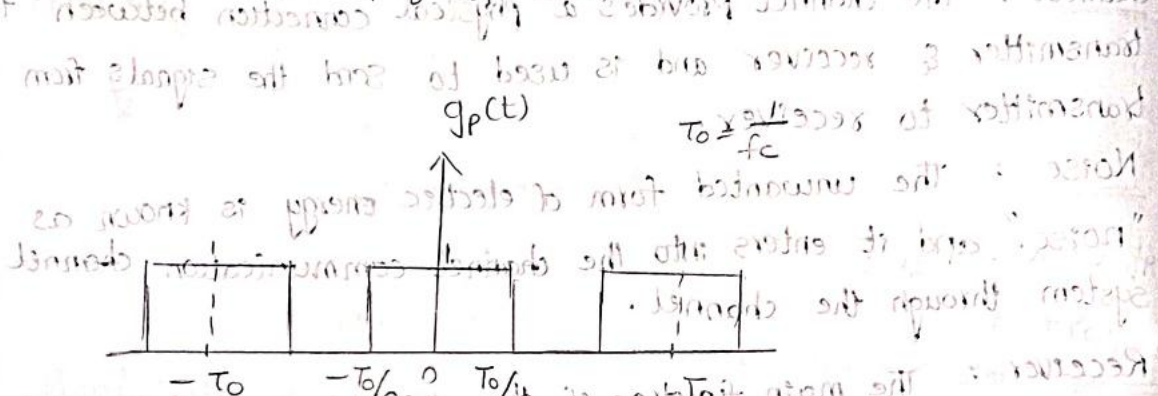
AM waves, switching modulator :



(a) switching modulator



(b) i/p & o/p relationship



(c) A Periodic Pulse train

$$g_p(t) = \frac{1}{2} + \frac{2}{\pi} \sum_{n=1}^{\infty} \frac{(-1)^{n+1}}{2n-1} \cos[2\pi f_c t (2n-1)]$$

$$g_p(t) = \frac{1}{2} + \frac{2}{\pi} \cos \omega_c t + \text{odd harmonic component}$$

$$\begin{aligned} \therefore V_2(t) &= V_1(t) \cdot g_p(t) \\ &= [A_c \cos \omega_c t + m(t)] \left[\frac{1}{2} + \frac{2}{\pi} \cos \omega_c t + \dots \right] \\ &= \frac{A_c}{2} \cos \omega_c t + \frac{m(t)}{2} + \frac{2}{\pi} \cos \omega_c t \cdot A_c \cos \omega_c t + \dots \end{aligned}$$

By connecting BPF then we get AM

$$\frac{A_c}{2} \cos \omega_c t + \frac{m(t)}{2} + \frac{2}{\pi} \cos \omega_c t$$

The switching modulator using a diode has been shown in fig. In the ckt the diode is assumed to be operating as a switch. The modulating signal $m(t)$ & the carrier signal $c(t)$ are connected in series with each other. Therefore, the i/p voltage to the diode is given by,

$$V_1(t) = c(t) + m(t) = A_c \cos 2\pi f_c t + m(t)$$

The amplitude of carrier is much larger than the amplitude of message signal and carrier signal $[c(t)]$ decides the status of the diode.

working operation and analysis:

Let us assume the diode acts as an ideal switch. Hence, it acts a closed switch when it is forward biased in the +ve half cycle of the carrier signal. whereas it acts as an open switch when it is reverse biased in the -ve half cycle of the carrier & offers an infinite impedance.

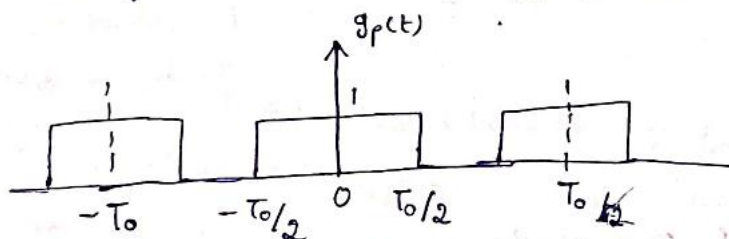
Therefore, the o/p voltage $V_2(t) = V_1(t)$ in the +ve half cycle of $c(t)$ and $V_2(t) = 0$ in the -ve neg half cycle of $c(t)$.

$$\text{Hence, } \begin{cases} V_2(t) = V_1(t) & \text{for } c(t) \geq 0 \\ V_2(t) = 0 & \text{for } c(t) < 0 \end{cases}$$

we can express $V_2(t)$ mathematically as under:

$$\begin{aligned} V_2(t) &= V_1(t) \cdot g_p(t) = [c(t) + m(t)] \cdot g_p(t) \\ &= [A_c \cos 2\pi f_c t + m(t)] \cdot g_p(t) \rightarrow \text{①} \end{aligned}$$

where $g_p(t)$ is a periodic pulse train of duty cycle equal to one half cycle period i.e., $T_0/2$ (where $T_0 = 1/f_c$)



Let us express $g_p(t)$ with the help of fourier series as under:

$$g_p(t) = \frac{1}{2} + \frac{2}{\pi} \sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos [2\pi f_c t (2n-1)]$$

$$g_p(t) = \frac{1}{2} + \frac{2}{\pi} \cos (2\pi f_c t) + \text{odd harmonic components}$$

Sub $g_p(t)$ in eq ① we get,

$$V_a(t) = [m(t) + A_c \cos 2\pi f_c t] \left\{ \frac{1}{2} + \frac{2}{\pi} \sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos(2n\pi f_c t) \right\}$$

Therefore,

$$V_a(t) = [m(t) + A_c \cos 2\pi f_c t] \left\{ \frac{1}{2} + \frac{2}{\pi} \cos(2\pi f_c t) + \text{odd harmonics} \right\}$$

The odd harmonics in this expression are unwanted, & therefore, are assumed to be eliminated.

Hence,

$$V_a(t) = \underbrace{\frac{m(t)}{2}}_{\text{modulating signal}} + \underbrace{\frac{A_c}{2} \cos 2\pi f_c t + \frac{2}{\pi} m(t) \cos(2\pi f_c t)}_{\text{AM wave}} + \underbrace{\frac{2A_c}{\pi} \cos(2\pi f_c t)}_{\text{Second harmonic of carrier.}}$$

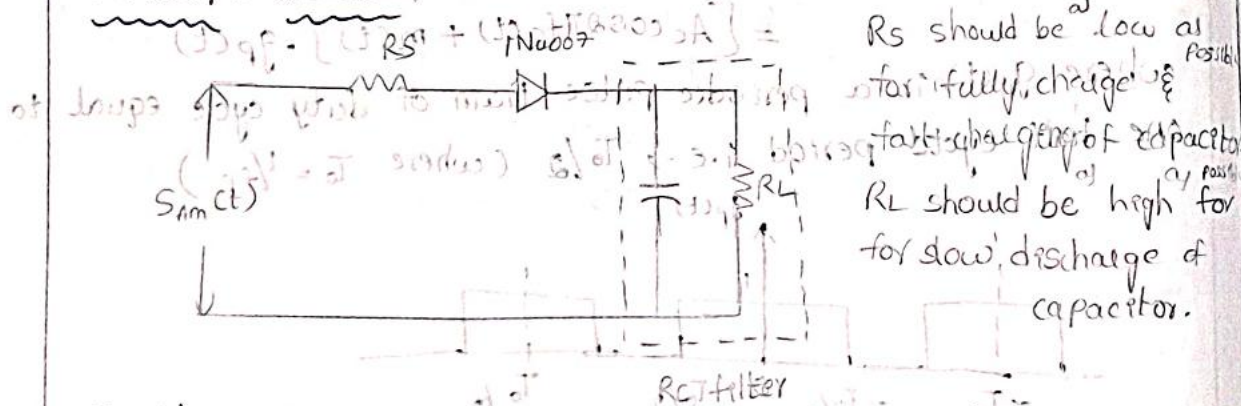
In this expression, the first and the fourth terms are unwanted terms whereas the second & third terms together represents the AM wave.

Combining the second and third terms together we obtain,

$$V_a(t) = \frac{A_c}{2} \left[1 + \frac{4}{\pi} m(t) \right] \cos(2\pi f_c t) + \text{unwanted terms}$$

This is the required expression for the AM wave with $m = \left[\frac{4}{\pi A_c} \right]$. The unwanted terms can be eliminated using a band-pass filter [BPF].

Envelope detector :



Envelope detector is a simple & very efficient device which is suitable for the detection of a AM signal.

The ckt diagram of envelope demodulator is shown in above figure.

The envelope detector consists of a diode & RC filter.

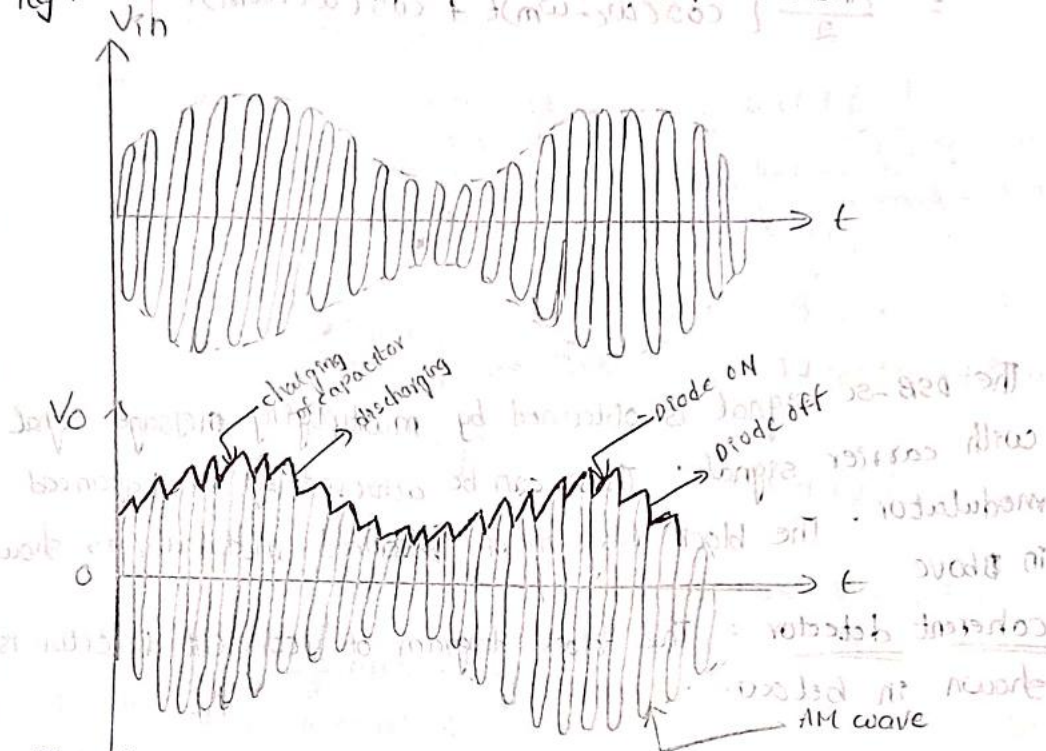
The modulated AM wave is applied at the i/p of the detector. In every +ve half cycle of i/p, the demodulator diode is forward biased & the capacitor, which is connected across the load resistance R charges to peak value of i/p Voltage.

As soon as the capacitor charges to the peak value, the diode stop conducting.

The capacitor will now discharge through R b/w the +ve peaks as shown in fig. The discharging process continues until the next +ve half cycle is applied.

When the i/p signal becomes greater than the capacitor Voltage, the diode conducts again & the process repeats itself.

The i/p & o/p waveforms for the envelope detector is shown in fig.



It shows the charging & discharging of the filter capacitor.

It may be observed from these waveforms that the envelope of the AM wave is being recovered successfully.

Here we have assumed that the diode is ideal & the AM wave applied to the i/p of the demodulator is supplied by a source having internal resistance R_s .

Note : R_s source resistance R_s should be as low as possible for fast & fully charging of capacitor & load resistance R_L should be as high as possible for slow discharge of capacitor.

DSB-SC using balanced modulator:

DSB-SC signal can be obtained by simply multiplying the message signal with the carrier signal.

$$\begin{aligned} \text{Let } m(t) &= A_m \cos \omega_m t \\ c(t) &= A_c \cos \omega_c t \end{aligned}$$

By doing multiplication of $m(t)$ & $c(t)$, we have the lower & upper sidebands without carrier as shown below.

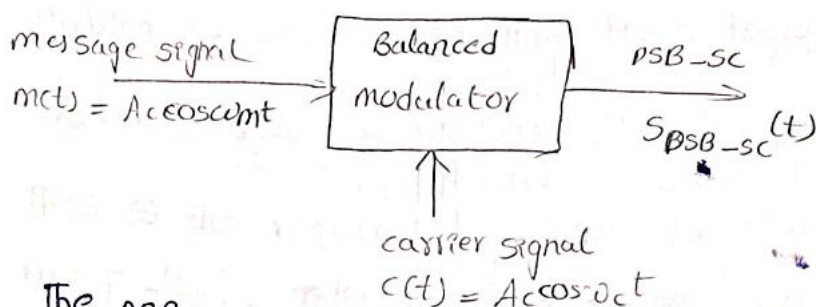
$$m(t) \cdot c(t)$$

$$= A_m \cos \omega_m t \cdot A_c \cos \omega_c t$$

$$= A_m A_c \cdot \cos \omega_m t \cdot \cos \omega_c t$$

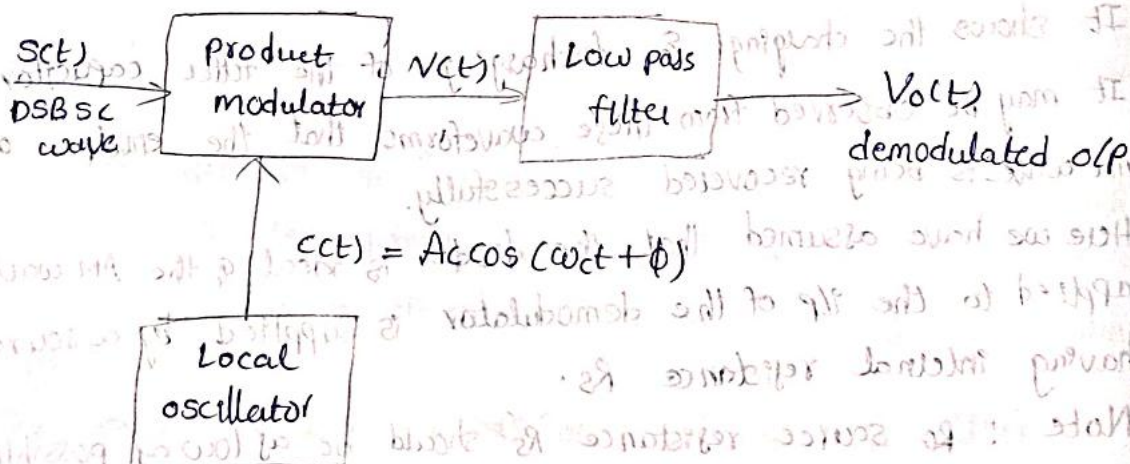
$$= A_m A_c \cdot \cos \omega_c t \cdot \cos \omega_m t$$

$$= \frac{A_m A_c}{2} [\cos(\omega_c - \omega_m)t + \cos(\omega_c + \omega_m)t]$$



The DSB-SC signal is obtained by multiplying message signal with carrier signal. This can be achieved by a balanced modulator. The block diagram of balanced modulator is shown in above.

coherent detector: The block diagram of coherent detector is shown in below.



In this process, the message signal can be extracted from DSBSC wave by multiplying it with a carrier, having the same frequency & phase of the carrier used in DSBSC modulation. The resulting signal is then passed through low pass filter. O/P of this filter is the desired message signal.

Let DSBSC wave be,

$$s(t) = A_c m(t) \cos \omega_c t$$

The o/p of local oscillator is,

$$c(t) = A_c \cos(\omega_c t + \phi)$$

where ϕ is the difference b/w the local oscillator signal & the carrier signal, which is used for DSBSC modulation.

From the fig, we can write the o/p of product modulator as

$$V(t) = s(t) \cdot c(t)$$

$$\Rightarrow V(t) = A_c \cos \omega_c t \cdot m(t) \cdot [A_c \cos(\omega_c t + \phi)]$$

$$= A_c^2 m(t) \cos \omega_c t \cdot \cos(\omega_c t + \phi)$$

$$= A_c^2 m(t) \cos(\omega_c t + \phi) \cos \omega_c t$$

$$= \frac{A_c^2 m(t)}{2} [\cos(\omega_c t + \phi - \omega_c t) + \cos(\omega_c t + \phi + \omega_c t)]$$

$$= \frac{A_c^2 m(t)}{2} [\cos \phi + \cos(2\omega_c t + \phi)]$$

$$= \frac{A_c^2 m(t)}{2} \cos \phi + \frac{A_c^2 m(t)}{2} \cos(2\omega_c t + \phi)$$

$$V(t) = \frac{A_c^2}{2} m(t) \cos \phi + \frac{A_c^2}{2} m(t) \cos(2\omega_c t + \phi)$$

In the above equation, the first term is the scaled version of message signal. It can be extracted by passing the above signal through a low pass filter.

Therefore, the o/p of low pass filter is

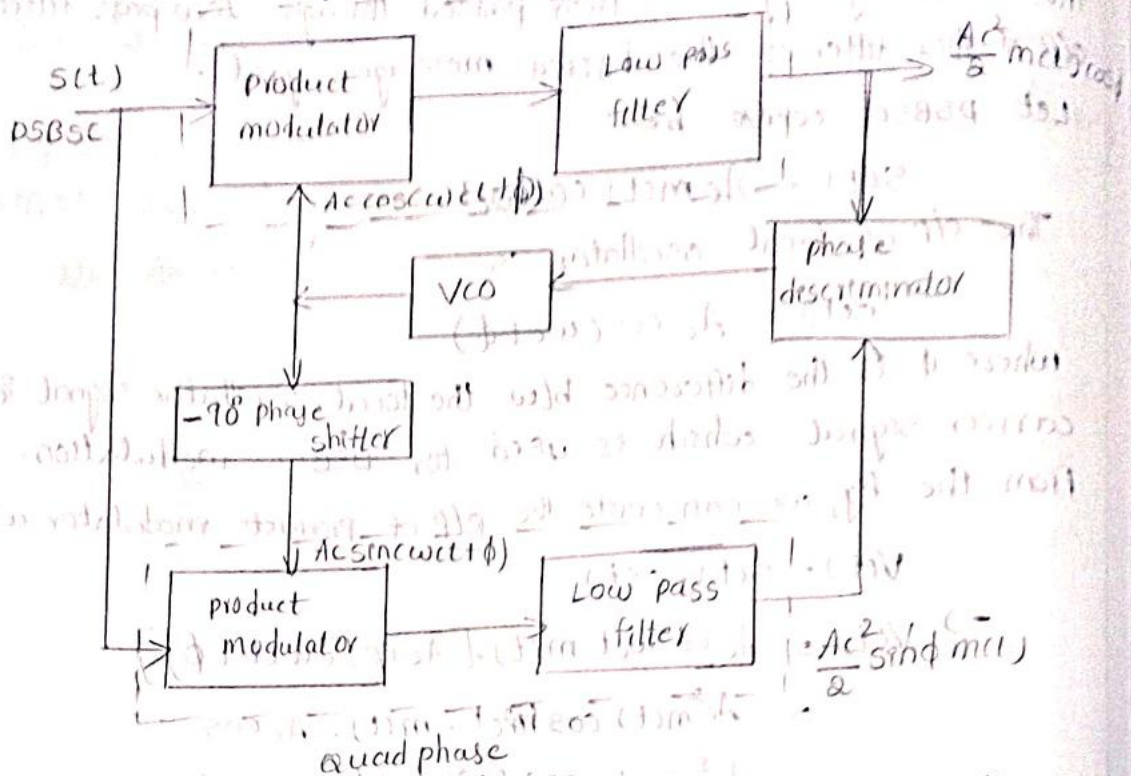
$$V_o(t) = \frac{A_c^2}{2} m(t) \cos \phi$$

The demodulated signal amplitude will be max, when $\phi = 0$. That's why the local oscillator signal & the carrier signal should be in phase.

The demodulated signal amplitude will be zero, when $\phi = \pm 90^\circ$.

This effect is called as "quadrature null effect".

COSTAS loop : - COSTAS loop is used to make both the carrier signal & the locally generated signal in phase. The following is the block diagram of costas loop.



costas loop consists of two product modulators with common i/p $S(t)$, which is DSBSC wave. The other i/p for both product modulators is taken from voltage controlled oscillator (VCO) with 90° phase shift to one of the product modulator as shown in fig.

WKT the equation of DSBSC wave is,

$$S(t) = A_c \cos(\omega_c t + \phi) \cdot m(t)$$

Let the o/p of VCO be,

$$c(t) = \cos(\omega_c t + \phi)$$

This o/p of VCO is applied as the carrier i/p of the upper product modulator.

Hence, the o/p of the upper product modulator is

$$V_1(t) = S(t) \cdot c(t)$$

$$= A_c \cos(\omega_c t + \phi) m(t)$$

$$= \frac{A_c}{2} m(t) [\cos(\omega_c t + \phi + \omega_c t) + \cos(\omega_c t + \phi - \omega_c t)]$$

$$= \frac{A_c}{2} m(t) [\cos(2\omega_c t + \phi) + \cos \phi]$$

$$= \frac{A_c}{2} m(t) [\cos(2\omega_c t + \phi) + \cos \phi]$$

$$= \frac{A_c}{2} \cos 2\omega_c t + \phi + \frac{A_c}{2} \cos \phi \cos m\omega_c t$$

The signal is applied as an i/p of the upper low pass filter. The o/p of this low pass filter is,

$$V_{o1}(t) = \frac{A_c}{2} \cos \phi \cos m\omega_c t$$

Therefore, the o/p of this low pass filter is the scaled version of message signal.

The o/p of 90° phase shifter is,

$$c_2(t) = \cos(\omega_c t + \phi - 90^\circ) = \sin(\omega_c t + \phi)$$

This signal is applied as the carrier i/p of the lower product modulator.

The o/p of lower product modulator is

$$V_2(t) = s(t) \cdot c_2(t)$$

$$V_2(t) = A_c \cos \omega_c t \cdot m(t) \cdot \sin(\omega_c t + \phi)$$

$$= \frac{A_c}{2} [\sin(\omega_c t + \omega_c t + \phi) + \sin(\omega_c t - (\omega_c t + \phi))] m(t) = \frac{A_c}{2} [\sin \phi \cos m\omega_c t + \sin(2\omega_c t + \phi) \cos m\omega_c t]$$

This signal is applied as an i/p of the LPF. The o/p of this low pass filter is,

$$V_{o2}(t) = \frac{A_c}{2} \sin \phi \cos m\omega_c t$$

The o/p of this LPF has 90° phase difference with the o/p of the upper LPF.

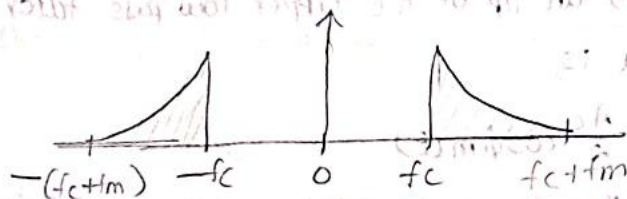
The o/p's of these two low pass filters are applied as i/p's of the phase discriminator. Based on the phase difference b/w these two signals, the phase discriminator produces a DC control signal.

This signal is applied as an i/p of VCO to correct the phase error in VCO o/p. Therefore the carrier signal & the locally generated signal are in phase.

$$\left[\frac{A_m \cos(\omega_c t + \phi) - A_m \sin(\omega_c t + \phi)}{2} \right] = \frac{A_m}{2} \cos(2\omega_c t + \phi)$$

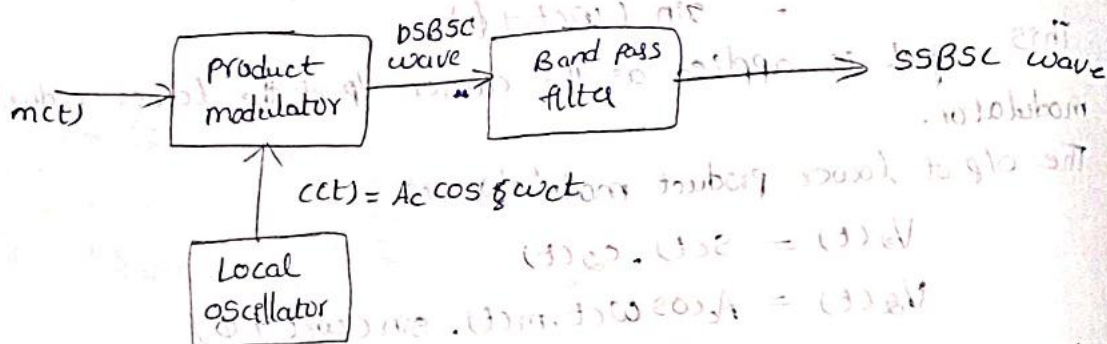
$$\left[\frac{A_m \cos(\omega_c t + \phi) + A_m \sin(\omega_c t + \phi)}{2} \right] = \frac{A_m}{2} \sin(2\omega_c t + \phi)$$

Frequency spectrum with reference



Frequency discrimination method for generation of SSB-SC:

The following fig shows the block diagram of SSB-SC modulator using frequency discrimination method.



(In this method, first we will generate DSB-SC wave with the help of the product modulator.)

$$\text{Let } m(t) = A_m \cos \omega_m t$$

$$c(t) = A_c \cos \omega_c t$$

$$\text{DSB-SC} = m(t) \cdot c(t)$$

$$= A_m \cos \omega_m t \cdot A_c \cos \omega_c t$$

$$= \frac{A_m A_c}{2} [\cos(\omega_c - \omega_m)t + \cos(\omega_c + \omega_m)t]$$

$$= \frac{A_m A_c}{2} [\cos(\omega_c - \omega_m)t + \cos(\omega_c + \omega_m)t]$$

Then the o/p of DSB-SC wave is applied as the i/p to the band pass filter. This band pass filter produces an o/p, which is SSB-SC wave.

If LSB is chosen for transmission then SSB-SC wave is given by,

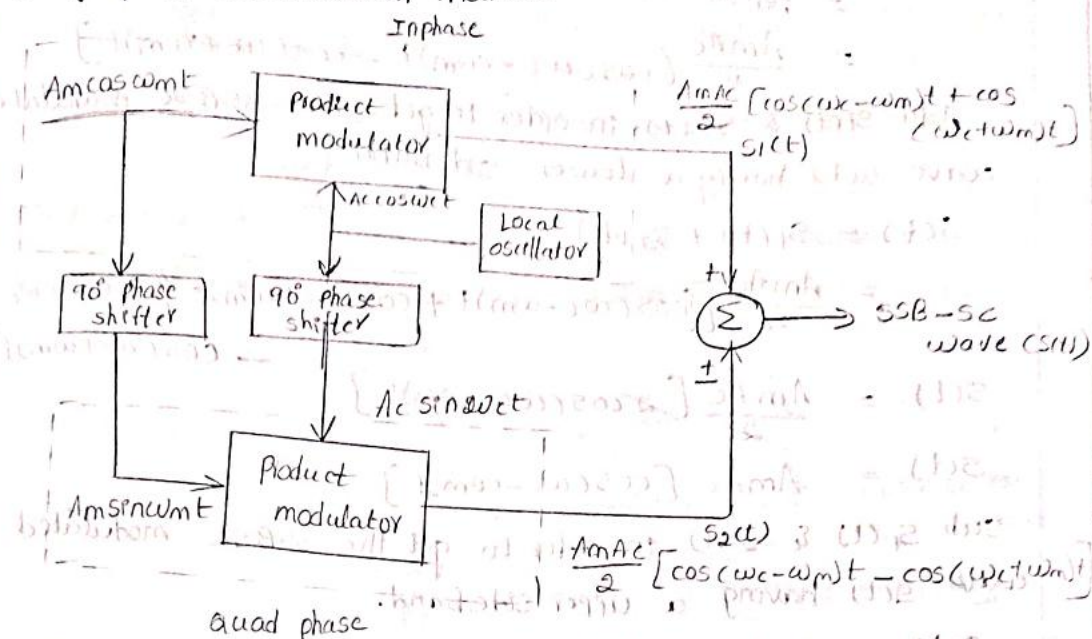
$$\text{SSB-SC} = \frac{A_m A_c}{2} [\cos(\omega_c - \omega_m)t]$$

If USB is chosen for transmission then SSB-SC wave is given by,

$$\text{SSB-SC} = \frac{A_m A_c}{2} [\cos(\omega_c + \omega_m)t]$$

phase discrimination method :

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



The block diagram consists of two product modulators, two 90° phase shifters, one local oscillator & one summer block. The product modulator produces an o/p, which is the product of two i/p's. The 90° phase shifter produces an o/p, which has phase of 90° with respect to i/p.

The local oscillator is used to generate the carrier signal. summer block produces an o/p, which is either the sum of two i/p's or the difference of two i/p's. based on polarity of i/p's.

The message signal $A_m \cos \omega_m t$ & the carrier signal $A_c \cos \omega_c t$ are directly applied as i/p's to the upper product modulator. so, the upper product modulator produces an o/p, which is the product of these two i/p's.

The o/p of upper product modulator is.

$$S_1(t) = A_m \cos \omega_m t \cdot A_c \cos \omega_c t$$

$$S_1(t) = \frac{A_m A_c}{2} [\cos(\omega_c - \omega_m)t + \cos(\omega_c + \omega_m)t]$$

The message signal and the carrier signal are shifted by 90° before applying as i/p's to the lower product modulator. so, the lower product modulator produces an o/p, which is the product of these two i/p's.

The o/p of lower product modulator is

$$\begin{aligned} S_2(t) &= A_m A_c \cos(\omega_c t - 90^\circ) \cos(\omega_c t - 90^\circ) \\ &= A_m A_c \sin \omega_c t \cdot \sin \omega_c t \\ &= \frac{A_m A_c}{2} [\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t] \end{aligned}$$

Add $S_1(t)$ & $S_2(t)$ in order to get the SSBSC modulated wave $S(t)$ having a lower sideband.

$$S(t) = S_1(t) + S_2(t)$$

$$= \frac{A_m A_c}{2} [\{\cos(\omega_c - \omega_m)t + \cos(\omega_c + \omega_m)t\} + \{\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t\}]$$

$$S(t) = \frac{A_m A_c}{2} [2\cos(\omega_c - \omega_m)t]$$

$$S(t) = A_m A_c [\cos(\omega_c - \omega_m)t]$$

Sub $S_1(t)$ & $S_2(t)$ in order to get the SSBSC modulated wave $S(t)$ having a upper sideband.

$$S(t) = S_1(t) - S_2(t)$$

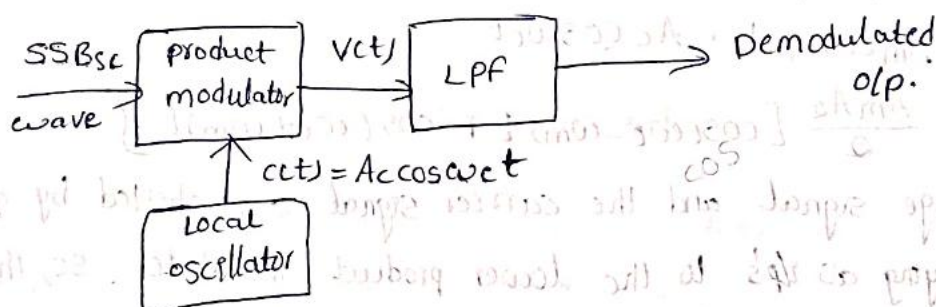
$$\begin{aligned} &= \frac{A_m A_c}{2} [\{\cos(\omega_c - \omega_m)t + \cos(\omega_c + \omega_m)t\} - \{\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t\}] \\ &= \frac{A_m A_c}{2} [2\cos(\omega_c + \omega_m)t] \end{aligned}$$

$$S(t) = A_m A_c [\cos(\omega_c + \omega_m)t]$$

Hence, by properly choosing polarities of i/p's at summer block, we will get SSBSC wave having a upper side band or a lower sideband.

Detection of SSB-SC using coherent detector:

Here, the same carrier signal is used to detect the message signal. Hence, this process of detection is called as coherent or synchronous detection. Following is the block diagram of coherent detector.



In this process, the message signal can be extracted from SSBSC wave by multiplying it with a carrier, having the same frequency & the phase of the carrier used in SSBSC modulation. The resulting signal is then passed through a LPF. The o/p of this filter is the desired message signal.

consider the SSBSC wave, having a lower sideband.

$$s(t) = \frac{A_m A_c}{2} [\cos(\omega_c - \omega_m)t]$$

The o/p of local oscillator is, $c(t) = A_c \cos \omega_c t$

from the fig, the o/p of product modulator is

$$V(t) = s(t) \cdot c(t)$$

$$= \frac{A_m A_c}{2} [\cos(\omega_c - \omega_m)t] \cdot A_c \cos \omega_c t$$

$$= \frac{A_m A_c^2}{4} [\cos(\omega_c - \omega_m)t + \cos(\omega_c + \omega_m)t]$$

$$= \frac{A_m A_c^2}{4} [\cos \omega_m t + \cos(2\omega_c - \omega_m)t]$$

$$= \frac{A_m A_c^2}{4} [\cos \omega_m t] + \frac{A_m A_c^2}{4} [\cos(2\omega_c - \omega_m)t]$$

In the above equation, the 1st term is the scaled version of message signal. It can be extracted by passing the above signal through a LPF.

Therefore, the o/p of LPF is,

$$V_o(t) = \frac{A_m A_c^2}{4} (\cos \omega_m t)$$

→ consider the SSBSC wave having a upper side band.

$$s(t) = \frac{A_m A_c}{2} [\cos(\omega_c + \omega_m)t]$$

The o/p of local oscillator is, $c(t) = A_c \cos \omega_c t$

from the fig, the o/p of product modulator is,

$$V(t) = s(t) \cdot c(t)$$

$$= \frac{A_m A_c}{2} [\cos(\omega_c + \omega_m)t] \cdot A_c \cos \omega_c t$$

$$= \frac{A_m A_c^2}{4} [\cos(\omega_c + \omega_m)t + \cos(\omega_c - \omega_m)t]$$

$$= \frac{A_m A_c^2}{4} [\cos \omega_m t + \cos(2\omega_c + \omega_m)t]$$

In the above equation, the 1st term is the scaled version of message signal. It can be extracted by passing the above equation through a LPF.

The o/p of LPF is

$$V_o(t) = \frac{A_m A_c^2}{4} [\cos \omega_m t]$$

Therefore, we get the same demodulated o/p in both the cases by using coherent detector.

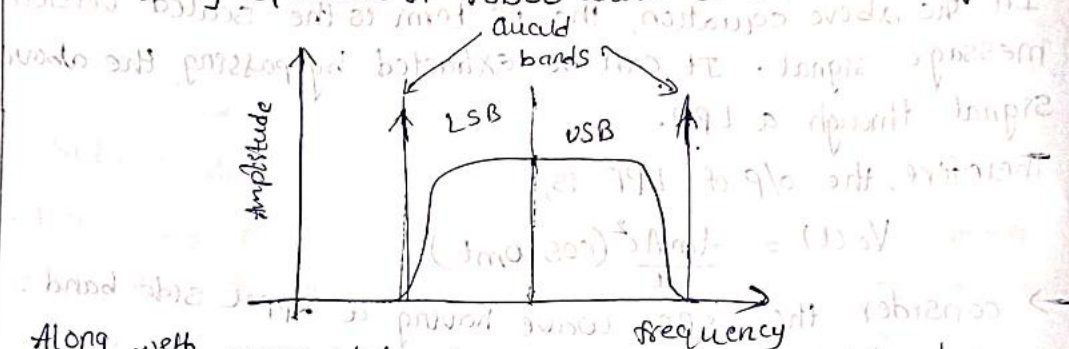
Principle of Vestigial side band modulation:

SSBSC modulated signal has only one side band frequency.

Theoretically, we can get one side band frequency component completely by using an ideal bandpass filter. However, practically we may not get the entire sideband frequency component. Due to this, some information gets lost.

To avoid this loss, a technique is chosen, which is a compromise b/w DSBSC & SSBSC. This technique is known as Vestigial side band suppressed carrier (VSBSC) technique.

VSBSC modulation is the process, where a part of the signal called as Vestige is modulated along with one sideband. The frequency spectrum of VSBSC wave is shown in fig.



Along with upper side band, a part of lower sideband is also being transmitted in this technique. Similarly, we can transmit the lower sideband along with a part of USB. A small band of very small width is laid on either side of VSB in order to avoid the interference. VSB is mostly used in TV transmissions.

Bandwidth: WKT the bandwidth SSBSC wave is f_m . Since the VSBSC modulated wave contains the frequency components of one side band along with vestige of another sideband, the B.W of it will be the sum of SSBSC modulated wave & vestige frequency f_v .

$$\Rightarrow f_m + f_v$$

Advantages :

1. Highly efficient.
2. Filter design is easy.
3. Possesses good phase characteristics.
4. Reduction in B.W. compared to AM & DSBSC wave.

Disadvantages :

1. B.W. is more when compared to SSBSC wave.
2. Demodulation is complex.

2. Angle modulation

Angle modulation is another method of modulating the sinusoidal carrier signal.

In angle modulation either frequency or phase of the carrier is varied according to the message signal, but in angle modulation carrier amplitude is kept constant.

Some of the modulation system is not linear & superposition doesn't apply. Such type of system is connected with a constant amplitude & phase angle is responsible for transmitting the signal.

Angle modulation is mainly classified into 2 types i.e.,

1. Frequency modulation

2. Phase modulation.

Frequency modulation: Frequency of the carrier signal is varied in accordance with the instantaneous amplitude of message signal.

Phase modulation: It is defined as the process of varying the phase of the carrier signal in accordance with the instantaneous amplitude of message signal.

→ The equation of angle modulation is given by,

$$V(t) = A \cos(\omega_c t + \phi)$$

where, A = amplitude

ω_c = angular frequency

ϕ = phase

t = time

Advantages

1. Noise reduction compared to AM.
2. Less use of power
3. It is more efficient
4. Improved system fidelity.

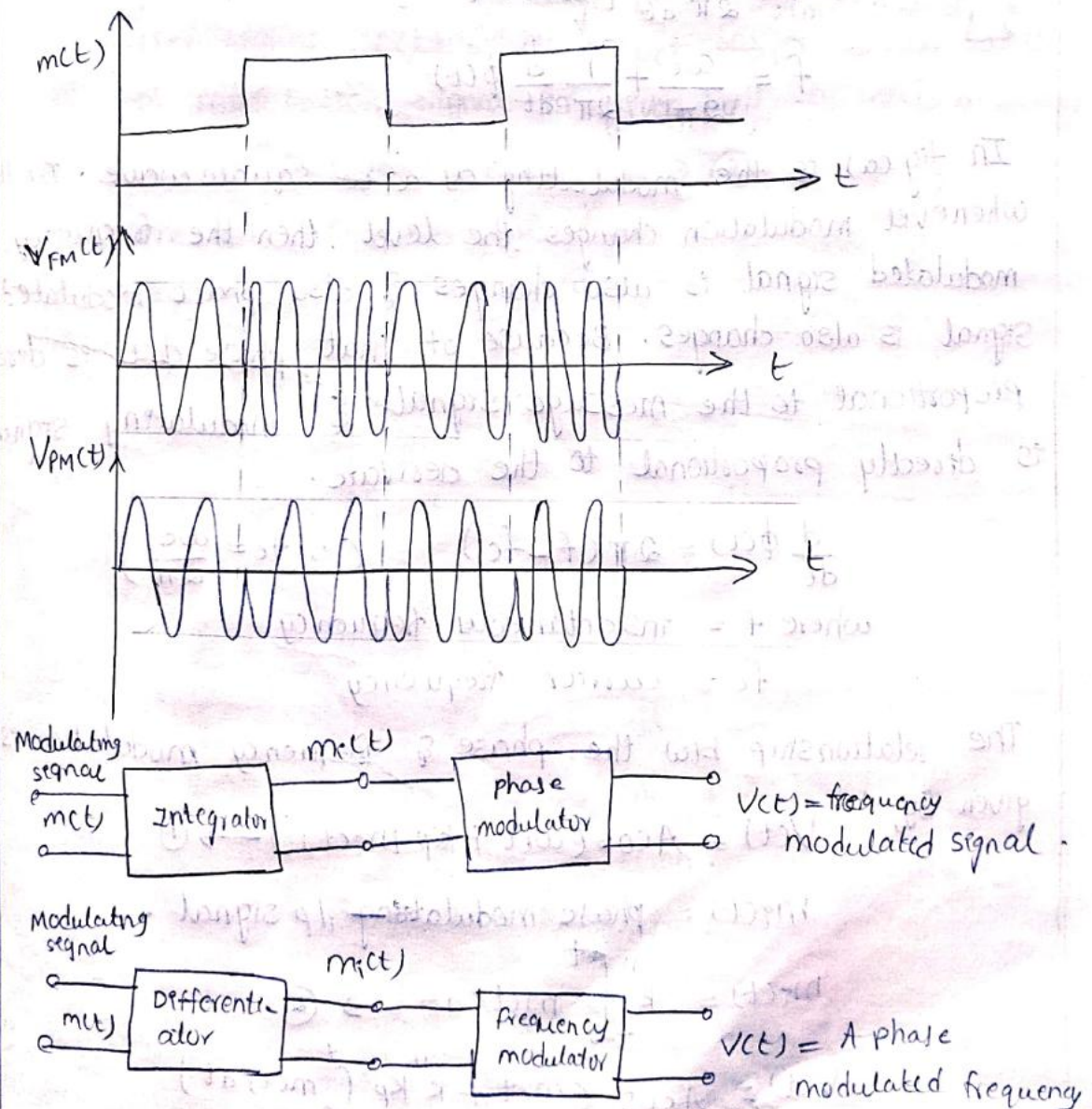
Disadvantages :

1. Increased bandwidth
2. It can be used in more complex ckt's.

Applications :

1. Radio broadcasting
2. satellite communications
3. Microwave communication
4. cellular radio
5. TV sound transmission
6. Two way mobile radio.

→ In angle modulation it is mainly used to increase the performance of radio communication.



The equation of angle modulation is given by,

$$V(t) = A \cos(\omega_c t + \phi)$$

But in sinusoidal $A \cos \omega_c t$ can be written as

$$A \cos \omega_c t = \text{real part of } (A e^{j\omega_c t})$$

If $\theta = \omega_c t$ (or) $2\pi f_c t$

then the phasor rotates in counter clock wise direction with an angular velocity of ω_c .

$A e^{j\theta}$ is a complex plane by phasor of length A .

Angle θ is measured counter clockwise from the real axis.

$V(t)$ is modulation around the angle $\theta = \omega_c t$.

Instantaneous frequency modulation is given by

$$f = \frac{1}{2\pi} \frac{d}{dt} [\omega_c t + \phi(t)]$$

$$f = \frac{\omega_c}{2\pi} + \frac{1}{2\pi} \frac{d}{dt} \phi(t)$$

In fig (a) is the modulating as a square wave. In this whenever modulation changes the level then the frequency modulated signal is also changes & also phase modulated signal is also changes. Because of that phase $\phi(t)$ is directly proportional to the message signal. & modulating signal is directly proportional to the derivate.

$$\frac{d}{dt} \phi(t) = 2\pi(f - f_c) \quad (\because f_c = \frac{\omega_c}{2\pi})$$

where f = instantaneous frequency

f_c = carrier frequency

The relationship b/w the phase & frequency modulation is given by,

$$V(t) = A \cos(\omega_c t + K_f m(t)) \rightarrow (1)$$

$m(t)$ = phase modulating signal.

$$m(t) = K \int_{-\infty}^t m(t) dt \rightarrow (2)$$

$$V(t) = A \cos(\omega_c t + K_f \int_{-\infty}^t m(t) dt)$$

Let $K_f = K_f$

$$V(t) = A \cos(\omega_c t + K_f \int_{-\infty}^t m(t) dt)$$

The instantaneous frequency is directly proportional to the modulation signal.

The combination of integrator & phase modulator produces frequency modulated o/p.

The combination of differentiator & frequency modulator produces a phase modulated o/p.

Single tone frequency modulation (or) Time domain for FM:

A single tone frequency modulation is a signal that consists of single frequency & it is represented graphically by sinusoidal wave.

The time domain equation for frequency modulation is

$$m_f(t) = A \cos \left[\omega_c t + k_f \int_{-\infty}^t m(t) dt \right]$$

proof: Let us consider message $m(t) = A_m \cos \omega_m t$ & carrier signal $c(t) = A_c \cos \omega_c t$ $\rightarrow$ ②

The FM modulation signal is given by,

$$S_{FM}(t) = A \cos \phi_i \rightarrow$$
 ③

where ϕ_i = instantaneous phase

The relationship b/w instantaneous frequency & instantaneous phase is defined by,

$$\omega_i = \frac{d\phi_i}{dt} \quad (\omega_i = \text{instantaneous freq})$$

$$\phi_i = \int \omega_i dt$$

$$\text{Let } \omega_i = \omega_c + k_f m(t)$$

$$\phi_i = \int (\omega_c + k_f m(t)) dt$$

$$\phi_i = \omega_c t + k_f \int m(t) dt$$

$$\phi_i = A_c \cos \left[\omega_c t + k_f \int m(t) dt \right]$$

Sub $m(t)$ value in above equation,

$$\phi_i = A_c \cos \left[\omega_c t + k_f \int A_m \cos \omega_m t dt \right]$$

$$\phi_i = A_c \cos \left[\omega_c t + \frac{k_f A_m}{\omega_m} \sin \omega_m t \right]$$

$$\phi_i = A_c \cos \left[\omega_c t + \frac{\Delta f}{\omega_m} \sin \omega_m t \right]$$

Spectrum analysis of sinusoidal FM wave by using Bessel function

The spectrum analysis of sinusoidal FM wave using Bessel function is used in tone modulated FM signal. A tone is used for to create a specific response or a tone is nothing but voice in communication is defined as the way a person speak to someone. The frequency spectrum of angle modulated signal is modulated by using single sinusoidal signal or tone.

If modulation frequency is held constant then the modulation index is increased.

The angle modulated signal can be written as,

$$V(t) = \cos(\omega_c t + \beta \sin \omega_m t) \rightarrow (1)$$

where $\sin \omega_m t$ = Tone signal

β = Bessel function or modulation index

In the above equation if the term $\sin \omega_m t$ is used for phase modulation then β is replaced by $\beta_p = K_p$.

Similarly, for frequency modulation, then β is replaced by $\beta_f = K_f$.

The term β_p & β_f is replaced by β .

$$\cos(\omega_c t + \beta \sin \omega_m t) = \cos \omega_c t \cdot \cos(\beta \sin \omega_m t) - \sin \omega_c t \sin(\beta \sin \omega_m t)$$

$$\Rightarrow \cos(\beta \sin \omega_m t) = J_0(\beta) + 2J_2(\beta) \cos 2\omega_m t + 2J_4(\beta) \cos 4\omega_m t + \dots + 2J_{2n}(\beta) \cos 2n\omega_m t$$

$$\Rightarrow \sin(\beta \sin \omega_m t) = 2J_1(\beta) \sin \omega_m t + 2J_3(\beta) \sin 3\omega_m t + 2J_5(\beta) \sin 5\omega_m t + \dots + 2J_{2n-1}(\beta) \sin (2n-1)\omega_m t$$

$$\rightarrow \cos(\omega_c t + \beta \sin \omega_m t) = \cos \omega_c t \left[J_0(\beta) + 2J_2(\beta) \cos 2\omega_m t + 2J_4(\beta) \cos 4\omega_m t + \dots \right] - \sin \omega_c t \left[2J_1(\beta) \sin \omega_m t + 2J_3(\beta) \sin 3\omega_m t + 2J_5(\beta) \sin 5\omega_m t + \dots \right]$$

$$= \left[\cos \omega_c t \cdot J_0(\beta) + 2J_2(\beta) \cos \omega_c t \cdot \cos 2\omega_m t + 2J_4(\beta) \cos \omega_c t \cdot \cos 4\omega_m t + \dots \right]$$

$$- \left[2J_1(\beta) \sin \omega_c t \cdot \sin \omega_m t + 2J_3(\beta) \sin \omega_c t \cdot \sin 3\omega_m t + 2J_5(\beta) \sin \omega_c t \cdot \sin 5\omega_m t + \dots \right]$$

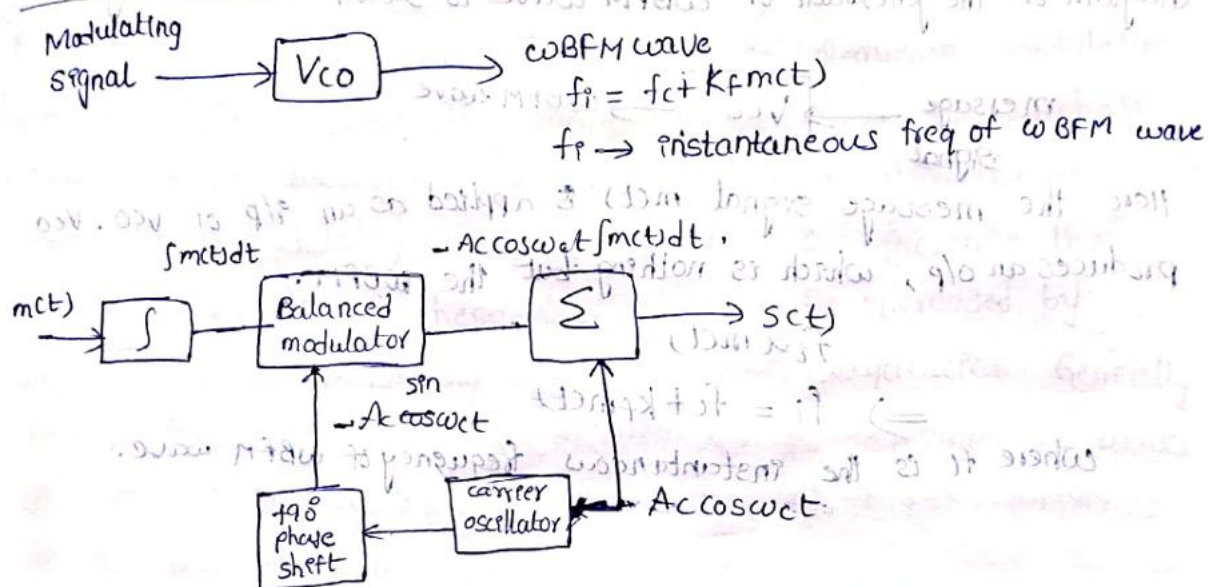
$$= \left[J_0(\beta) \cos \omega_c t + 2J_2(\beta) [\cos(\omega_c t - 2\omega_m t) + \cos(\omega_c t + 2\omega_m t)] + 2J_4(\beta) [\cos(\omega_c t - 4\omega_m t) + \cos(\omega_c t + 4\omega_m t)] - [2J_1(\beta) \sin(\omega_c t - \omega_m t) + 2J_3(\beta) \sin(\omega_c t - 3\omega_m t) - \cos(\omega_c t + \cos 3\omega_m t) - \cos(\omega_c t + \cos 5\omega_m t) - \dots] \right]$$

Types of FM :

There are 2 types of FM

1. wide band FM
2. Narrow band FM

wide band FM :



NBFM

$\beta > 1$

→ WBFM is used in FM broadcasting in which music & speech are transmitted upto 75kHz deviation.

In frequency modulation is used in audio communication & data transfer & FM have a infinite bandwidth.

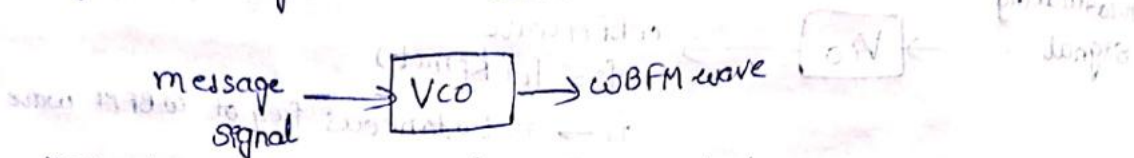
when efficiency is important in that time we are choosing NBFM when signal has a better quality in that time we are using WBFM.

& it is carried bandwidth upto 20kHz. It is used in FM radio, TV, frequency broadcasting & audio communication. WBFM is mainly used in Voltage controlled oscillator. The purpose of Vco is to control the voltage & it provides the frequency along with message signal. WBFM is used for transmitting higher data. The o/p of WBFM wave is $f_i = f_c + K_f m(t)$

NBFM : These frequency modulation has small bandwidth compared to WBFM. there modulating index is less than 1 ($\beta < 1$). The NBFM consists of one carrier & along with sidebands. The frequency range of NBFM is 30Hz to 3KHz. The o/p of NBFM

wave is $\text{Accos}(\omega t - A \sin \omega t f_m(t)) dt$

WBFM: This method is called as the direct method because we are generating a wide band FM wave directly. In this method, Voltage controlled oscillator (VCO) is used to generate WBFM. VCO produces an o/p signal, whose frequency is proportional to the i/p signal voltage. This is similar to definition of FM wave. The block diagram of the generation of WBFM wave is shown in below fig.



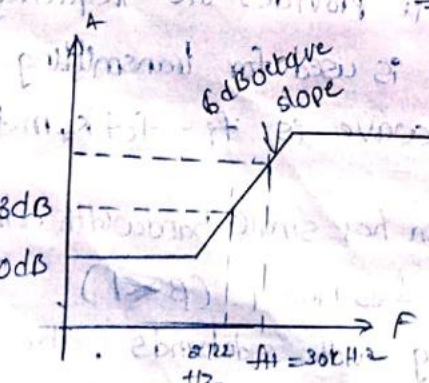
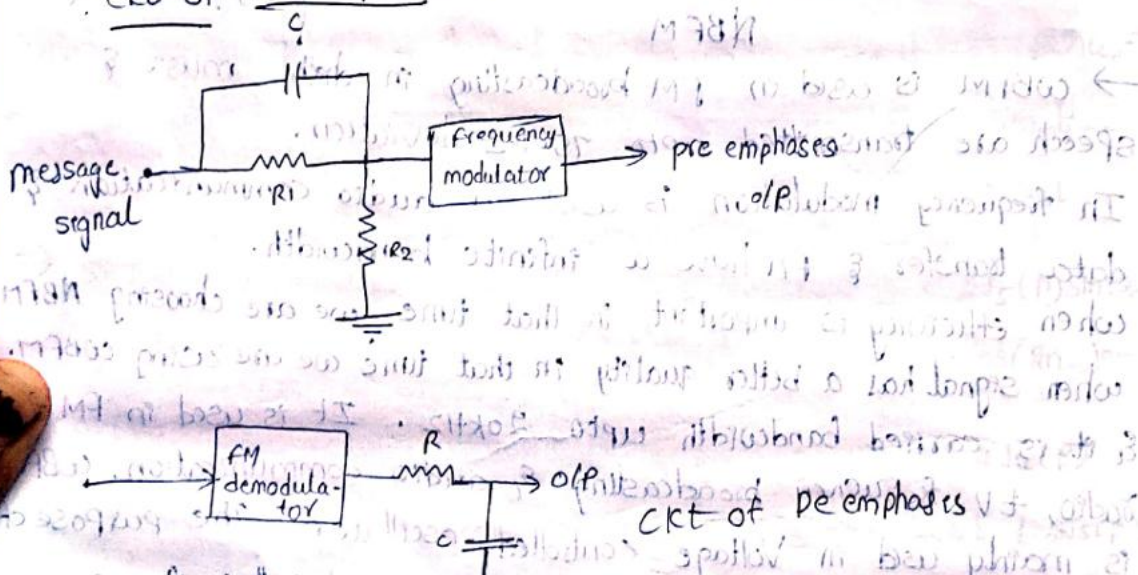
Here the message signal $m(t)$ is applied as an i/p of VCO. VCO produces an o/p, which is nothing but the WBFM.

$$f_i \propto m(t)$$

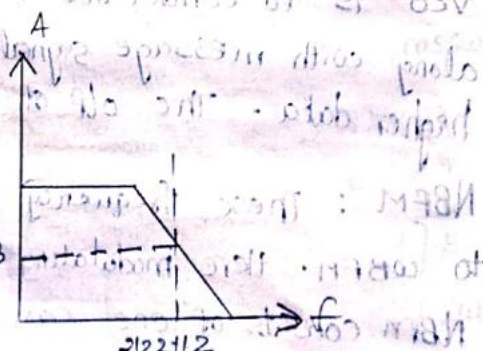
$$\Rightarrow f_i = f_c + k_f m(t)$$

where f_i is the instantaneous frequency of WBFM wave.

Ckt of Pre-emphasis:



(a) Pre-emphasis



(b) De-emphasis

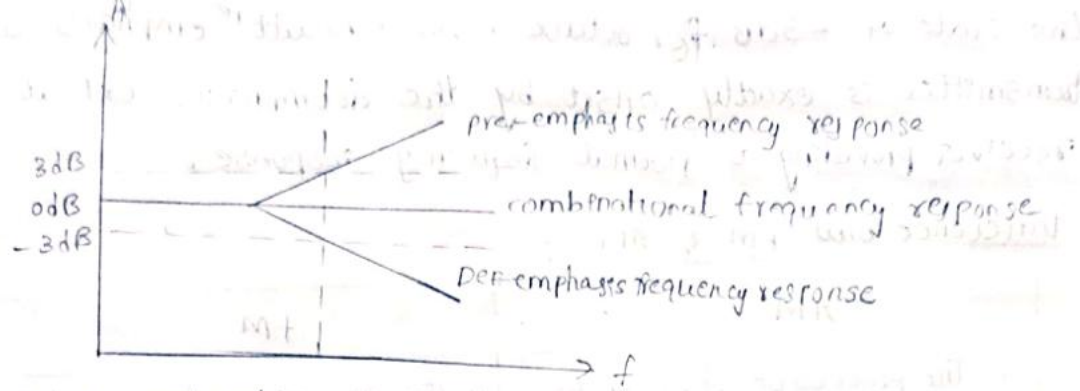


fig: combined curve of pre-emphasis and de-emphasis

Pre-emphasis and de-emphasis are used in frequency modulation. Pre-emphasis are used at transmitter side and de-emphasis are used at receiver side for higher modulating frequencies.

These higher modulating frequency are used ^{to} increase the frequency deviation & these deviation can be increased by amplitude of the modulating signal at higher frequencies. Generally,

Pre-emphasis works at high frequencies & de-emphasis works at low frequency (below 2kHz).

In ckt of pre-emphasis at the transmitter the modulating signal is passed through a simple N/w which amplifies high frequency more than low frequency components. That's why we are taking HPF. The

combination of resistor & capacitor giving time constant $t = RC$

Here the time constant of 75 μs . such type of ckt the cutoff frequency is $f_{co} = 2122 Hz$ that means when the frequency is higher than 2122 Hz will be linearly enhanced. The o/p amplitude

increases with frequency at a rate of 6dB per octave.

The preemphasis ckt increases then the higher frequency signal is also increases. This ckt is also used for to improve the signal to noise ratio. The pre-emphasis upper break frequency is

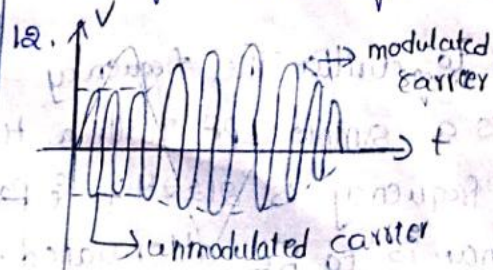
given by,

$$f_u = \frac{R_1 + R_2}{2\pi R_1 R_2 C}$$

A De-emphasis ckt is used for to written the frequency response to its normalized. This is a simple LPF with a time constant of 75 μs . Its cutoff frequency is 2122 Hz & ~~passer~~ causes the signal above this frequency is to be attenuated.

the rate of -3dB per octave. As a result^{re} emphasis at the transmitter is exactly offset by the deemphasis ckt at the receiver providing a normal frequency response.

Difference b/w FM & AM :

AM	FM
1. The process of changing the amplitude of carrier signal in accordance with instantaneous amplitude of message signal is called as "Amplitude modulation."	1. The process of changing the frequency of carrier signal in accordance with instantaneous amplitude of message signal is called as "Frequency modulation."
2. In AM, amplitude is varying	2. In FM, amplitude is constant
3. Noise is more	3. Noise is less
4. In AM, the frequency spectrum consists of only two side bands.	4. In FM, there is infinite no. of side bands
5. Bandwidth = $2f_m$	5. Bandwidth = $2(\Delta f + f_m)$
6. Modulation index = $\frac{\Delta V_m}{V_c}$	6. Modulation index = $\frac{\Delta f}{f_m}$
7. Bandwidth is less	7. Bandwidth is large
8. No. of channels can't be transmitted through a single frequency.	8. No. of channels can be transmitted through a single frequency.
9. It is less complex	9. It is more complex.
10. Ground wave & sky wave propagation is used to transmit the signals.	10. Space wave propagation is used to transmit the signals.
11. Information is stored in amplitude frequency of carrier signal.	11. Information is stored in frequency amplitude of carrier signal.
12. 	12. 

13. Radio, TV broadcasting, Police wireless, cabs etc.
14. Transmitted power is constant

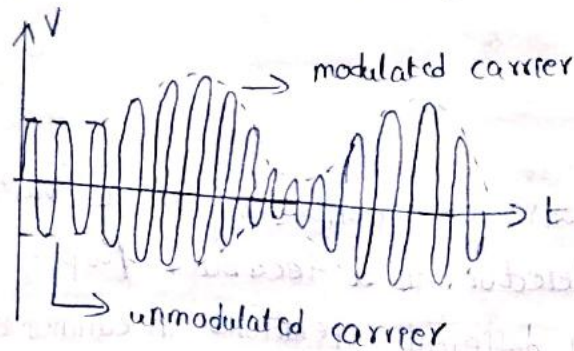
13. TV broadcasting, audio, Video etc.
14. Transmitted power is Varied with modulation index

constant average power in angle modulation :

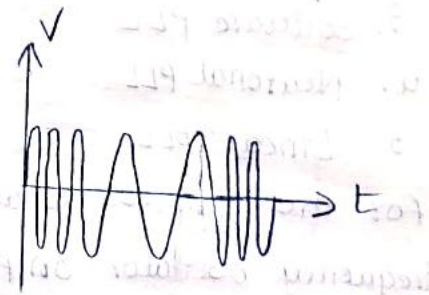
The relationship b/w the power, Voltage & resistor is given by,

$$P = \frac{V^2}{R}$$

The wave-forms of AM & FM waves - is shown below.



Modulated wave of AM



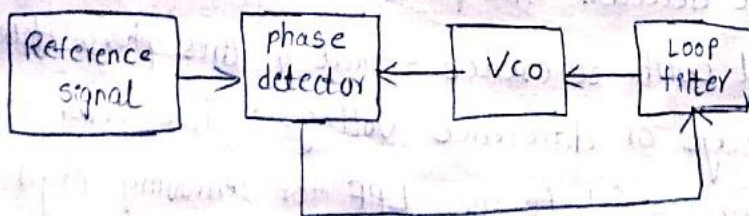
Modulated wave of FM

In AM the amplitude is varying & for FM wave the frequency is kept constant.

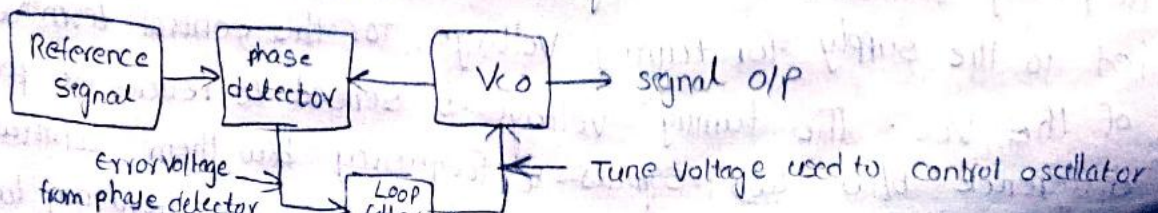
The information can fall on the amplitude of message signal, we can loss the information in amplitude modulation.

→ In frequency modulation, the amplitude of message signal is constant. since we can't loss any information.

PLL angle modulation :



(or)



PLL means phased locked loop. PLL is an electronic ckt with a voltage ~~of~~ controlled oscillator that constantly adjust to match frequency of an i/p signal. PLL is also defined as the control system which generates an o/p signal whose phase is related to the phase of the i/p signal. PLL are used to generate, stabilize, modulate, demodulate, filter or recover a signal from noise where data has been interrupted. There are several different types of PLL i.e.,

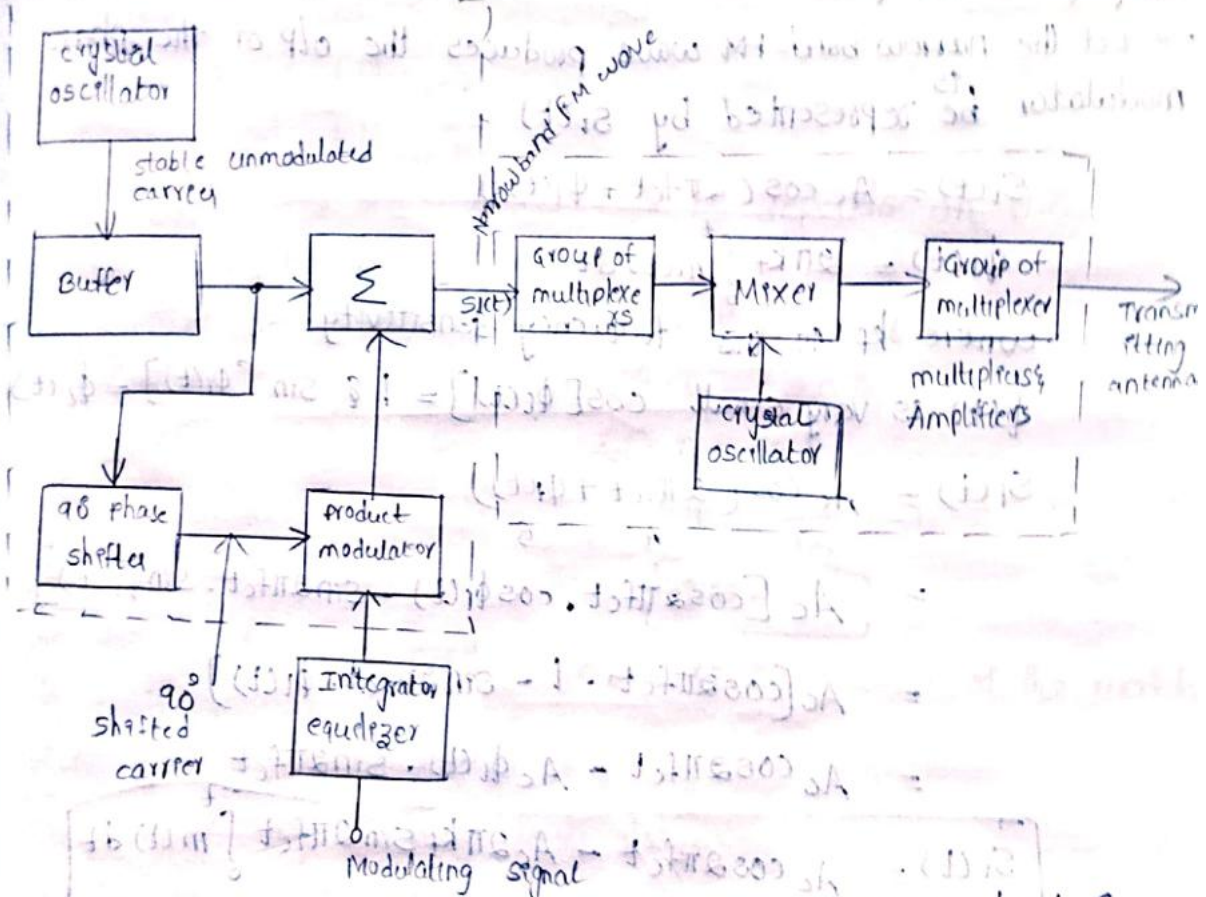
1. Analog PLL
2. Digital PLL
3. Software PLL
4. Neuronal PLL
5. Linear PLL

For the simplest is an electronic ckt consisting of a variable frequency oscillator or phase detector in a feedback loop. PLL is also used for to control different operations in communication system such as computers & electronic applications. The major concept behind the PLL operation is comparison of phases of two signals. The phase difference b/w i/p & o/p signals can be used for controlling the loop frequency. In many communication systems PLL is used for different purposes to synchronize the two signals with different frequencies & also to remove the large noise. The basic PLL consists of phase detector, VCO, & loop filter.

The VCO of PLL produces a signal & these signal from the VCO is given to the phase detector. The phase detector is used for to compare these signal with reference signal & this phase detector produces error voltage or difference voltage. The error signal of the phase detector is fed to the LPF for removing high frequency elements of the signal. The o/p of the loop filter is fed to the supply for tuning voltage for the control terminal of the VCO. The tuning voltage is sensed to reduce the phase difference b/w two signals & frequency b/w them. Initially PLL doesn't lock & error voltage drags the VCO frequency towards

the reference until the error can't be reduced any further. If PLL is locked, then the steady state error voltage will be produced. steady state error represents that there is no phase difference b/w reference signal & V_{CO} . Thus, we can say the frequency of two signals is exactly same.

Generation of FM signal (Armstrong method)



To generate FM signal by using two methods i.e. direct & indirect method.

- In direct method of generation of FM LC oscillator is used.
- The LC oscillator are not stable for the communication or the broadcast purpose. so the direct method can't be used for broadcast application.
- The alternate method is to use the indirect method is called as "armstrong method of FM generation."
- In these method FM is obtained through phase modulation.
- A crystal oscillator can be used, hence the frequency stability is very high & these method widely used in practice.
- The armstrong method uses the phase modulator to generate a

frequency modulated wave.

→ The operation is divided into 2 parts i.e. part-1 is used to generate a narrow band FM wave using a ^{product} phase modulator and part-2 use the frequency multipliers & mixers to obtain the required values of frequency deviation, carrier, modulation index.

part-1:

The modulating signal $m(t)$ is passed through a ^{product} integrator before applying to the phase modulator.

→ Let the narrow band FM wave produces the o/p of the ^{product} phase modulator be represented by $S_1(t)$

$$S_1(t) = A_c \cos(2\pi f_c t + \phi_1(t))$$

$$\phi_1(t) = 2\pi k_f \int_0^t m(t) dt$$

where k_f is the frequency sensitivity

$\phi(t)$ is very small $\cos[\phi_1(t)] = 1$ & $\sin[\phi_1(t)] = \phi_1(t)$

$$S_1(t) = A_c \cos(2\pi f_c t + \phi_1(t))$$

$$= A_c [\cos 2\pi f_c t \cdot \cos \phi_1(t) - \sin 2\pi f_c t \cdot \sin \phi_1(t)]$$

$$= A_c [\cos 2\pi f_c t \cdot 1 - \sin 2\pi f_c t \cdot \phi_1(t)]$$

$$= A_c \cos 2\pi f_c t - A_c \phi_1(t) \cdot \sin 2\pi f_c t$$

$$S_1(t) = A_c \cos 2\pi f_c t - A_c 2\pi k_f \sin 2\pi f_c t \int_0^t m(t) dt$$

Transmission B.W of FM wave:

Transmission B.W of FM wave is having two cases i.e.

1. Theoretical case

2. practical case

In theoretical case the B.W of FM signal is infinite since it contains infinite no. of side bands.

B.W $\propto$ Infinite

In practical cases the FM wave is effectively limited to a

finite no. of significant side bands.

B.W $\propto$ finite

index. i.e., $B.W \propto m_f$.

According to the carson's rule the transmission B.W of fm wave is

$$B.W = 2(\Delta f + f_m) \text{ Hz}$$

where Δf is ^{peak} frequency deviation ($\Delta f = m_f \cdot f_m$)

f_m is modulating frequency

The carson's rule approximate ^{the} B.W necessary to transmission frequency modulated.

Frequency modulation is classified 2 types i.e.,

1. Narrow band FM

2. wide band FM

In both frequency modulations will be depends on B.W

Bandwidth of narrow band FM :

In NBFM the modulation index is very small i.e., $m_f \ll 1$

According to the carson's rule $B.W = 2(\Delta f + f_m) \text{ Hz}$

WKT, ~~Δf~~ $m_f \ll 1$

$$m_f = \frac{\Delta f}{f_m} \ll 1 \Rightarrow \Delta f \ll f_m$$

$$B.W = 2f_m$$

In NBFM the bandwidth is equal to the twice of the modulating frequency.

B.W of wide band FM :

In WBFM, the B.W is very large

$$m_f \gg 1$$

$$m_f = \frac{\Delta f}{f_m} \gg 1$$

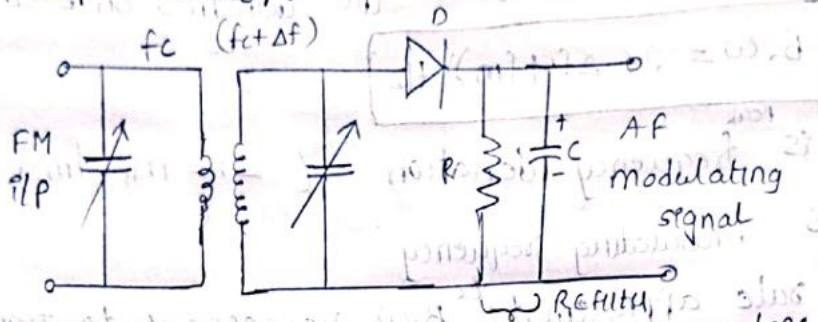
$$\Delta f \gg f_m$$

$$B.W = 2\Delta f$$

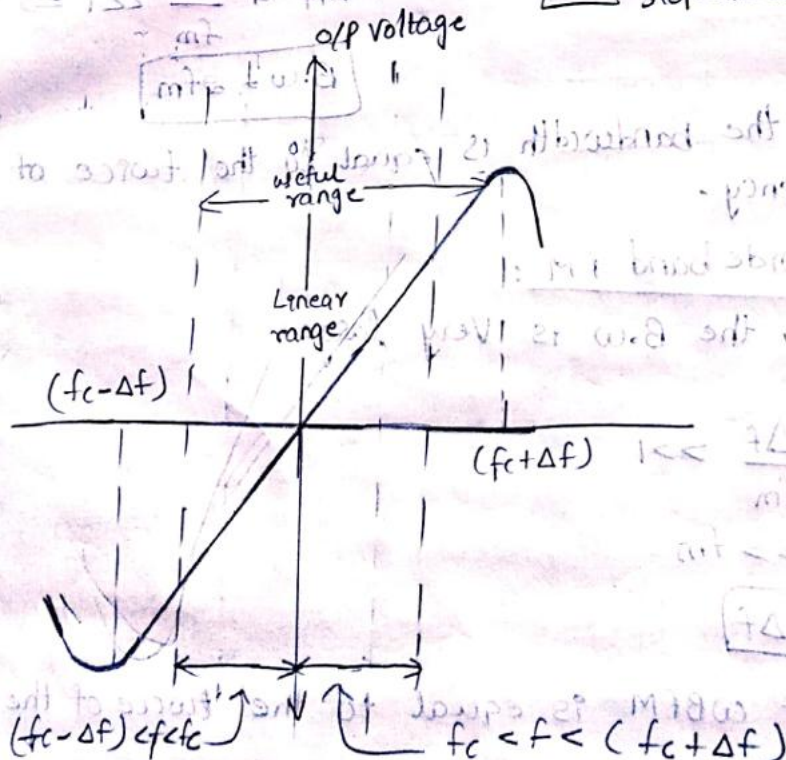
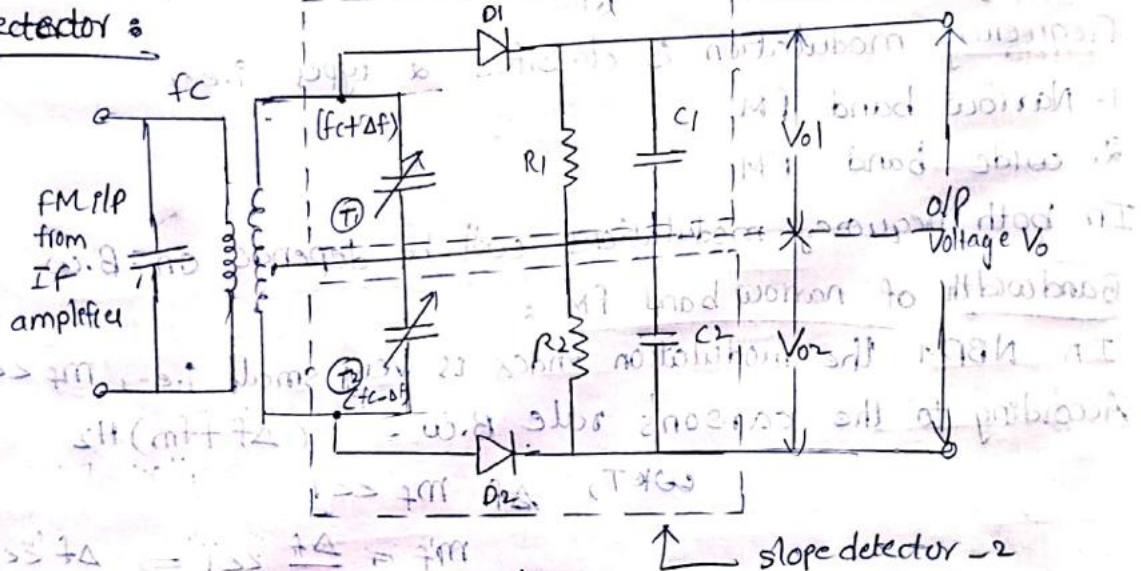
The B.W of WBFM is equal to the twice of the frequency deviation.

Detection of FM signal by using balanced slope detector:

Simple slope detector:



Balanced slope detector:



Detection of FM signal :

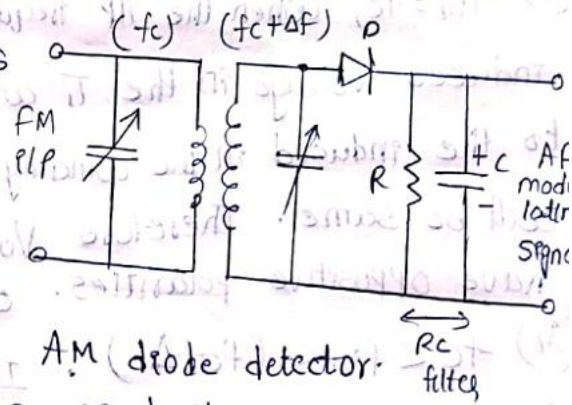
The demodulation process of FM waves is exactly opposite to the frequency modulation.

- The In demodulation, original modulating signal is getting at the demodulation o/p.
- The i/p of demodulator is FM wave.
- Basically AM detector is a rectifier. But FM demodulator is a basically frequency to amplitude converter.
- It converts the frequency variations in FM wave at the i/p into amplitude variations at its o/p to recover the original modulating signal.

Simple slope detector :

The ckt diagram of a simple slope detector is shown below.

- The o/p voltage of the tank ckt is then applied to a simple diode detector of an RC load with proper time constant.



- This detector is identical to the AM (diode detector).
- The basic slope detector is its simplicity.
- There are some drawbacks in this.
 - * It is inefficient
 - * It is linear only over a limited frequency range
 - * It is difficult to adjust the primary windings of the transformer must be tuned to slightly different frequency.
- * To overcome the drawbacks of the simple slope detector a balanced modulator is used.

Balanced slope detector :

The ckt diagram of balanced slope detector is shown in figure.

- In this ckt the balanced slope detector consists of two slope detector ckt.
- The i/p transformer has a center tapped transformer because of these reason the i/p voltages to the two slope detectors are 180° out of phase.

- There are three tuned ckt's
- primary is tuned to f_c i.e., f_c & the upper tuned ckt of the secondary (T_1) is tuned above f_c by Δf i.e., its resonant frequency is $(f_c + \Delta f)$, similarly the lower tuned ckt of secondary is tuned below f_c by Δf at $(f_c - \Delta f)$.
- R_{1C_1} & R_{2C_2} are the filters used to bypass the RF ripple.
- V_{o1} & V_{o2} are the o/p Voltage of two slope detectors.
- The final o/p Voltage is obtained by taking the subtraction of individual o/p Voltage V_{o1} & V_{o2}

$$V_o = V_{o1} - V_{o2}$$

- In this ckt operation by dividing the i/p frequency into three ranges

(i) $f_{in} = f_c$, when the i/p frequency is equal to f_c , then the induced voltage in the T_1 winding of secondary is exactly equal to the induced in the winding T_2 . Then both diodes D_1 & D_2 will be same. Therefore V_{o1} & V_{o2} will be identical but they have opposite polarities. o/p Voltage is $V_o = 0$.

(ii) $f_c < f_{in} < (f_c + \Delta f)$. In this range of i/p frequency the induced voltage in the winding T_1 is higher than that induced in T_2 . The i/p to D_1 is higher than D_2 . Hence the +ve o/p V_{o1} of D_1 is higher than the -ve o/p V_{o2} of D_2 . There the o/p Voltage V_o is +ve. As the i/p frequency increases towards $(f_c + \Delta f)$, the +ve o/p Voltage increases.

- The o/p frequency goes outside the range of $(f_c - \Delta f)$ to $(f_c + \Delta f)$, the o/p Voltage will falls due to the reduction in tuned ckt response.

Advantages :

- It is more efficient
- It has better linearity.

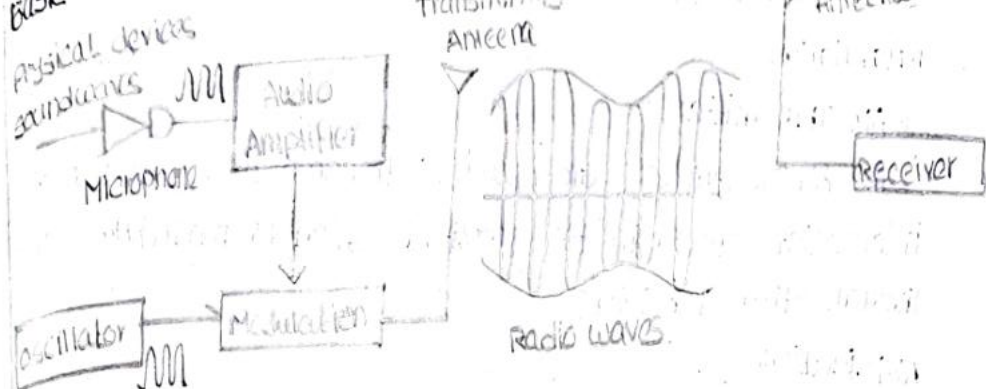
Disadvantages :

- even linearity is good.
- Amplitude limiting is not provided.

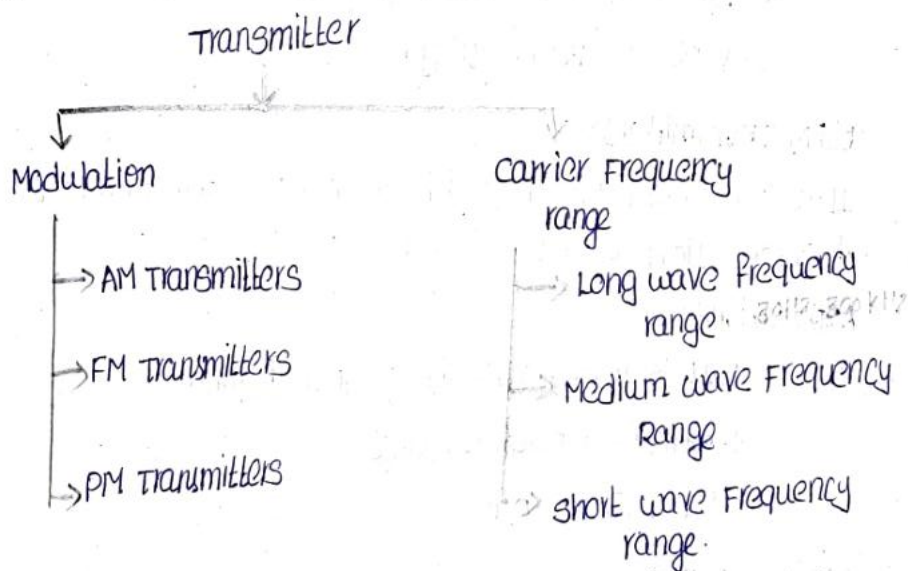
UNIT- 3

Transmitters

Basic block diagram:-



Classification of Transmitters:-



- Transmitters are the devices which are used to transmit the signal to the receiver through the channel.
- The basic block diagram of transmitter is given above.
- The microphone is used as input to transmit the sound waves from the microphone to the Audio amplifier.
- The audio amplifier amplifies the signal and gives it to Modulation.
- The oscillator which provide high frequency signals are also given as input to Modulation.
- The Modulation process takes place between the audio signal and high frequency signal and are transmitted through antenna.

• The signal is transmitted through the channel and reaches the receiver antenna.

• The receiver receives the signal through the receiver antenna.

Classification of Transmitters:-

1. Modulation:-

(a) AM Transmitters:-

The AM Transmitters are used to transmit the signal with the information stored in the amplitude which is modulated and travels through other receivers.

Applications:-

1. Radio broadcasting on short, medium, long frequency range.

2. Radio telephony (or) telegraphy on short wave

3. TV picture Broadcasting, etc.

(b) FM Transmitters:-

It is a transmitter in which frequency is transmitted as a source of information to the receiver.

Applications:-

1. It is having very high frequency range

2. ultra high frequency range

3. Radio Broadcasting

(c) PM Transmitters:-

pulse modulation transmitters uses a train of pulse as a signal to the receiver.

Applications:-

• It is used in almost all types of digital transmitters.

2. carrier frequency range:-

1. Long wave frequency range (30Hz - 300kHz):-

• In this the carrier frequency range is 30Hz - 300kHz is used to transmit long wave distance signal

Ex:- aeronautical and navy, navigation, etc

2. Medium wave Frequency range:-

- In medium wave Frequency the range is 300kHz to 3000kHz to transmit in a medium range.
- It is mainly used in AM broadcasting

3. short wave Frequency range:-

- It is used to transmit signals in a very short range between 3 to 30kHz.
- It's applications include two way radio communication, etc,

AM Transmitters:-

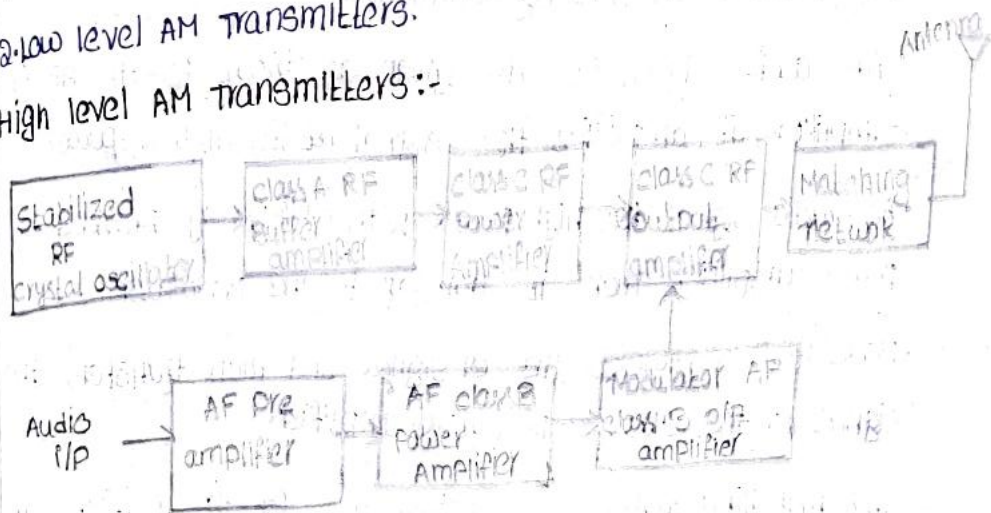
Amplitude modulation Transmitters are based on the power level and type of modulation used.

The AM Transmitters is classified into two types:-

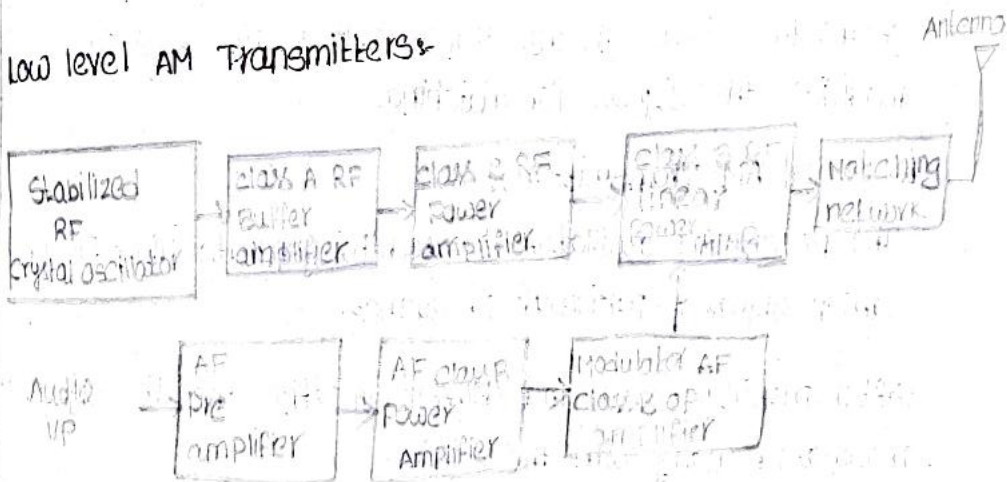
1. High level AM Transmitters

2. Low level AM Transmitters.

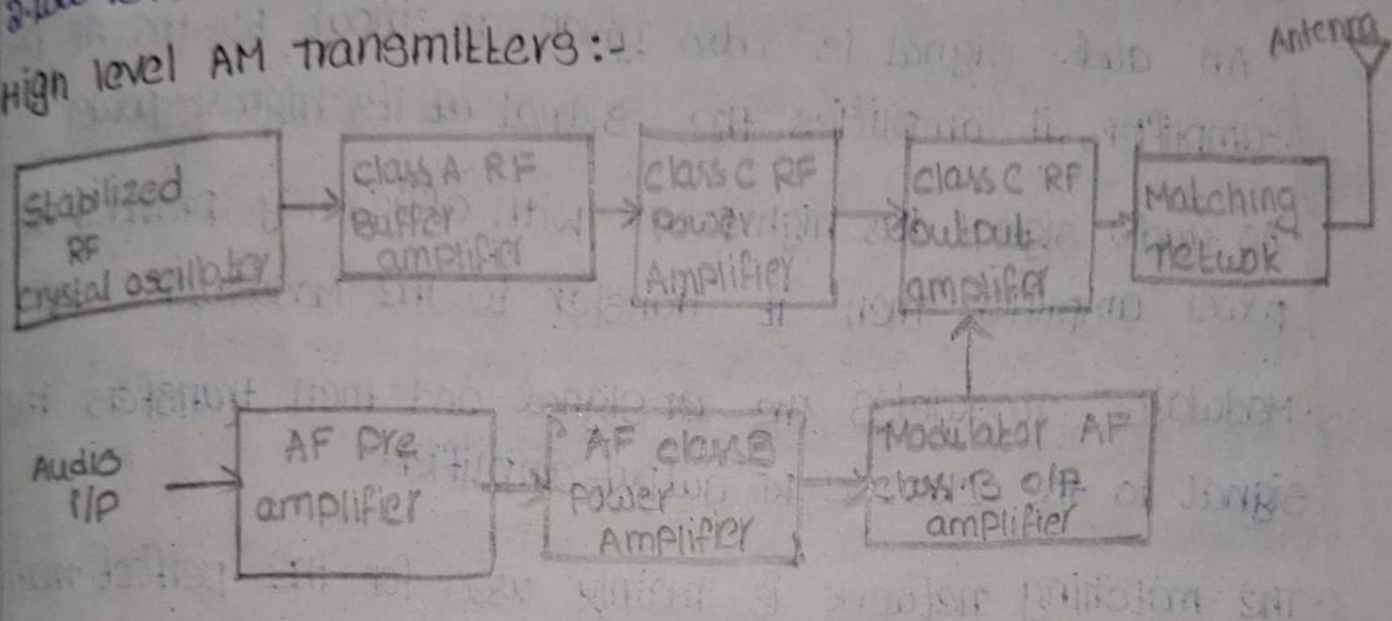
High level AM Transmitters:-



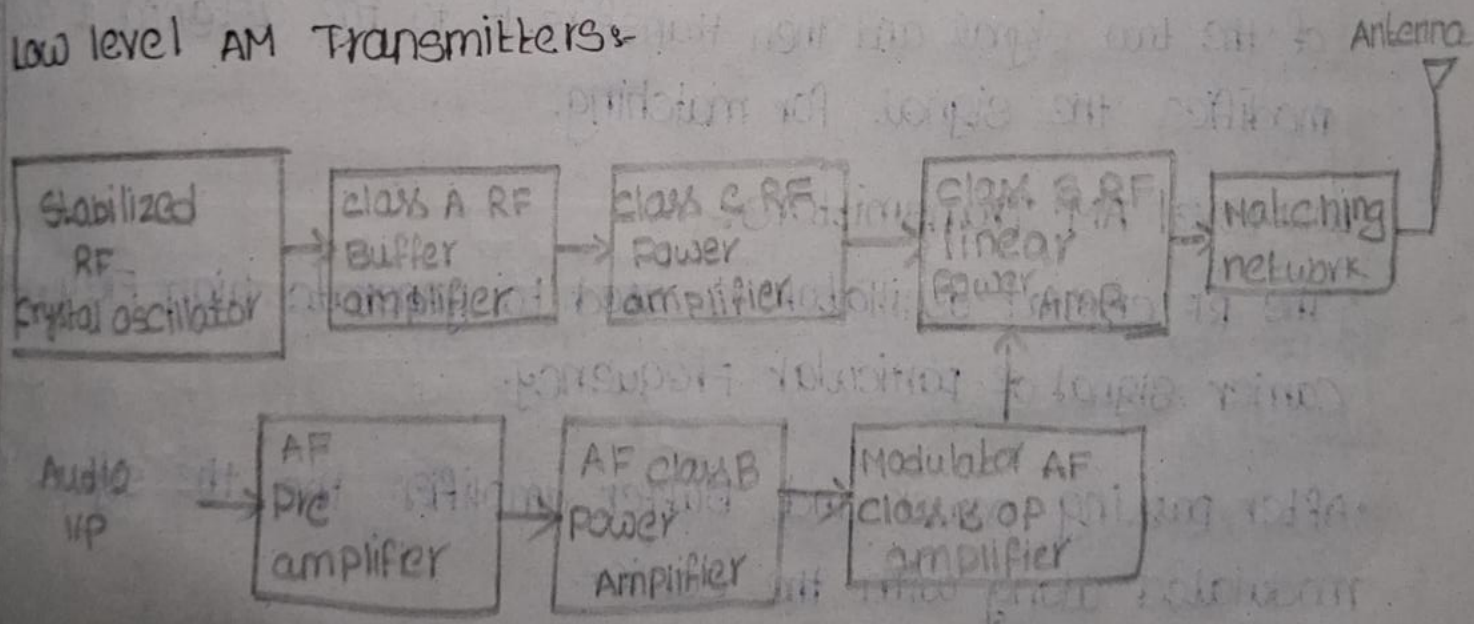
Low level AM Transmitters:-



High level AM transmitters:



Low level AM Transmitters:



High level AM Transmitters:

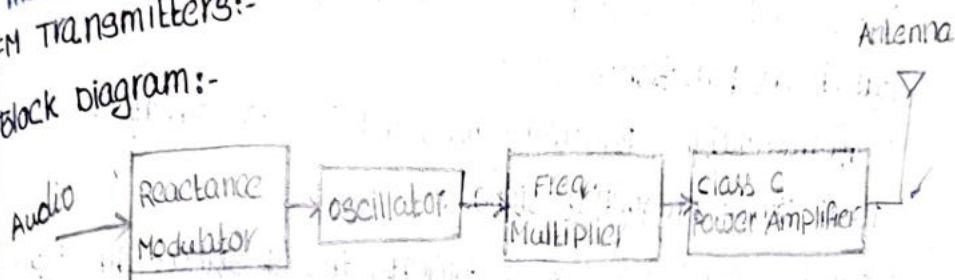
- The high level Transmitters use high level modulation and it depends on the transmitting power of the AM Transmitter.
- It is modulated in the order of the kilowatts where the low level uses only a few watts of transmitting power.
- The carrier signal generated by the stabilized crystal oscillator is passed to the Buffer amplifier
- The modulating signal is amplified at a high power level in Buffer amplifier. Buffer amplifier is mainly used to match impedance level and to transfer efficiently
- The class power amplifier provides power efficiency to the modulating signal and the class C RF output amplifier modulates the signal which can directly transmitted.
- An audio signal is also given as input to the AF pre-amplifier, it amplifies the signal at its highest power
- As audio signal has high power with efficiency provided by the power amplifier then it transfer to the modulator.
- Modulator modulates the signal and then transfers the signal to the class C RF output amplifier.
- The matching network is mainly used for the perfect matching of the two signal and then transfers it to the antenna, it also modifies the signal for matching.

Low Level AM Transmitters:

- The RF crystal oscillator is used to generate high Frequency carrier signal of particular Frequency.
- After passing power and buffer amplifier the the signal is modulated along with the carrier.
- Here the linear amplifier are used in order to avoid the waveform

- distortion in AM wave and it is useful to raise its power level.
 - As audio signal is also given as input by pre amplify and providing with its power & Frequency is passed to the modulator.
 - the modulator modifies it in its audible range and make it sufficient to the matching network to pass through antenna.
 - The overall efficiency is much lower compared to high level modulation.
- FM Transmitters:-**

Block diagram:-



- FM transmitter is a device that is used to generate a radio frequency carrier wave whose frequency is varied according to the message signal.
- The function of the reactance modulator is to modify the audio by making its range and frequency efficient.
- The oscillator is mainly used to generate carrier signal with the input audio signal.
- As the carrier and audio signal both are passed at a time, the frequency is low because of the combination of two signals.
- To obtain the required frequency for the signal to transmit, we are using frequency multiplier.
- It multiplies the frequency as $2x$, $3x$, $4x$ and so on, based on the transmission of the signal. Consider x as the signal passed.
- The final power amplifier develops in carrier power to be transmitted and often have lower power amplifier and it also helps in matching the antenna impedance to the load of the signal passed.

FM Receiver :-

A receiver is an electronic circuit or device that receives a signal, processes it and produces an output signal.

Types of Receivers :-

1. AM receiver :- Receives AM signals

2. FM receiver :- Receives FM signals

3. Digital Receiver :- Receives digital signals

Based on Function :-

1. Superheterodyne Receiver :- uses a mixer to convert the RF signal to a lower frequency signal.

2. Direct conversion Receiver :- converts the RF signal directly to the base band signal. [Tuned Radio Frequency Receiver]

Based on Applications :-

1. Radar receiver

2. Television receiver

3. Wireless communication (or) communication Receivers

4. Radar communication

4. AM Broadcast receivers

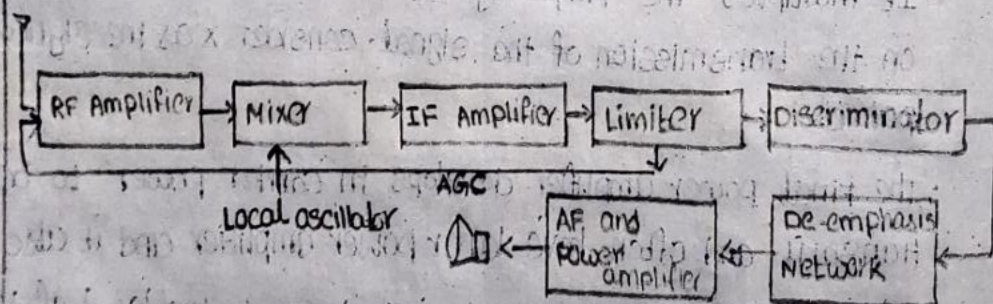
5. Navigation communication

5. FM Broadcast Receivers

FM Receiver :-

An FM receiver is a device that receives and demodulates frequency modulated radio signals extracting the original audio or information.

Block diagram :-



1. Antenna and RF tuning:- An FM receiver starts by taking incoming signal from antenna.
- An FM receiver then feeds the signal to the RF tuned section which selects the desired radio frequency from incoming electromagnetic waves.
 - The stage uses a tuned circuit, often a resonant LC circuit to select the specific FM broadcast station.
 - The tuned circuit allows the receiver to select the desired FM frequency from the radio frequency band.

2. Mixing and IF amplifier:-

- The selected radio frequency an intermediate frequency that signal which is then amplified and to remove noise and unwanted signals ensuring only the desired signal passed through.

3. Mixer:-

- It is used to mix the RF signal with the local oscillator signal within the receiver.

4. Demodulation:-

- The demodulator often a discriminator that converts the frequency variations in the IF Amplifier (which represent the original signal or audio signal) back to the audio signal, thereby recovering the original signal.

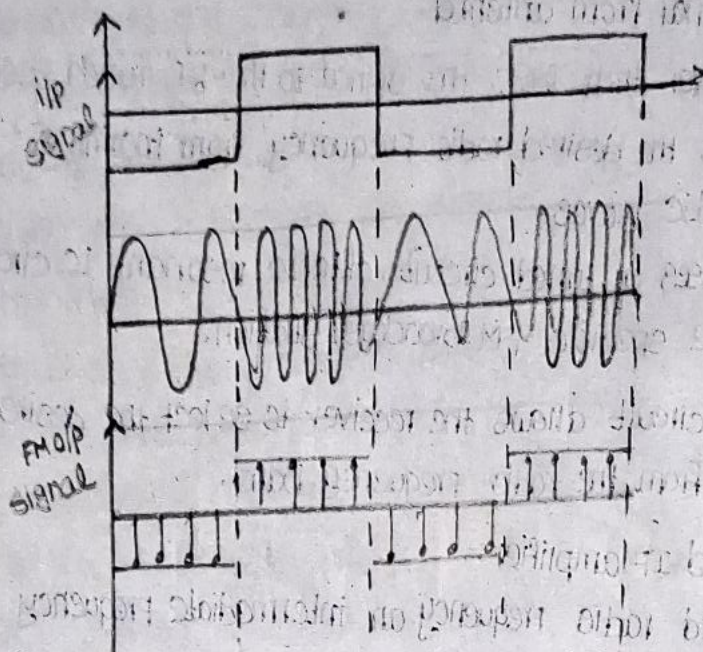
5. Amplification:-

- The recovered audio signal is then amplified to a level suitable for driving a speaker or headphone.

3. Output:-

- The amplified audio signal is sent to a speaker allowing the listener to hear the broadcast.

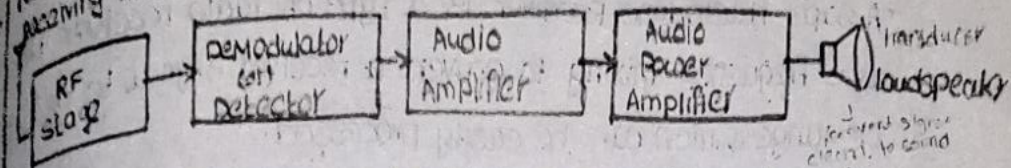
output waveform of FM receiver:-



comparison between AM and FM receiver :-

AM receiver	FM receiver
1. AM signal is applied as input to receiver	1. FM signal is applied as input to receiver
2. sound is not clear	2. sound is more clear in FM
3. Frequency range is 545-1600 KHz	3. Frequency range is 88MHz-108MHz
4. IF range: 445KHz	4. IF: 10.7MHz
5. Limiter is not used in AM	5. Limiter is used to control the frequency
6. BW occupies 10KHz space (less bandwidth than FM)	6. BW occupies up to 200KHz
7. Envelop detector is used to detect signals	7. In this, FM type of demodulator is used to detect signals
8. No need for Deemphasis	8. Deemphasis is used
9. Bandwidth = $2W$	9. Bandwidth = $2(D+1)W$

Tuned Radio Frequency:-



Tuned radio frequency (TRF) receiver is the simplest radio receiver. The above figure shows the block diagram of TRF receiver.

- The first stage is RF stage which contains two or three RF amplifiers. These are the tuned amplifiers for tuning at input and output sides.

- The receiver antenna receives signals from different sources but however with the help of tuned RF amplifiers desired signal is selected.

- The selected signal is very weak and is again amplified in RF stage.

- The demodulator detects the modulated signal and at the end of demodulator we get modulating signal (i.e., audio signal).

- The audio amplifier amplifies the audio signal.

- After this, the audio signal is amplified by a power amplifier to a desired power level and give it to the loudspeaker.

- The loudspeaker is a transducer which changes electrical signal into sound signal.

Drawbacks of TRF receiver:-

1. The TRF receiver suffers to oscillate at high frequencies

2. The selectivity of TRF is poor

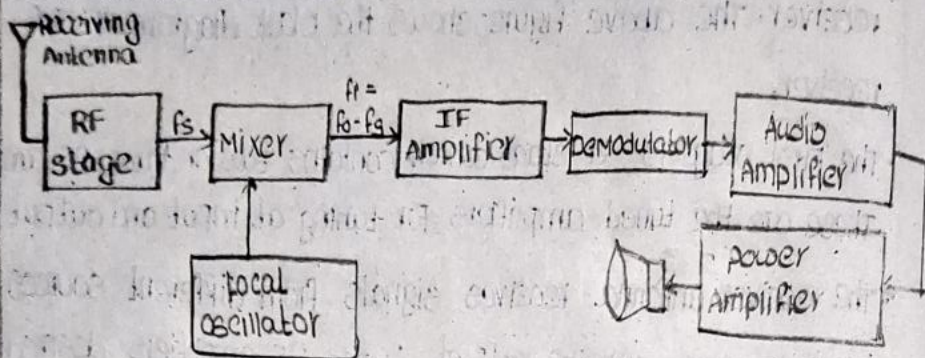
3. TRF receiver faces bandwidth variations over the tuning range.

Therefore, to overcome the drawbacks of the TRF receiver we use the superheterodyne receiver.

In most of the cases superheterodyne receiver is used, only in some cases TRF receiver is used.

2. super heterodyne receiver :-

- A super heterodyne receiver is a type of radio receiver that uses frequency mixing to convert a received signal to a fixed IF range which can be easily processed.
- It is the most widely used radio receiver.



- The RF stage contains two or more amplifiers which amplifies the incoming message signal (f_s) coming from the receiver antenna.
- Another signal i.e., local oscillator signal is combined with the tuned message signal by using a mixer.
- After passing through the mixer, the signal is given to the IF Amplifier.
- The IF amplifier provides most of the gain (i.e., sensitivity) and selectivity of the receiver.
- The IF amplifier have narrow bandwidth, so it rejects all the other frequencies except Intermediate Frequency (Low Frequency).
- Then the signal is applied to the demodulator, it extracts the original modulating signal (i.e., audio signal).
- The Audio amplifier provide particular voltage level to audio signal.
- The power is also amplified to the required power level by using a power Amplifier.
- At last, the loudspeaker converts the electrical signal to the audio. Thus the original signal is received.
- The word heterodyne stands for mixing that can be observed in the RF stage. Therefore, this receiver is called

superheterodyne receiver.

Advantages:-

1. No variations in the bandwidth
2. High sensitivity and selectivity.
3. High adjacent channel rejection.

Image Frequency:-

- We know that the superheterodyne receiver is a better receiver than a tuned radio frequency receiver.
- But in superheterodyne receiver, we face a problem of image frequency. The problem is that in the frequency conversion process carried out by a local oscillator along with mixer that it produces an undesired frequency in addition to the incoming frequency.
- We know that the local oscillator frequency is always made higher than the incoming signal frequency. i.e.,

$$f_o = f_s + f_i$$

Local oscillator frequency = desired frequency + intermediate frequency

$$f_i = f_o - f_s$$

- Now consider frequency f_{si} manages to reach mixer i.e.,

$$f_{si} = f_o + f_i$$

- The term, f_{si} is defined as image frequency and is defined as the signal frequency plus twice the intermediate frequency.

$$f_{si} = f_o + f_i$$

$$f_{si} = f_s + f_i + f_i$$

$$f_{si} = f_s + 2f_i$$

AUTOMATIC GAIN CONTROL (AGC):-

- We observe that when sound comes from a loudspeaker, it starts fading (variations in signal) with a variety of as much as 20 or 40 dB.
- If the carrier is minimum, the loudspeaker output gets inaudible and gets mixed in noise. Or if the carrier is maximum, the loudspeaker output becomes intolerably large.

- the simplest and the most effective means adopted to balance the fading is the automatic gain control (AGC) or Automatic volume control (AVC)
- The AGC system reduces the amplitude variation due to fading from a high value of 30 to 40 dB to a small value of 3 to 4 dB.

Amplitude Limiting:-

Amplitude limiting involves restricting the maximum amplitude of a signal to prevent distortion or exceeding the capacity of the receiver or channel.

• In analog system like AM and FM, amplitude limiting helps maintain signal quality by preventing overmodulation and by protecting the receiver from variations in signal strength due to noise

• This can be two types:- soft-limiting (gradually reducing the signal near the limit) or by hard limiting (completely clipping the signal exceeding a threshold)

Ex:- In FM receivers amplitude limiters are employed to reduce unwanted amplitude variations produced by noise

RF section characteristics:-

the RF section mainly consists of three characteristics:-

(i) sensitivity

(ii) selectivity

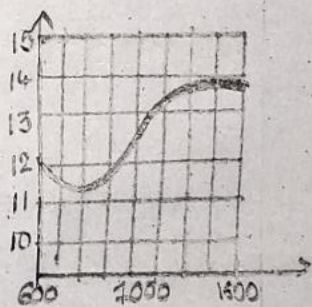
(iii) fidelity

• These are also called as receiver parameters of superheterodyne receiver

(i) sensitivity:-

• The sensitivity of a radio receiver may be defined as its ability to amplify weak signals.

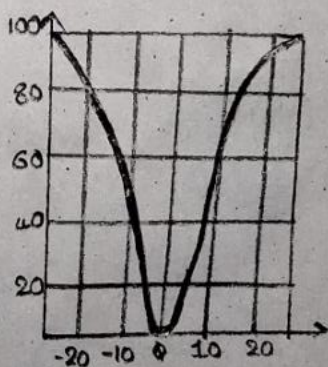
• The typical value of sensitivity are 150μ volt for small broadcast and 1μ volt or below for high broadcast receivers.



(ii) selectivity:-

• The selectivity of a receiver may be defined as the ability to reject unwanted signals.

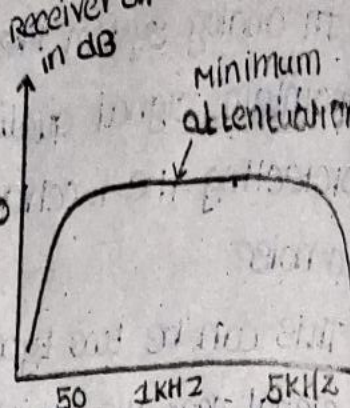
• Selectivity is the main factor which determines the adjacent channel receiver of a receiver.



(iii) Fidelity:-

The Fidelity is the ability of a receiver to reproduce all the modulating frequencies equally.

The Fidelity depends on the frequency response of the AF amplifier.



UNIT - IVIntroduction of Noise

Imp 5/2/25

• A noise is an unwanted signal which always interfere with the original or message signal.

(or)

• A noise is defined as, it is a undesirable electric signal that falls with the passband of the signal.

• A noise is anything that interfere with the communication process with the speaker and audience.

• These interfering signal is usually random fluctuations in voltage or current.

• Noise can be divide into two types

1. correlated noise

2. uncorrelated noise.

1. correlated noise:-

• correlation exists a relationship between signal and noise is occurred. therefore correlated noise exists only when the signal is presented.

2. uncorrelated noise:-

• uncorrelated exists all the time whether there is a signal or not.

• The noise can arise from different types of sources, the sources can be done,

1. Natural source (External noise)

2. Manmade source (Industrial noise)

3. Fundamental source (Internal noise)

1. Natural sources of noise:-

• The natural source of noise that will give rise to noise or electronic storms, solar flares and radiation in space.

• The noise originating known as extraterrestrial

• The extraterrestrial

1. solar noise

2. cosmic noise

• solar noise is the radiation from the

• the cosmic noise is identical to the also are large ho

5/2/25 2. Manmade source

• the manmade noise process in a current

EX:- Electrical machine

aircraft, leakage

3. Fundamental

• This type of noise communication

• Internal noise

• Internal noise

1. short noise

2. partition noise

3. Low Frequency

4. High Frequency

5. Thermal noise

1. Short noise

• The short noise

• short noise.

in all active

• The short noise

- the noise originating from the sun and outer space is known as extraterrestrial noise.
- the extraterrestrial noise can be subdivided into two groups:-

1. solar noise
2. cosmic noise

• solar noise is the electrical noise and that will take the radiation from the sun.

• the cosmic noise is generated from the stars, this is identical to the noise radiated by the sun because stars also are large hot bodies.

2. Manmade sources of noise :- (Internal noise)

• the manmade noise is generated due to the make and break process in a current carrying circuit

Ex:- Electrical motors, welding machines, switching gears, automobile, aircraft, leakage of high voltage transmission.

3. Fundamental source of noise :- (Internal noise)

• this type of noise is generated internally or within the communication system.

• Internal noise is randomly distributed entire frequency

• Internal noise is classified into 5 types

1. short noise

2. partition noise

3. Low Frequency noise (or) Flicker noise

4. High Frequency noise

5. Thermal noise

1. Short noise :-

• the short noise is produced due to the short effect

• short noise is produced in all amplifying devices rather than in all active devices

• the short noise is produced because of random variations

in electrons.

- the exact formula for shot noise can be obtain only for diodes
- the mean square short noise current for a diode is given as

$$I_n^2 = 2(I + I_0)q_e B \text{ amp}^2$$

where I is the direct current across the junction

I_0 is the saturation current

q_e is the electronic charge and

B is the effective noise bandwidth

- The shot noise is inversely to the transconductance and directly proportional to the output current.

Ex:- diodes and electron tubes.

2. partition noise:-

- It is generated in a circuit when a current has to divide between two or more paths

Ex:- diodes

3. Low Frequency noise (Flicker noise):-

- Here, this noise is generated below few KHz. In this type of noise only particular type of noise is to be appear.
- The power spectral density of this noise is increases then the frequency is decreases.

Ex:- vacuum tubes.

4. High Frequency noise (transit-time noise):-

- It is generally observed in semiconductor devices when the transit time of charge carriers crossing a junction is comparable with the time period of the signal.

5. Thermal noise:-

- A thermal noise is random and its referred as white noise or Johnson noise.

It is a random variable noise, that is generated in a resistor (or) the resistive components of complex impedance due to atoms, molecules and electrons]

Receiver Model OF Noise:-



Fig:- A noisy receiver Model

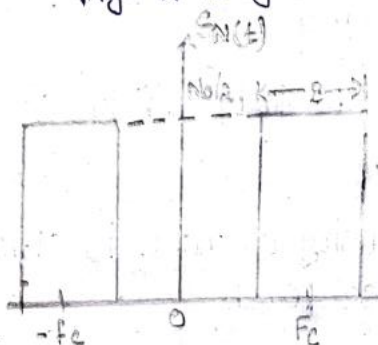
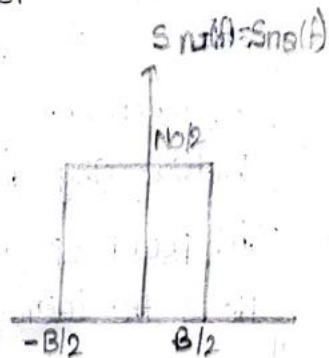


Fig:- Ideal characteristic of BPF noise



plot of $S_{NI}(f)$ & $S_{Na}(f)$

- Receiver model in noise is used for carrying out the noise with related analysis
- Receiver noise is directly related to thermal noise dynamic stress and GPS receivers.
- There are three types of noise in receiver.
 1. Thermal noise
 2. Dark current noise
 3. quantum noise
- Thermal noise:- The electronic noise generated by the thermal of the electrons inside an electrical conductor at equilibrium.
- Dark current noise:- It is the statistical variations of dark current across the sensor.
- quantum noise:- It is also called as quantum mottle. It is the random process due to fluxations in the no. of photons reacting the detector from point to point.

operation of noise receiver model:-

- In a circuit of noise receiver model, initially $s(t)$ is the incoming modulated signal and $w(t)$ is the additive white gaussian noise.
- Therefore, the receiver signal is the addition of $s(t)$ and $w(t)$.
- The combined signal $s(t)+w(t)$ is passed through the band pass filter, but it is not a actual filter, it represents combine filtering action of the tuned amplifiers used in the receiver prior to the demodulator.
- The bandwidth of BPF is just sufficient to pass $s(t)$ signal without any distortion.
- The power spectral density of noise is $N_0/2$ and here, N_0 is the average noise power.
- The bandwidth of BPF is equal to the transmission bandwidth of message signal $s(t)$.
- The mid band frequency of BPF is f_c with the power spectral density $S_N(f)$ and the noise $n(t)$ due to the passed $w(t)$ to the BPF.
- Generally, $f_c \gg B_T$ in this condition Filter noise $n(t)$ is a narrow band noise. Hence, Filter noise $n(t)$ in canonical form is given by

$$n(t) = n_I(t) \cos 2\pi f_c t - n_Q(t) \sin 2\pi f_c t$$

where $n_I(t)$ is called inphase noise

$n_Q(t)$ = Quad phase noise

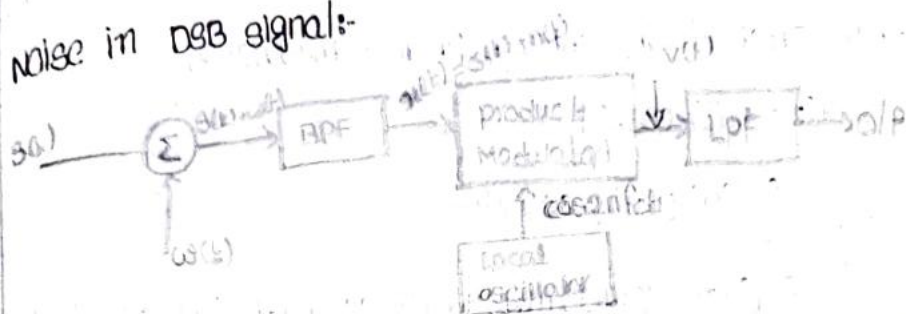
- The output of bandpass Filter is $x(t) = s(t) + n(t)$ and these signal is applied to the input of the demodulator. and $s(t)$ is depends on type of modulation and average noise power

$$\text{Avg noise power} = 2 \times \frac{N_0}{2} \times B_T$$

$$= N_0 B_T$$

$$SNR = \frac{\text{Avg power in } s(t)}{\text{Avg power in filter noise } n(t)}$$

noise in DSB signal:-



- In DSB signal we are using two sidebands along with carrier signal is used for to transmitting the signal.
- DSB signal is again subdivided into two types, one is DSBSC and DSBFC.

DSBSC Receiver noise:-

- This type of model is used for to analysis the coherent detector

operation:-

- In this circuit diagram $s(t)$ is the DSBSC modulated signal and $w(t)$ is the additive white gaussian noise.
- For combining these two signals is passed through BPF. In
- In Band pass Filter it Filtering the signal and amplifying it and it can produce filter noise or narrow band noise because the bandwidth of BPF is very small that's why it can produce narrowband type of noise.

=> [Next point]

Let us consider $s(t) = c A_c \cos(2\pi f_c t) m(t)$ — (1)

$$n(t) = n_I(t) \cos(2\pi f_c t) - n_Q(t) \sin(2\pi f_c t) \text{ — (2)}$$

Filter noise eq in canonical form to find the power is

$$P = \int_{-f_m}^{f_m} S_m(f) df$$

Then, the o/p of BPF is

$$x(t) = s(t) + n(t)$$

$$= C A_c \cos(2\pi f_c t) m(t) + n_I(t) \cos 2\pi f_c t - n_Q(t) \sin 2\pi f_c t$$

then after passing through product modulator,

$$v(t) = x(t) \cdot \cos 2\pi f_c t$$

$$= [s(t) + n(t)] \cos 2\pi f_c t$$

$$= [C A_c \cos(2\pi f_c t) m(t) + n_I(t) \cos 2\pi f_c t - n_Q(t) \sin 2\pi f_c t] \cos 2\pi f_c t$$

where C is the scaling factor

$$= C A_c \cos^2(2\pi f_c t) m(t) + n_I(t) \cos^2 2\pi f_c t - n_Q(t) \sin 2\pi f_c t \cdot \cos 2\pi f_c t$$

$$\text{Here, } \cos^2 \theta = \frac{1 + \cos 2\theta}{2}, \quad \sin \theta \cos \theta = \frac{\sin 2\theta}{2}$$

$$= C A_c \left[\frac{1}{2} + \frac{1}{2} \cos 4\pi f_c t \right] m(t) + n_I(t) \left[\frac{1}{2} + \frac{1}{2} \cos 4\pi f_c t \right] - n_Q(t) \left[\frac{\sin 4\pi f_c t}{2} \right]$$

$$= \left[\frac{1}{2} C A_c + \frac{1}{2} C A_c \cos 4\pi f_c t \right] m(t) + \frac{1}{2} n_I(t) + \frac{1}{2} \cos 4\pi f_c t [C A_c m(t)] + n_I(t) - n_Q(t) \left[\frac{\sin 4\pi f_c t}{2} \right]$$

$$= \frac{1}{2} C A_c m(t) + \frac{1}{2} n_I(t) + \frac{1}{2} \cos(4\pi f_c t) [C A_c m(t) + n_I(t)] - \frac{1}{2} n_Q(t) \sin 4\pi f_c t$$

then, after passing through LPF, we get

$$v(t) = \frac{1}{2} C A_c m(t) + \frac{1}{2} n_I(t)$$

- After that we have multiplied the signal $x(t)$ by A , local oscillator $\cos \omega_c t$ or $\cos 2\pi f_c t$ and then this product is passed through the LPF. In LPF, it is used for it allows low frequency components and it accumulates the high frequency components and then, that low frequency components produce as output.

Noise in SSB signal:-

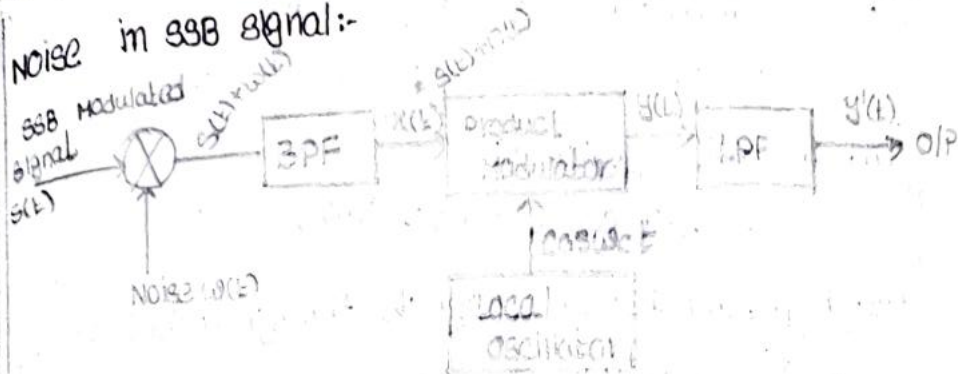


Fig:- A noisy receiver model of SSB signal.

- In SSB signal, we are going to use single side band for transmitting the signal, it contains more power compared to DSB signal.
 - A receiver model of SSB model are used for analysis of coherent detection of SSB signal.
- operation:-

- In this circuit diagram $s(t)$ is the SSB modulated signal and $w(t)$ is the additive white gaussian noise.
- For combining these two signals is passed through BPF.
- In Bandpass Filter, it filters the signal and amplify it and it can produce filter noise or narrow band noise.
- Because the bandwidth of BPF is very small that's why it can produce narrowband type of noise.
- After that we have multiplied the signal $x(t)$ with a local oscillator $\cos(\omega_c t)$ or $\cos(2\pi f_c t)$ and then this product is passed through LPF.
- In LPF, it is used for allowing low frequency components and it accumulates the high frequency components and then that low frequency components produce as output.

The ssb modulating signal is given as

$$s(t) = A_c \cos \omega_c t m(t) \pm A_c \sin \omega_c t m(t)$$

and the noise signal is

$$n(t) = n_I(t) \cos \omega_c t - n_Q(t) \sin \omega_c t$$

After passing these two signals through a BPF, we get

$$x(t) = s(t) + n(t)$$

$$= [A_c \cos \omega_c t m(t) \pm A_c \sin \omega_c t m(t)] + [n_I(t) \cos \omega_c t - n_Q(t) \sin \omega_c t]$$

Then the signal $x(t)$ is passed is multiplied by the local oscillator signal $\cos \omega_c t$ by using a product modulator, we get

$$y(t) = x(t) \cos \omega_c t$$

$$= [A_c \cos \omega_c t m(t) \pm A_c \sin \omega_c t m(t)] + [n_I(t) \cos \omega_c t - n_Q(t) \sin \omega_c t] \cos \omega_c t$$

$$= A_c \cos^2 \omega_c t m(t) \pm A_c \sin \omega_c t \cos \omega_c t m(t) + n_I(t) \cos^2 \omega_c t - n_Q(t) \sin \omega_c t \cos \omega_c t$$

$$\because \cos^2 \theta = \frac{1 + \cos 2\theta}{2} \quad ; \quad \sin A \cos B = \frac{\sin 2\theta}{2}$$

$$= \cos^2 \omega_c t [A_c m(t) + n_I(t)] + \sin \omega_c t \cos \omega_c t [A_c m(t) - n_Q(t)]$$

$$= [A_c m(t) + n_I(t)] \left[\frac{1 + \cos 2\omega_c t}{2} \right] + [A_c m(t) - n_Q(t)] \frac{\sin 2\omega_c t}{2}$$

$$= \frac{1}{2} A_c m(t) + \frac{1}{2} n_I(t) + \frac{1}{2} A_c m(t) \cos 2\omega_c t + \frac{1}{2} n_I(t) \cos 2\omega_c t + [A_c m(t) - n_Q(t)] \frac{\sin 2\omega_c t}{2}$$

Then, after passing through LPF, it allows only low frequencies and the output is given as

$$y'(t) = \frac{1}{2} A_c m(t) + \frac{1}{2} n_I(t)$$

• And we know the output signal power is given by

$$S = \frac{1}{4} A_c^2 P$$

• And the output of noise power is given by

$$(SNR_o) = \frac{P_m N_o}{4}$$

NOISE PERFORMANCE IN A.M RECEIVER using Envelope

DETECTOR:-

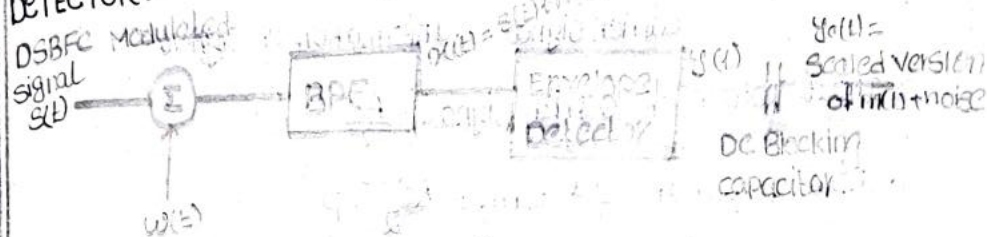


Fig:- A Noisy Model of A.M Receiver

• In an AM wave both sidebands and carrier are used for to transmit the signal.

operation:-

• In AM receiver, DSBFC modulated signal $s(t)$ and noise is applied to the combine signal.

• After combining these two signals, it is passed through the BPF and it will produce the output as $s(t) + n(t)$. Here $n(t)$ is a narrow band noise because BPF is having low bandwidth.

• Again, it is passed through the envelope detector, it is one type of AM detector, it is used to extract the signal from modulated wave

• In this output, some dc signal will appear for eliminating dc signal, we are going to use blocking capacitor.

• By using this we can eliminate the dc signal in the output of envelope detector, finally we get the output as

ected version of $m(t)$ and noise.

Input SNR or predetection SNR:-

• Consider the AM equation as

$$\begin{aligned} s(t) &= A_c (1 + k m(t)) \cos \omega_c t \\ &= A_c \cos \omega_c t + k m(t) \cos \omega_c t \end{aligned}$$

• Input SNR or predetection noise

$$\begin{aligned} s(t) &= A_c \cos \omega_c t + A_c k m(t) \cos \omega_c t \\ &\quad \downarrow \quad \quad \quad \downarrow \\ &\quad \text{carrier signal} \quad \text{Information of signal} \end{aligned}$$

• Input power of carrier signal = $\frac{A_c^2}{2}$

• Input power of 2nd term = $\frac{A_c^2 k^2}{2} P$

• Average input signal power = $\frac{A_c^2}{2} + \frac{A_c^2 k^2}{2} P$
 $= \frac{A_c^2}{2} [1 + k^2 P]$

• Average input noise power = Noise of PSD $\times B \cdot W$
 $= \frac{N_0}{2} \times 2 f_m = N_0 f_m$

$$\text{I/P SNR} = \frac{\text{Avg I/P signal power}}{\text{Avg I/P noise power}} = \frac{A_c^2 (1 + k^2 P)}{2 N_0 f_m}$$

O/P SNR or post detection SNR:-

• A noise model of BPF o/p $x(t) = n(t) + \theta(t)$

$$\therefore n(t) = n_I(t) \cos \omega_c t - n_Q(t) \sin \omega_c t$$

$$x(t) = A_c (1 + k m(t)) \cos \omega_c t + n_I(t) \cos \omega_c t - n_Q(t) \sin \omega_c t$$

$$x(t) = [A_c + A_c k m(t) + n_I(t)] \cos \omega_c t - n_Q(t) \sin \omega_c t$$

o/p of Envelope detector $y(t) = |x(t)|$

$$= \sqrt{[A_c + A_c k m(t) + n_I(t)]^2 + n_Q^2(t)}$$

$$\begin{aligned} y(t) &= \underbrace{A_c}_{\text{D.C}} + \underbrace{A_c k m(t)}_{\text{msg signal}} + \underbrace{n_I(t)}_{\text{Inphase component}} \end{aligned}$$

• After passing through DC capacitor

$$y(t) = A_c k_m(x) + n_1(t)$$

• o/p of signal power = $\frac{A_c^2}{2} k^2 P$

• o/p Noise power = Noise of PSD $\times B.W$

$$= \frac{N_0}{2} \times 2f_m = N_0 f_m$$

∴ o/p SNR (SNR)_o = $\frac{\text{o/p signal power}}{\text{o/p Noise power}} = \frac{A_c^2 k^2 P}{2 N_0 f_m}$

Noise in FM Receiver:-

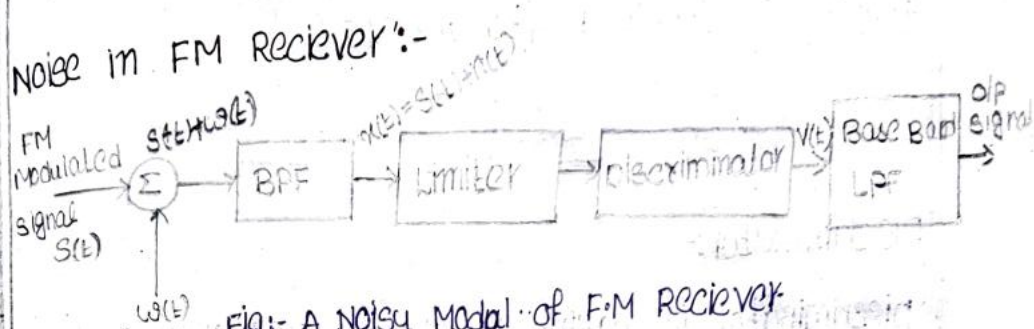


Fig:- A Noisy Model of F.M Receiver

• This type of FM receiver are used for to carry out the noise analysis of a FM system.

• In above block diagram, $s(t)$ represent the receive FM signal or FM modulated signal having a carrier frequency f_c and transmission bandwidth B_T .

• We assume that all the transmitted power lies inside the frequency band $f_c \pm B_T/2$.

• Combining these two signals i.e., FM and noise $w(t)$, they produce $s(t) + w(t)$ that is passed through the BPF.

BPF:-

• The BPF has a mid frequency (f_c) and bandwidth is B_T .

• This is essential in order to pass the FM signal without any distortion.

Noise:-

• B_T is small as compared to f_c , that's why this noise is

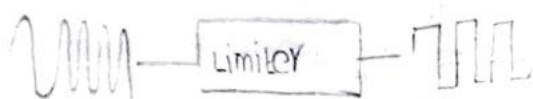
produced as narrowband noise.

$n(t)$ is represented in the form of inphase and quadrature phase component denoted by $n_I(t)$ & $n_Q(t)$.

Limiter:-

In frequency modulation, only carrier frequency is changed and the amplitude remains constant.

But due to noise is added to the FM wave, its amplitude is changes. To avoid this from happening an amplitude limiter is connected after the BPF.



Discriminator:-

Discriminator is having two types, frequency & phase discriminators.

These two discriminators to provide frequency & phase to the signal.

But FM discriminator consists of two parts, namely slope network having a purely reactive transfer function and envelope detector.

Both these parts are implemented as integral part of single unit.

LPF:-

These LPF is connected after the discriminator, the bandwidth of this filter is just large to pass highest frequency component of the message signal.

This filter will remove all out of band frequency components and present in the discriminator output.

Analysis of noise performance of a FM system:-

• The output of BPF consists of filter components $n(t)$ of the input noise $w(t)$.

• The filter noise $n(t)$ can be expressed in terms of inphase and quadrature noise components.

• The modulated FM signal is given as

$$s(t) = A_c \cos(2\pi f_c t + \phi(t))$$

• The noise signal is given as

$$n(t) = n_I(t) \cos 2\pi f_c t - n_Q(t) \sin 2\pi f_c t$$

where $n(t)$ can be in terms of envelope & phase

$$n(t) = x(t) \cos(2\pi f_c t + \psi(t))$$

where $x(t) \rightarrow$ rayleigh distribution and is given as

$$x(t) = (n_I^2(t) + n_Q^2(t))^{1/2}$$

$$\psi(t) = \tan^{-1} \left(\frac{n_Q(t)}{n_I(t)} \right)$$

• Then, the output of BPF is given as

$$x(t) = s(t) + n(t)$$

$$x(t) = A_c \cos(2\pi f_c t + \phi(t)) + x(t) \cos(2\pi f_c t + \psi(t))$$

$$\text{where } \phi(t) = 2\pi k_f \int_0^t m(t) dt$$

• And the output of discriminator is given by

$$\theta(t) - \phi(t) = \tan^{-1} \left[\frac{x(t) \sin[\psi(t) - \phi(t)]}{A_c + x(t) \cos[\psi(t) - \phi(t)]} \right]$$

$$\theta(t) = \phi(t) - \tan^{-1} \left[\frac{x(t) \sin[\psi(t) - \phi(t)]}{A_c + x(t) \cos[\psi(t) - \phi(t)]} \right]$$

part-B:-

pulse modulation

• A pulse modulation is a type of modulation in which the signal is transmitted in the form of pulses.



- In pulse modulation is used to transmit analog information
- In these continuous signals are sampled at regular interval.
- By using analog modulation, pulse modulation is classified into
 1. pulse Amplitude Modulation (PAM)
 2. pulse width Modulation
 3. pulse position Modulation

pulse Amplitude Modulation (PAM):-

- It is a form of signal modulation where the message information is encoded in the amplitude of series of signal pulses or
- It is analog pulse modulation, in which the amplitude of carrier pulse is varied according to the sampled value of the message signal.

pulse width Modulation (PWM):-

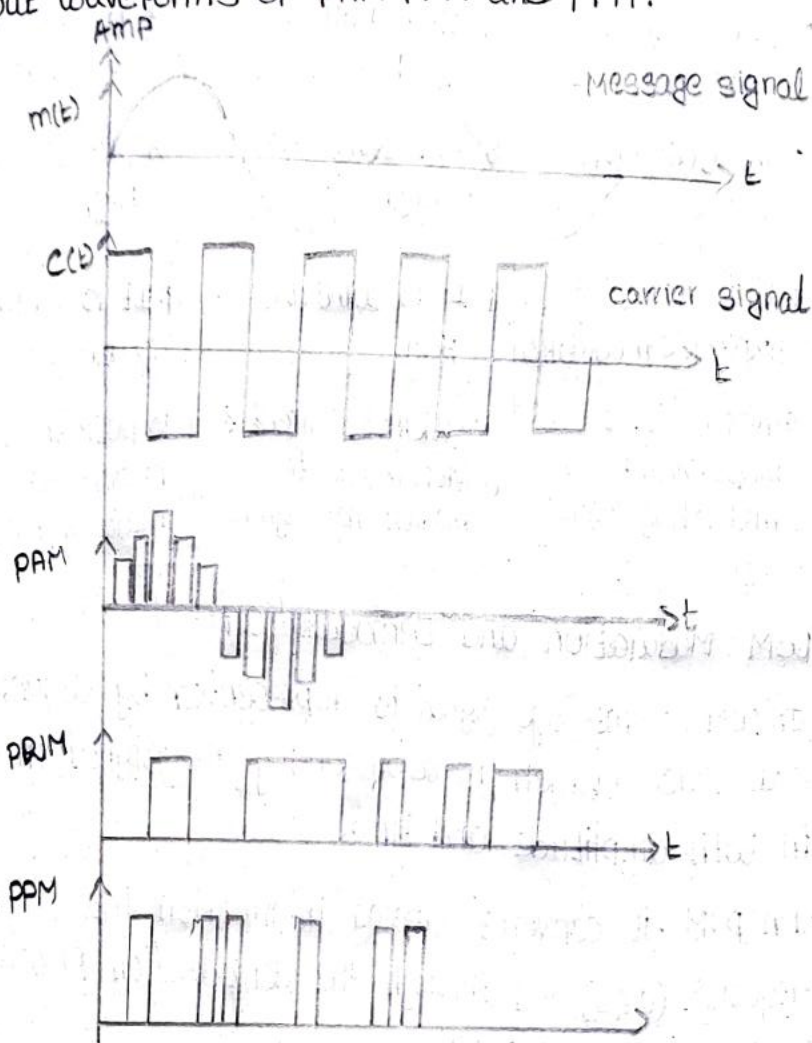
- In this the width of the carrier pulse is varied according to the modulating signal
- It is a method of reducing the average power delivered by an electrical signal.
- In this modulation, the average voltage fed to the load is controlled by tuning the switch between supply the load on and off at a fast rate.

Pulse position Modulation (PPM):-

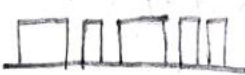
- It is the one form of signal modulation. Here, the position of the carrier pulse is varied according to the modulating signal.

• In this modulation, in which "M" message bits are encoded by transmitting a single pulse in one of the 2^m possible required time shift.

output waveforms of PAM, PWM and PPM:-



Comparison between PAM, PWM and PPM:-

PAM	PWM	PPM
1. In this amplitude of carrier pulse is varied acc. to message signal.	1. In this, the width of the carrier is varied acc. to message signal.	1. In this, the position of carrier pulse is varied acc. to the message signal.
2. 	2. 	2. 
3. B.W is at the depends on Amplitude	3. B.W depends on rise time	3. B.W depends on rise time
4. Noise is High	4. Noise is low	4. Noise is low

5. Transmission power is varied

6. Bandwidth decreases

7. Immunity is decreases

8. The SNR ratio is low

9. It is same as Amplitude modulation

10. Amplitude is proportional to modulating signal.

5. Transmission power is varied

6. Bandwidth Increase

7. Immunity is increases

8. The SNR ratio is high

9. It is same as F.M

10. Width is directly proportional to modulating signal

5. Transmission power remains constant

6. Bandwidth increases

7. Immunity is increases

8. The SNR ratio is high

9. It is same as P.M.

10. position is directly proportional to modulating signal.

PCM Modulation and Demodulation:-

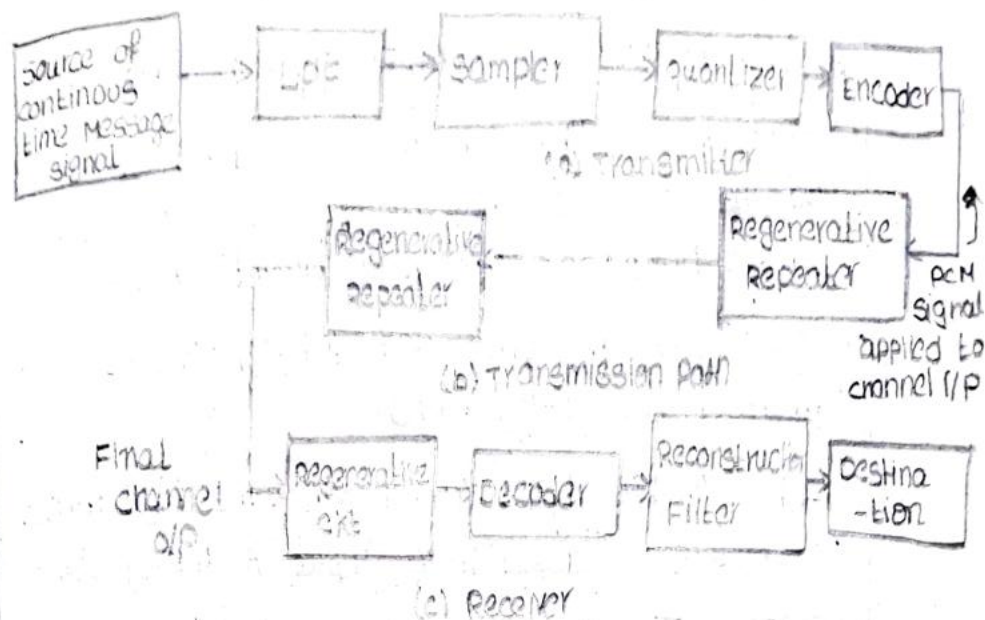
• In PCM, a message signal is represented by sequence of code pulses, which is representing the signal in discrete in both amplitude and time.

• In PCM, it converts analog information into an binary sequence (0 & 1) and through the demodulation process, we can obtain the original signal.

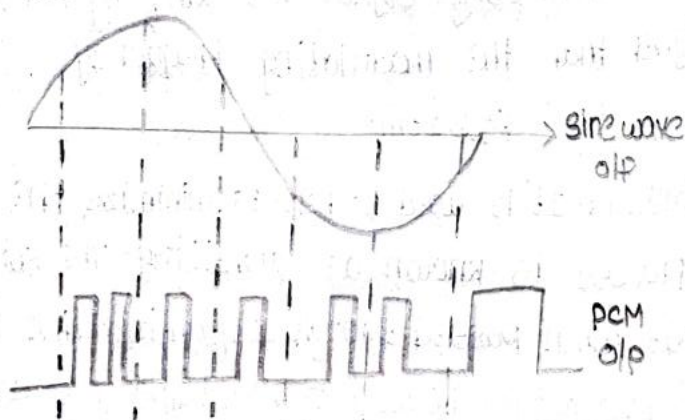
• In PCM techniques are used for a series of numbers or digits in a binary form. Hence, this process is called as Digital modulation.

• The Amplitude of the signal is sample and indicated by the binary code (0 & 1).

Block diagram of PCM:-



PCM signal waveforms:-



• PCM is mainly classified into two types. They are

1. Differential pulse code modulation (DPCM)
2. Adaptive differential pulse code modulation (ADPCM)

• In DPCM, is a signal encoding process which adds (n) functionalities based on the prediction of the sampled of the signal.

• ADPCM is a technique in which the size of the quantization steps is varied to allow for the reduction of the required bandwidth to a given SNR

• The PCM process is done through the following steps:-

1. sampling, 2. quantization and 3. coding

• In PCM process, it is possible to digitize all forms of analog data.

Ex:- music, voice, telemetry, Full motion video.

• To obtain PCM from an analog waveform at the transmitter end and to convert that message signal into a binary form, a process is known as quantization.

• At the receiver, by using demodulation the signal is converted into other forms.

• **LPF:-** A LPF helps in removing the high frequency components included in the input of analog signal.

• **sampler:-** It helps to collect the signal at any time of the message signal. The sampling frequency is higher than the modulating frequency.

$$f_s \geq 2f_m$$

17b

• **Quantizer:-** It is used to help to minimize the error through the process is known as quantizing. The sampled output when passed through a quantizer then it reduces the unnecessary beats and also helps in compressing the obtained values.

• **Encoder:-** The encoder is used for digitizing the analog signal.

• **Regenerative repeater:-** It is used to compensate for the signal loss and also to reform the signal. It also helps to increase the signal strength.

• **Decoder:-** The decoder helps to form the original signal by decoding the pulse coded waveform.

• **Reconstruction Filter:-** It helps to obtain the original signal in PCM, the given analog signal that is

to be digitized coded and sampled. To obtain the original signal, the whole process is repeated in a reverse pattern.

Advantages:-

1. This is used in long distance communication
2. Higher noise immunity
3. Efficiency is high.

Disadvantages:-

1. Bandwidth is high
2. PCM is a complex process because it involves encoding, decoding and quantization.

Quantization Noise:-

- It refers to the noise that arises when a continuous (or) discrete random variable is converted into a digital representation with fewer levels.
- Noise is mainly controlled by using code with larger values.
- This quantization is mainly used for how the signal is transmitted through the process to the output is known as quantization.

Comparison between FDM and TDM

FDM	TDM
1. FDM refers to Frequency Division Multiplexing 2. FDM is preferred as analog signal for transmitting	1. TDM refers to Time division Multiplexing 2. TDM is preferred as digital signal for transmitting

3. The signal which are to be multiplexed are added in the time domain but they occupy different slots in frequency domain

4. Synchronization is not required

5. Less efficiency

6. FDM requires more less complex circuitry

7. In FDM the frequency bands is allocated to their individual slots.

8. It is suitable for slower data rate terminals

4. The signal are to be multiplexed can occupy entire bandwidth in the time domain

4. Synchronization is required.

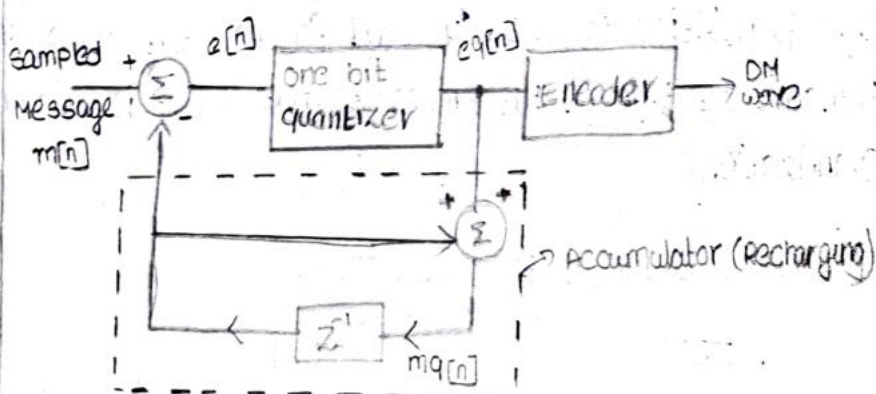
5. More efficiency

6. TDM is having more complex in the transmitter & receiver.

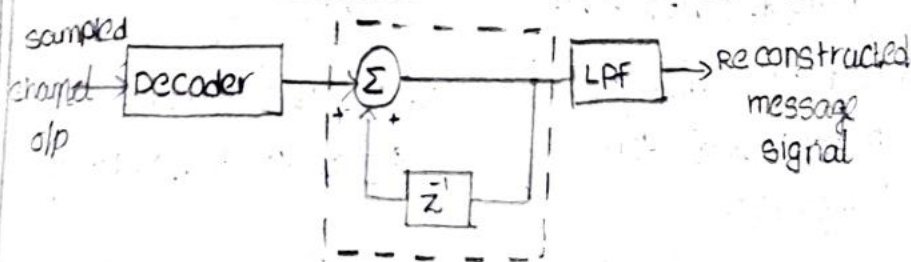
7. In TDM, the time slots are allocated to their individual bands.

8. It is suitable for higher data rate terminals based on the no. of channels.

Delta Modulation:-



(a) Transmitter



(b) Receiver

UNIT-V

DIGITAL MODULATION SCHEMES

Introduction:-

- A Digital Modulation is the process of encoding digital information signal into the Amplitude, phase or Frequency of the transmitted signal.
- Digital Modulation is one of the modulation method in which discrete signal are used to modulate carrier wave and remove noise from the waves is known as digital modulation.
- Digital Modulation is always provides numbers or discrete values (0;1)
- Digital Modulation provides increased data capacity and it provides more information security and high speed accessibility and high quality transmission.
- Digital Modulation is mainly used to reduce the communication noise.
- Digital modulation is used in cable TV and broadcast system.
- In analog Modulation, the signal can be represent any value within a range but in digital modulation it can only represent discrete values of 1 and 0.
- In analog modulation it is difficult to separate noise from the signal but in digital modulation it is easy to separate signal from the noise.
- In analog modulation, it produces signal in this carrier signal is changing the information but in digital modulation, it produce a signal whose value is changes at a specific interval of time.

Digital IP
data Binary 224/1111

Digital modulation



Types of digital modulation:-

Digital modulation is classified into three types.

1. Amplitude shift keying (ASK) (or)

Binary amplitude shift keying (BASK)

2. Frequency shift keying (FSK) or BFSK

3. phase shift keying (PSK) or

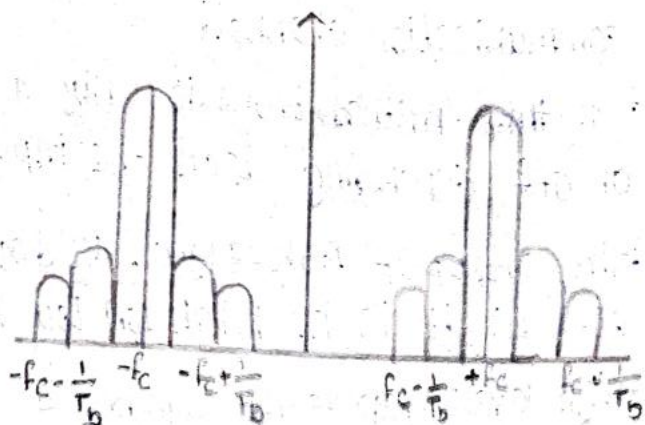
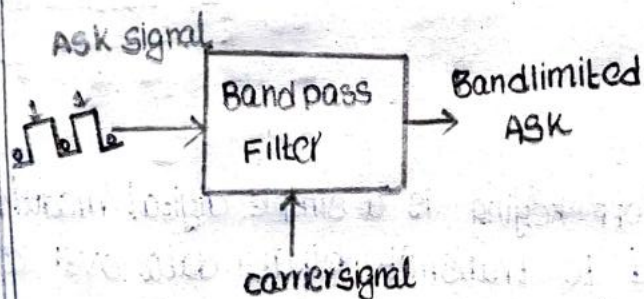
=> PSK is again classified into two types:-

(a) BPSK (or) 2 phase shift keying 0° (or) 180°

(b) quadrature phase shift keying (QPSK) $0^\circ, 90^\circ, 180^\circ, 270^\circ$

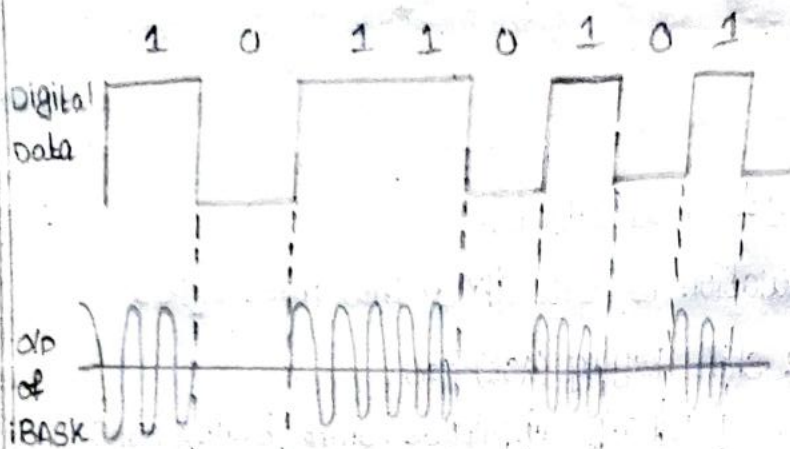
1. Amplitude shift keying (or) BASK (or) ON-OFF keying:-

Generation of BASK:-

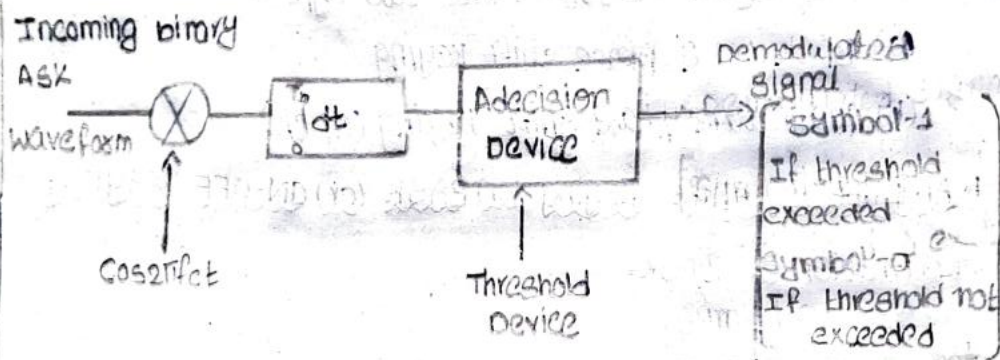


PSD of ASK signal

Graphical Representation of ASK:-



Demodulation of ASK:-



• ASK (or) BASK (or) ON-OFF keying is a simple digital modulation technique. It is used to transmit binary data over a communication channel.

• In this method, there is only 1 unit energy carrier ON or OFF depending upon the input binary sequence.

• The output of ASK signal depends on the input data either it is in high (or) low level.

• High levels represent symbol '1' and low level represent symbol '0'.

• The ASK waveform can be represented as

$$S_1(t) = \sqrt{2P_s} \cos(2\pi f_c t)$$

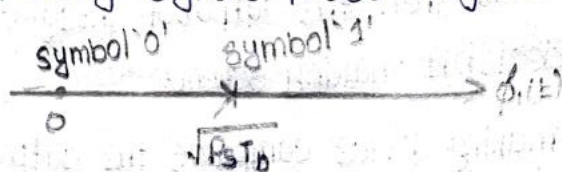
where P_s is power spectral density
 f_c is carrier frequency

• If to transmit symbol '0' then the signal $s(t)=0$ that means no signal is transmitted and no signal $s(t)$ contains some complete cycle of carrier frequency. Hence, the ASK waveform represent like the ON-OFF signal.

• If to transmit symbol '1' the signal

$$s(t) = \sqrt{\frac{2}{T_b}} \sqrt{P_s T_b} \cos(2\pi f_c t)$$

• This means there is only one carrier function $\phi_c(t)$ then the corresponding signals phase diagram



signal space diagram

• Here one end will be at zero and other end is to be $\sqrt{P_s T_b}$ then the signal between distance between two signal point is

$$d = \sqrt{P_s T_b} = \sqrt{E_b}$$

Generation:-

- For the generation of BASK, Amplitude shift keying (ASK) signal is applied as input to the BPF.
- ASK is the digital modulation technique in which discrete signal are used to modulate and also to remove the noise from the signal.
- Here, the ASK signal and the carrier signal acts as input to the Band pass Filter.
- The function BPF is to allow certain frequency range to pass through it and rejects other frequencies.
- After passing through the BPF, the signal gets modulated as Bandlimited ASK

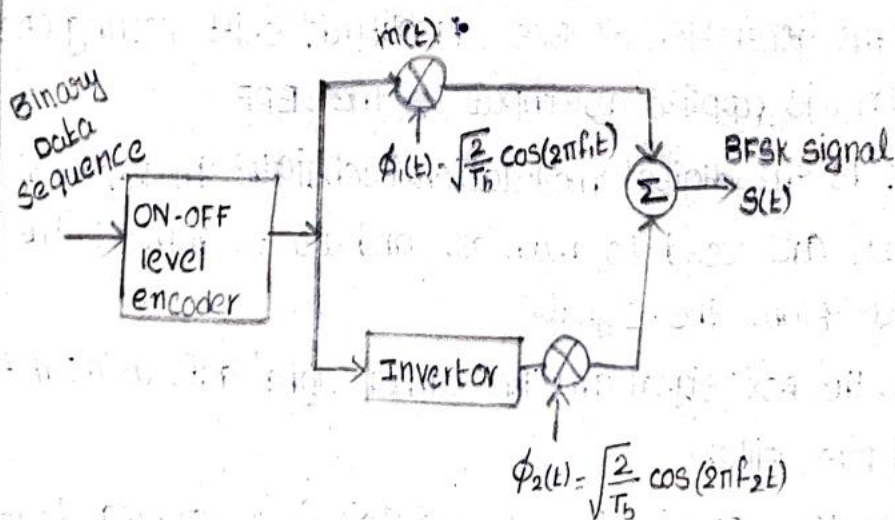
Demodulation:-

- The demodulation of BASK is the process of extracting the original binary data from the modulated signal.
- The input of the demodulator is the modulated ASK signal along with the carrier signal $\cos 2\pi f_c t$.
- The input is then fed to the product modulator. In this, the two signals are combined and multiplied and then it is given to the integrator.
- The integrator operates on the output of multiplier for successive bit intervals. Then the output of integrator goes to the input of decision making scheme.
- Now the decision making device compares the output of integrator with threshold device. It makes decision -1 when the threshold is exceeded and symbol -0 when the threshold is not exceeded. Hence, we get the output as demodulated signal.

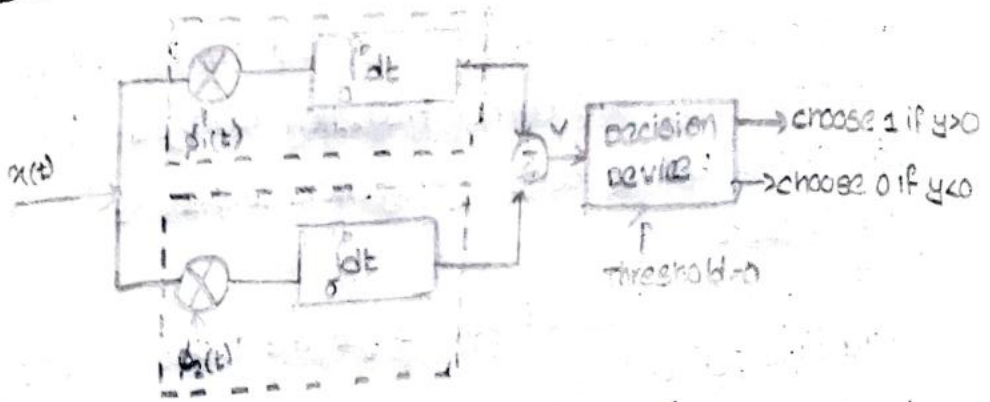
24/2:-

2. Binary Frequency Shift Keying:-

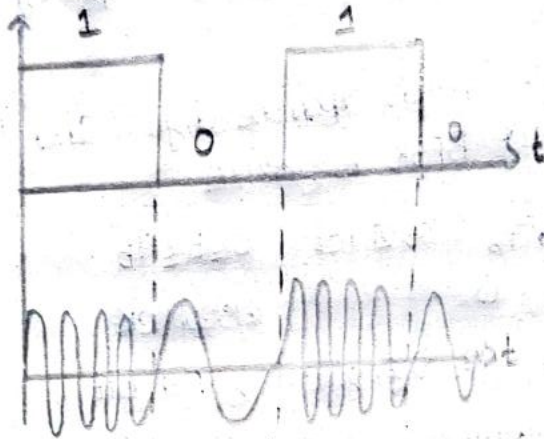
Generation of BFSK:-



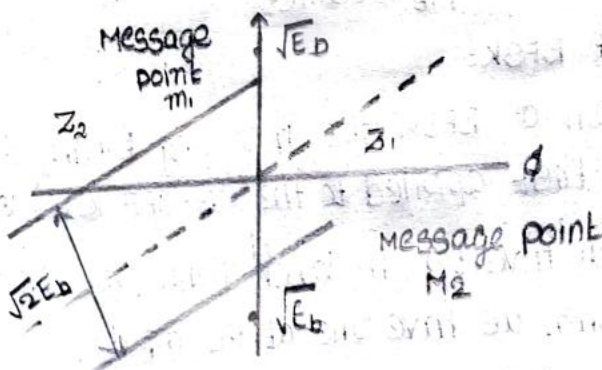
Demodulation of BFSK signal:-



Graphical Representation of BFSK signal:-



signal space diagram of BFSK signal:-



- Binary Frequency shift keying is one type of digital modulation technique. Here carrier signal contains two predefined frequencies based on the input data sequence 1 (or) 0.
- BFSK is used for to transmit the binary data through the channel.
- In BFSK symbol 1 and 0 are related to each other by transmitting symbol with different frequency by a

Fixed amount.

- In BFSK the information is stored in frequency.
- In BFSK only the carrier frequency changed depends upon the input binary sequence 1, 0. Here amplitude is kept to be constant.
- If the input data is 1 then we get the input frequency more and time period less.
- If the input data is 0, then the carrier frequency is less and time period is more.
- The output of BFSK depends upon the digital and input data, then the equation of BFSK is given as

$$s(t) = \begin{cases} \sqrt{\frac{2E_b}{T_b}} \cos 2\pi f_c t & 0 \leq t \leq T_b \\ 0 & \text{elsewhere} \end{cases}$$

where E_b is the signal energy per bit

- Similarly the orthonormal basic function is given by

$$\phi(t) = \sqrt{\frac{2}{T_b}} \cos 2\pi f_c t$$

Generation of BFSK:

- In generation of BFSK, the incoming binary data sequence is first applied to the ON-OFF level encoder.
- By using an inverter in lower channel, we in effect make sure that when we have one at the input.
- In this case the oscillator with frequency in the upper channel is switched on while the oscillator with frequency f_2 in the lower channel is switched off. With the result f_1 is transmitted.
- Similarly when we sample zero at the oscillator in the upper channel is switched off and oscillator in lower channel is switched on. With the result frequency f_2 is transmitted.

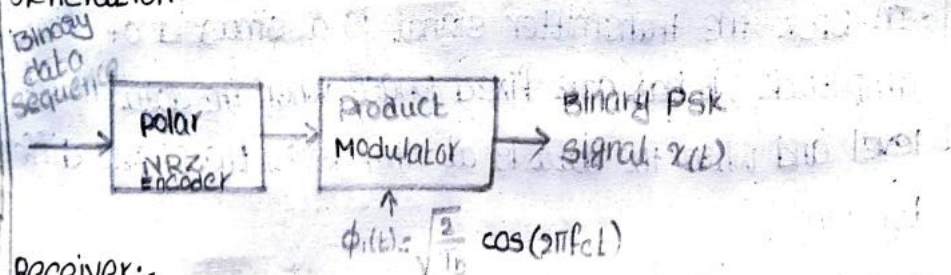
• In the transmitter, assume that two oscillators are synchronised so that the requirement of two orthonormal basic function $\phi_1(t)$ and $\phi_2(t)$, then the frequency of modulated wave is shifted with a continuous phase in accordance with the input binary data.

coherent detection of BFSK:-

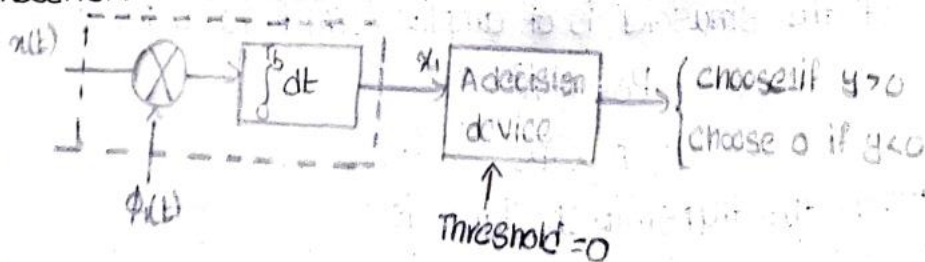
- To detect the original binary sequence we may use the receiver, it consist of two correlator with a common input which are supplied with a coherent reference signals $\phi_1(t)$ and $\phi_2(t)$
- The correlator output are subtracted from each other and the resulting difference is γ
- If $\gamma > 0$, the receiver decides in favor of 1. on the other hand if $\gamma < 0$, it decides in favor of 0
- The receiver makes a random guess in favour of 1 or 0 if $\gamma = 0$.

3. Binary phase shift keying [BPSK]:-

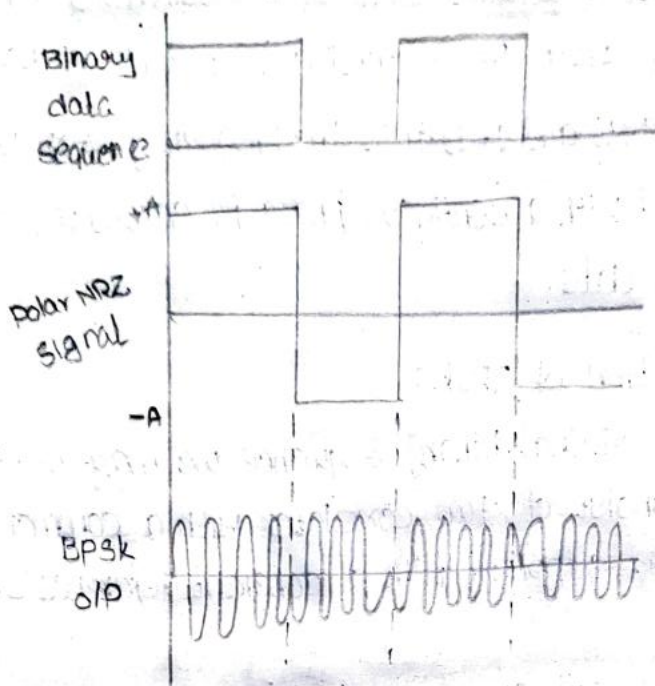
Generation:-



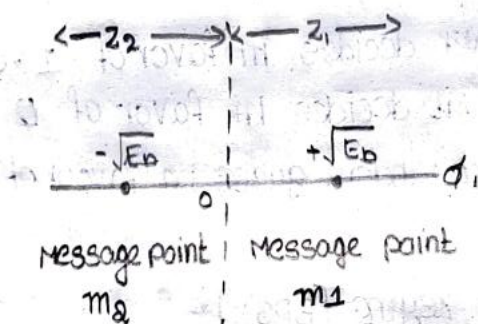
Receiver:-



Graphical Representation of BPSK:-



Signal space diagram of BPSK:-



• In BPSK the transmitter signal is a sinusoid of fixed amplitude, it has one fixed phase when the data is at one level and when the data is at other level, the phase differs by 180° .

• If the sinusoid is of amplitude (A) it has a power

$$P_s = \frac{1}{2} A^2$$

$$\Rightarrow A = \sqrt{2P_s}$$

• Thus the transmitter signal is

$$s(t) = \sqrt{2P_s} \cos(2\pi f_c t) \quad \text{when Symbol 1 (0}^\circ\text{)}$$

$$s(t) = \sqrt{2P_s} \cos(2\pi f_c t + \pi) \quad \text{when Symbol 0 (180}^\circ\text{)}$$

- In BPSK, the data $b(t)$ is of binary digits with voltage level $+1V$ and $-1V$. When $b(t) = 1$, then it is logic level 1. When $b(t) = -1$, then the logic level will be '0'.
- Then the V_{BPSK} can be rewrite as

$$V_{BPSK} = b(t) \sqrt{2P_s} \cos \omega_c t$$

- Similarly, the receiver signal of BPSK

$$V_{BPSK} = b(t) \sqrt{2P_s} \cos(\omega_c t + \theta).$$

Generation of BPSK:-

- BPSK is a digital modulation technique, in which the phase is changed according to the carrier signal.
- The input of the BPSK is the binary data sequence, which is applied to polar Nonzero Encoder.
- The polar NZR encoder works as when the signal is 0 and 1, it produces output both when 0 and 1.
- Usually when the input is 0, there is no signal but when the 0 is applied to polar NZR, it produces the signal.
- Then the signal is given input as to product modulator.
- In the product modulator the given signal and given orthonormal basic function are multiplied. Hence, the output is the binary PSK signal.

Receiver (or) Demodulation:-

- To detect the original signal i.e., binary data sequence, we are using the receiver part.
- It consists of one correlator in which a product modulator and integrator. When the input is applied to the correlator, the input signal along with the $\phi(t)$ signal are modified and then given to the decision maker.

- In the decision device, the given signal assume x_i along with the threshold signal are applied:

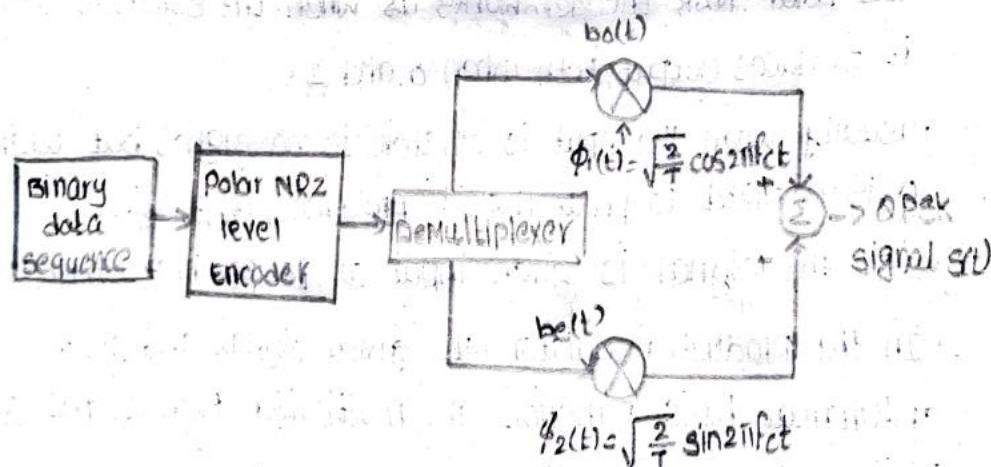
- The decision device gives the output '1' when the output $y > 0$. otherwise when the output is '0', it means that $y < 0$.

- From the graphical representation, it is clear that the output has a sudden phase shift when the polar NRZ changes from 1 to 0 (or) From 0 to 1. i.e., called as Binary phase shift keying.

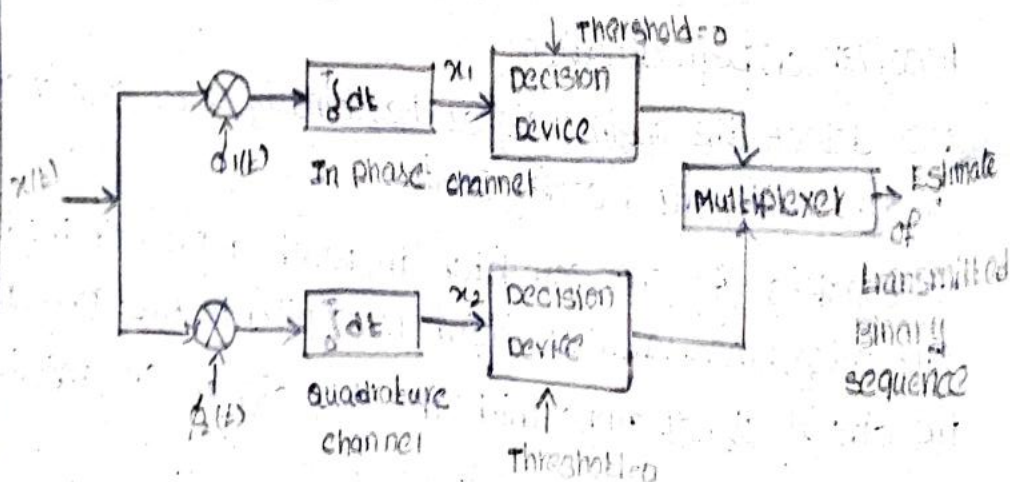
Quadrature phase shift keying [QPSK] :-

Quadri (or) Quarternary phase keying.

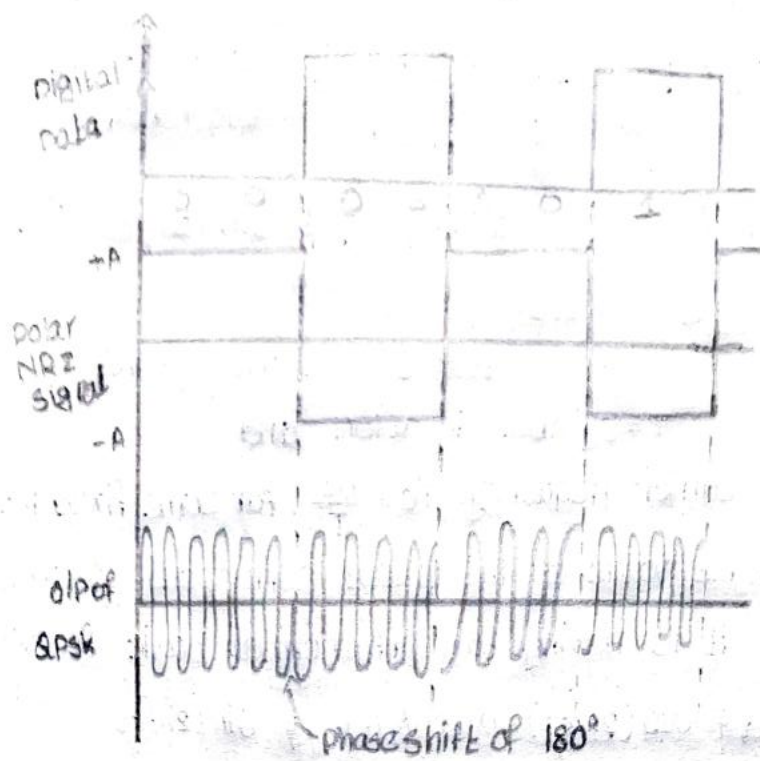
Generation of QPSK:-



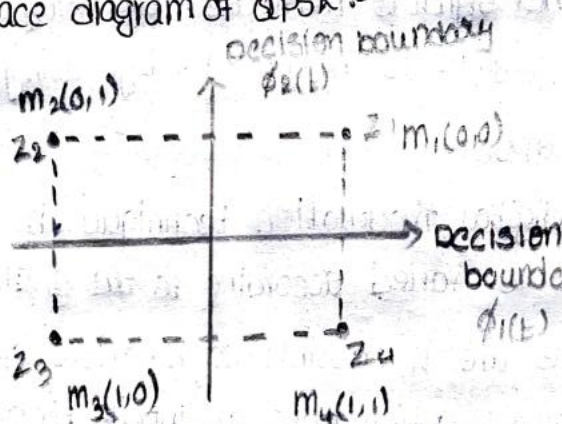
Demodulation of QPSK:-



Waveform of QPSK:-



Signal space diagram of QPSK:-



- It is a bandwidth conserving (efficient) modulation scheme known as coherent quadrature phase shift keying.
- In QPSK as with BPSK, the information is all stored in phase.
- QPSK is the one type of digital modulation technique. Here, the phase of the carrier signal is varied depending upon the input binary data.
- The phase of the carrier signal, it takes one or four values like $\frac{\pi}{4}$, $\frac{3\pi}{4}$, $\frac{5\pi}{4}$ and $\frac{7\pi}{4}$ with the phase shift of 0° , 90° , 180° and 270° .

The equation of QPSK is given by

$$s(t) = \begin{cases} \sqrt{\frac{2E}{T}} \cos(2\pi f_c t + (2i-1)\frac{\pi}{4}) & 0 \leq t \leq T_b \\ 0 & \text{elsewhere} \end{cases}$$

$$s(t) = \sqrt{\frac{2E}{T}} \cos(2\pi f_c t + (2i-1)\frac{\pi}{4}) - \sqrt{\frac{2E}{T}} \sin(2\pi f_c t + (2i-1)\frac{\pi}{4})$$

where $i = 1, 2, 3, \dots$

E = Transmitter signal energy and

T = Symbol duration and

The carrier frequency $f_c = \frac{n_c}{T}$ for some fixed integer n_c .

- Each possible value of the phase correspond to unit digit
- In QPSK it is having two orthonormal basic functions i.e.,

$$\phi_1(t) = \sqrt{\frac{2}{T}} \cos(2\pi f_c t) \text{ and } \phi_2(t) = \sqrt{\frac{2}{T}} \sin(2\pi f_c t)$$

- In BPSK, carrier signal is modulated by using 1 bit, but in QPSK carrier signal is modulated by using 2 bits.

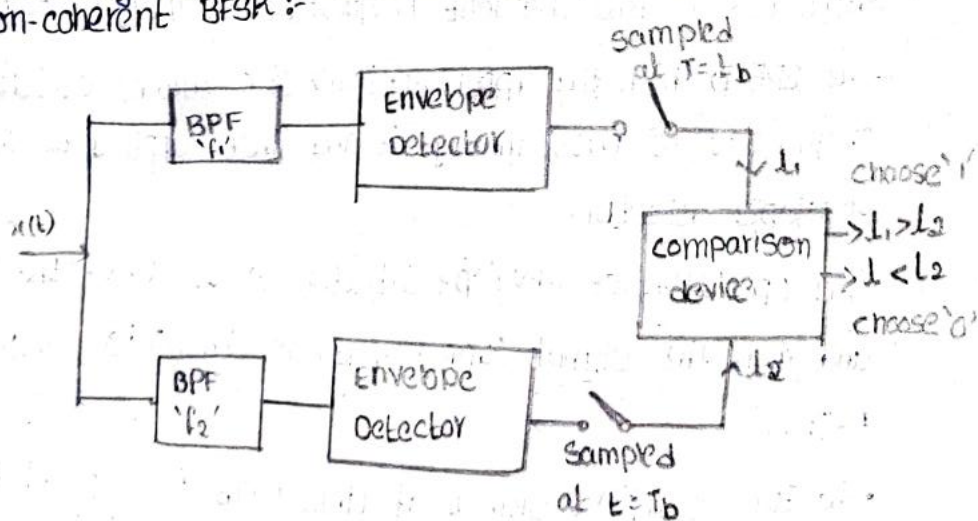
Generation of QPSK:-

- QPSK is a digital modulation technique, in which the carrier signal is varied according to the given input signal.
- As QPSK, there are 4 possible combinations of binary digits i.e., 00, 01, 10, 11. which acts as input binary data sequence.
- Then the polar NRZ Encoder encodes the 0's and 1's values to all 1's that is the non return zero function.
- The output of the encoder is then applied to the demultiplexer.
- Demultiplexer separates the given signal in even and odd binary data.
- Then for both binary and odd binary data, two different orthogonal basic functions are applied to get the two different outputs from the data.
- At last, these two signals or data is combined and acts as a QPSK signal $s(t)$.

Demodulation of QPSK:-

- The demodulation is used to detect the original binary 2-digit input sequence.
- Here the input signal $x(t)$ along with the two different orthonormal functions $\phi_1(t)$ and $\phi_2(t)$ are combined and then applied to the integrator.
- The upper correlator is in phase to the channel whereas the lower correlator is in quadrature to the channel.
- Consider the upper signal is x_1 and lower signal as x_2 .
- After both the correlators, the decision device along with the threshold = 0 is applied to the device, it compares the decision of applied signal and that of threshold and gives the output to the multiplexer.
- The operation of multiplexer is to combine the both the signals and the output is binary data sequence.
- From the waveforms we can observe that the o/p of QPSK will have phase shift of 180° . That's why it is called as Quadrature Phase Shift Keying.
- The signal space shows the four possible combinations of binary data along with their decisions and four regions.

⇒ Non-coherent BFSK:-



(OR)

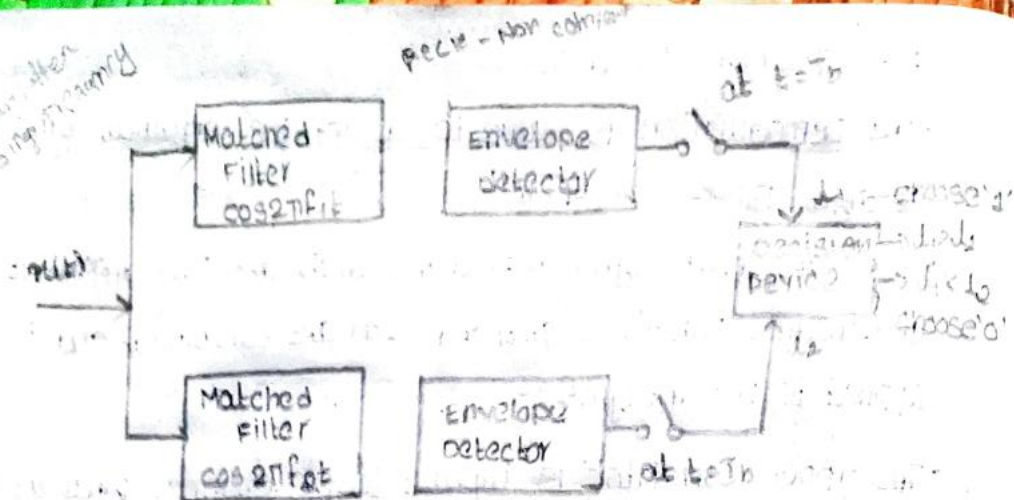


Fig:- Non coherent receiver (or)
detection of BPSK signal

operation:-

- coherent BPSK is mainly used in the transmitter side with single frequency and single wavelength.
- whereas in non coherent is used in the transmitter receiver side with multiple frequencies and multiple wavelength.
- The standard equation for BPSK is given as

$$x(t) = \sqrt{2P_s} \cos(2\pi f_c t) \quad 0 \leq t \leq T_b$$

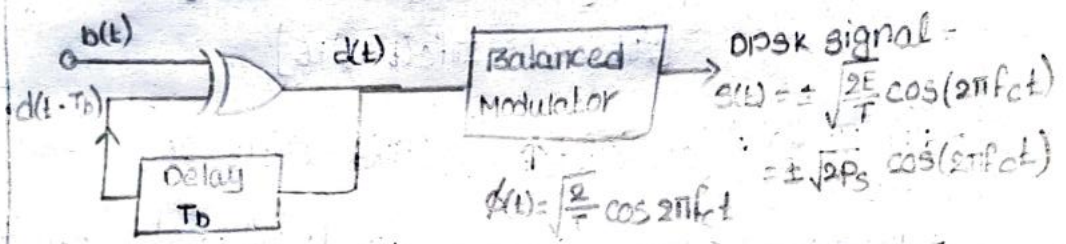
$$= \sqrt{\frac{2E}{T}} \cos(2\pi f_c t)$$

- The input $x(t)$ is applied to the Bandpass Filter.
- As it is non-coherent, single frequency is applied and then separated as two different frequencies for applying to the BPF.
- The BPF both on the upper and lower side allows certain frequencies to pass through it and then applied to the envelope detector.
- The operation of envelope detector is to detect the signal and then the signal (upper & lower signal) is sampled at $t=T_b$.
- The two sampled signal at time $t=T_b$ is applied to the comparison (or) decision device.

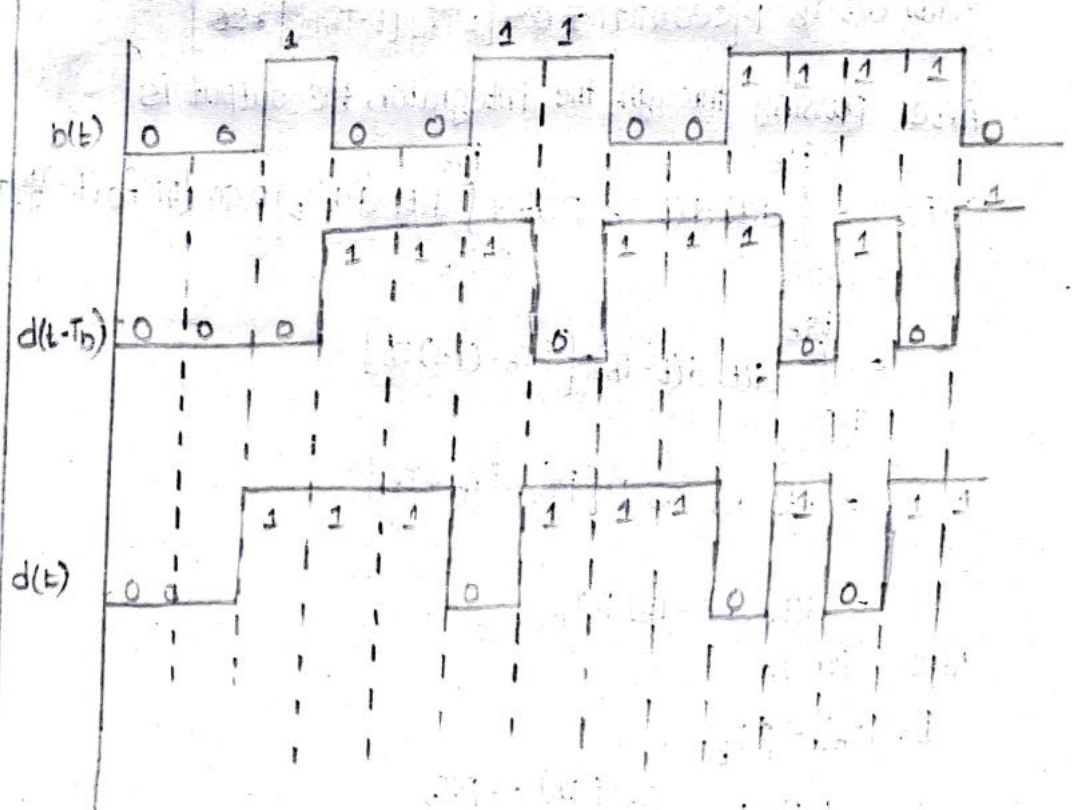
- The upper signal is denoted by d_1 and lower signal is denoted by d_2
- The comparison device compares the two incoming signal
- If the upper signal is greater than lower signal i.e., $d_1 > d_2$ it makes the decision as '1'
- otherwise if the upper signal is less than lower signal i.e., $d_1 < d_2$, it makes the decision as '0'.
- Hence, the final ~~is~~ output depends on the decision device and on the sampled signals.

Differential phase shift keying (DPSK):-

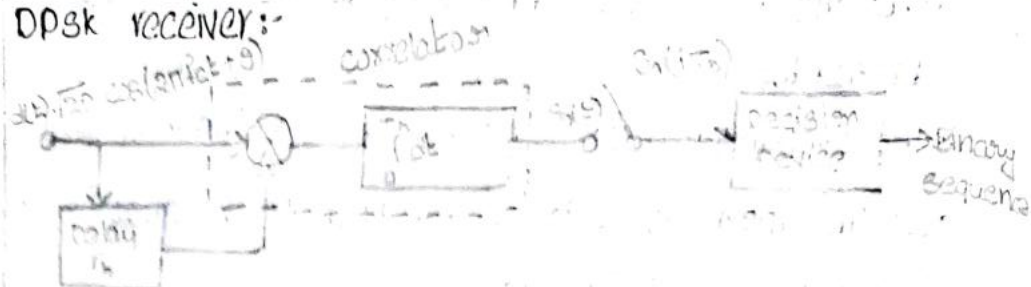
DPSK Transmitter:-



DPSK waveform:-



DPSK receiver:-



$$d(t-T_b) \sqrt{2P} \cos(2\pi f_c(t-T_b) + \theta)$$

$$S(t) = d(t) \sqrt{2P} \cos(2\pi f_c t + \theta) \cdot d(t-T_b) \sqrt{2P} \cos(2\pi f_c(t-T_b) + \theta)$$

$$= d(t) d(t-T_b) 2P \cos(2\pi f_c t + \theta) \cos(2\pi f_c(t-T_b) + \theta)$$

$$\because \cos A \cos B = \frac{\cos(A+B) + \cos(A-B)}{2}$$

$$= d(t) d(t-T_b) P \cos[(2\pi f_c t + \theta) + (2\pi f_c(t-T_b) + \theta)] + \cos[(2\pi f_c t + \theta) - (2\pi f_c(t-T_b) + \theta)]$$

$$= d(t) d(t-T_b) P [\cos 2\pi f_c t T_b + \cos(4\pi f_c t - 2\pi f_c T_b + 2\theta)]$$

$$= d(t) d(t-T_b) P [\cos 2\pi f_c T_b + \cos[4\pi f_c [t - \frac{T_b}{2}] + 2\theta]] \quad \because f_c = \frac{n}{T_b}$$

$$= d(t) d(t-T_b) P [\cos 2\pi n + \cos[4\pi f_c [t - T_b/2] + 2\theta]]$$

After passing through the integrator, the output is

$$S_o(iT_b) = \int_{(i-1)T_b}^{iT_b} d(t) d(t-T_b) P dt + \int_{(i-1)T_b}^{iT_b} d(t) d(t-T_b) P \cos[4\pi f_c (t - \frac{T_b}{2}) + 2\theta] dt$$

$$= \int_{(i-1)T_b}^{iT_b} d(t) d(t-T_b) P [iT_b - (i-1)T_b] dt$$

$$= d(t) d(t-T_b) P [iT_b - (i-1)T_b + T_b]$$

$$= d(t) d(t-T_b) P T_b$$

After in the

$$S_o(iT_b) = P T_b$$

$$d(t) = 1 \quad S_o(iT_b) = -P T_b$$

$$d(t) = 0 \quad S_o(iT_b) = +P T_b$$

definition of DPSK:-

DPSK is a non-coherent version of PSK. It eliminates the need of coherent reference signal at the receiver by combining two basic operations at the transmitter:

1. differential encoding of the input binary wave and
2. phase-shift keying, Hence the name, DPSK.

principle:- The i/p sequence of binary bit is modified such that the next bit depends upon the previous bit. Therefore in the receiver the previous received bits are used to detect the present bit.

Generation:-

- DPSK is a technique of BPSK, in which there is no reference phase signal. Here, the transmitted signal itself can be used as a reference signal.

- In DPSK, the input sequence is given to the XNOR gate and its output is again fed back as other input through

1-bit delay

- The output of XNOR is also given to balance modulator

- The output of XNOR is $d(t)$ and other signal $\phi(t)$ is given to the balance modulator.

- Hence, the output is the DPSK signal and here the phase shift changes in accordance to the XNOR gate.

Receiver:-

- The transmitted signal is applied as input to the receiver part. Hence the input signal is

$$s(t) = d(t) \sqrt{2P} \cos(2\pi f_c t + \theta)$$

- The input is given to the correlator along the one bit delay T_b

$$= d(t - T_b) \sqrt{2P} \cos(2\pi f_c (t - T_b) + \theta)$$

- The correlator has two parts, one is product modulator and other is the integrator

M-Ary Modulation Techniques :-

- BASK, BFSK, BPSK, DPSK, all are binary modulation techniques. In all these techniques they are represented using two symbols consisting of either 1 or 0.

- The maximum bandwidth of BASK is $2f_b$ and BFSK is $4f_b$ and BPSK is $2f_b$ and DPSK is f_b , to reduce the bandwidth two more bits are combined to form a one symbol.

- The no. of symbols is always greater than 2 i.e., given by

$$M = 2^n$$

Where n is the no. of bits combined

M is the total no. of symbols or levels

- The signal representation which uses the combination of two or more bits to represent a single symbol is known as M-ary (Multilevel signaling scheme).

- M level signal is mainly used to modulate the carrier and this technique is known as M-ary digital modulation technique.

- In this, one of the M possible symbols is transmitted during each symbol duration T_b is given by

$$T_f = T_b \log_2 M$$

where T_b is the 1 bit duration, then it can also be written as

$$T_f = nT_b$$

- Based on the modulation, M-ary are classified as

1. M-ary ASK
2. M-ary FSK
3. M-ary PSK
4. M-ary QAM

In M-ary ASK, they are M different amplitude levels of carrier as

$$s_i(t) = A_i \cos(2\pi f_c t)$$

where $A_i = (2^{1/M} - 1)A$

Let $M=4$ & $A=1$, then we have to get four signal amplitudes will be $-1V, +1V, -3V$ and $+3V$ volts, then the M-ASK signal will

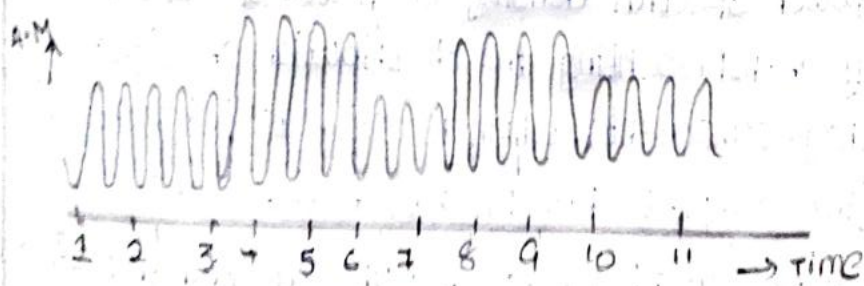
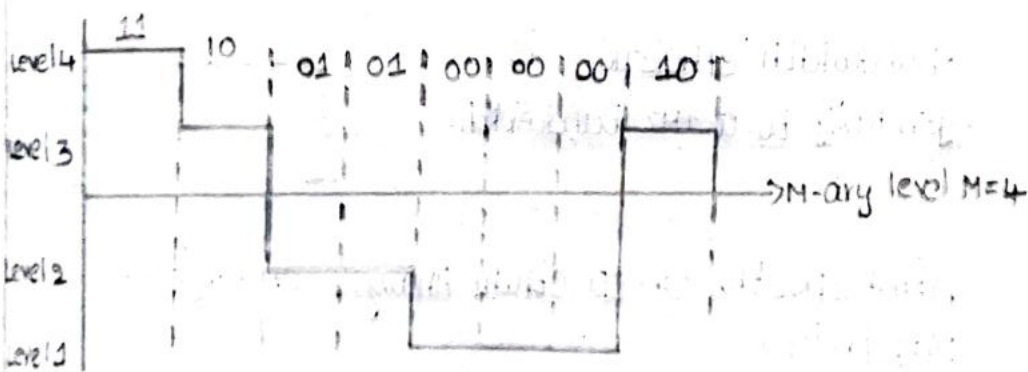
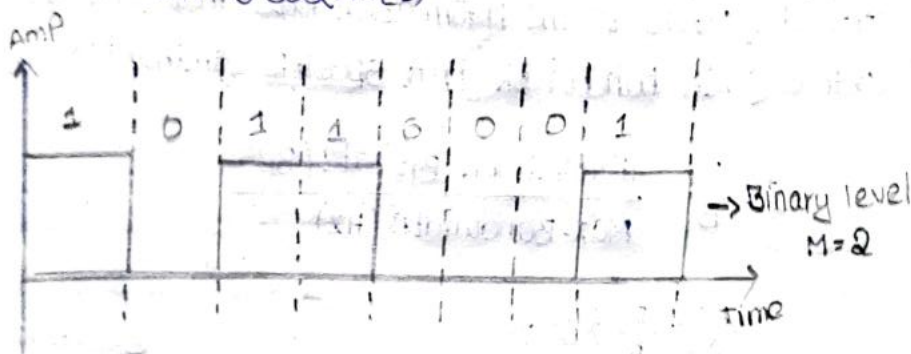
$$s_1(t) = A_i \cos(2\pi f_c t)$$

$$\Rightarrow s_1(t) = \cos(2\pi f_c t)$$

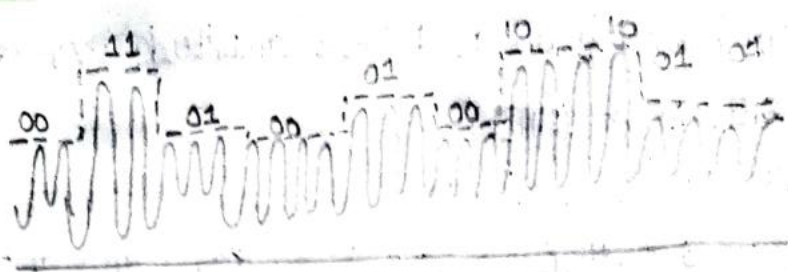
$$s_2(t) = -\cos(2\pi f_c t)$$

$$s_3(t) = -3 \cos(2\pi f_c t)$$

$$s_4(t) = 3 \cos(2\pi f_c t)$$



Binary 2-level 1 bit in 1 symbol duration.



4-ASK: 4 level 2 bit in 1 symbol duration

Bandw

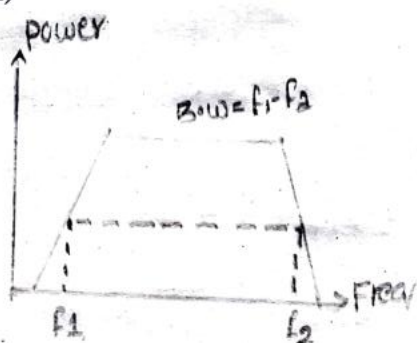
• The amount of data that can be transmitted over broadband in a given communication system (or)

Bandwidth efficiency (or) spectral efficiency (or) spectrum efficiency refers to the information rate that can be transmitted over a given bandwidth in a specific communication system.

$$\eta = \frac{R}{B} = \frac{\text{Transmission Bit rate (BPS)}}{\text{Max. Bandwidth (Hz)}}$$

$$\% \eta = \frac{R}{B} \times 100 \%$$

• Bandwidth efficiency is generally to a 1Hz bandwidth.



power spectra (or) spectrum Analysis of digital Modulation:-

- The power spectral density or power spectrum provides a way of representing the distribution of signal frequency components which is easier to interpret visually than the complex DFT (Discrete Fourier Transform).
- It is mainly used for to measure the dynamic frequency domain responses of an integrated circuit.
- Here, analysis is nothing but a spectrum of frequencies such as energies and eigen values etc.,

A Base Band Transmission and BaseBand Receiver:-

- A baseband signal or a baseband transmission transfer digital signal in the form of a train of pulses or a A baseband is the transmission of signal in its original and unmodulated signal.
- these type of transmission, only one path is available to send and receive digital signals between dividers.
- In these signals transfer in a baseband systems are not moved from their original frequency as no sinusoidal carrier wave is used because band pass signal is having high frequency without doing modulation, they directly send the signals or information.
- A baseband signal contains three types
 1. Multimedia Base Band signal
 2. Low level BBS
 3. High level BBS
- A multimedia is the combination of audio, video and other data.
- A low frequency baseband signal, a signal it contains low frequency usually it is below 20Hz.
- In high frequency baseband signal with high frequency components within the baseband range.
- A binary baseband signal logic-1 can be represented as +V and logic-0 can be represented as -V.
- A baseband receiver is used to detect the correct voltage level from the noisy mixed received signal.

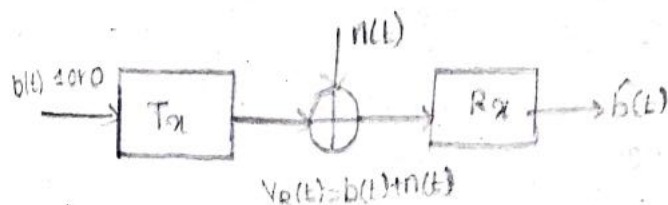
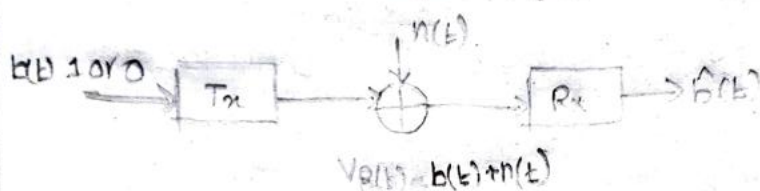


Fig:- Baseband transmission signal

operations:-

1. In baseband Transmission, the binary input 1 or 0 is applied to the transmitter.
2. The transmitter transmits the original signal to the channel
3. In channel some external noise is added to the signal and are given to the Receiver
4. The receiver detects the noise and allows only original binary signal to pass through it.

Optimal Reception of Digital signal :-



- The duty of the receiver is to detect the correct voltage level (logic level) from the noisy mixed received signal
- An optimal receiver is one that is design to minimize the probability of receiver that a decision maker occurs.
- The probability of increase in sampled value is equal to the decrease in sample value that means receiver detects $+V$ is the sampled value at that instance +ve and it will detect a $-V$, if the sampled is negative
- In digital communication system what the receiver receives is the sum of transmitter signal and channel noise
- The optimal reception of digital signal is based on the minimum error probability.
- A receiver is said to be optimum if it is minimum probability error
- The optimal receiver can be separated in two stages:-
 1. Detector which filters and sampled the received signals

2. Decision device, which uses the samples to make its decision.

probability of Error:-

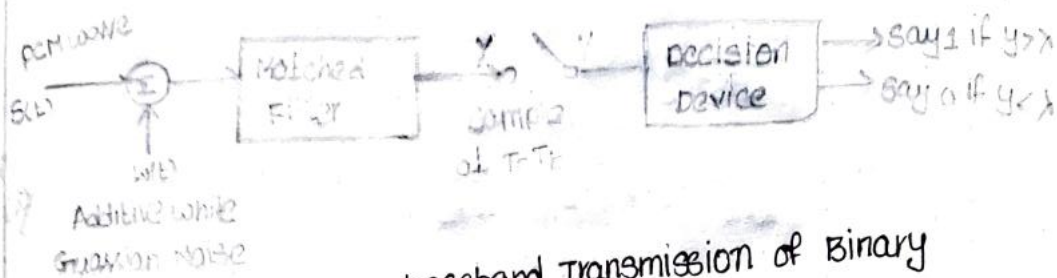


Fig:- Receiver for baseband transmission of Binary Encoded PCM wave using polar NRZ

Derivation:-

The output of matched filter is given by combination of two signals is

$$x(t) = +A + w(t)$$

$$x(t) = -A + w(t)$$

For Matched Filter $y = \int_0^{T_b} x(t) dt$

'0' way Trans $y = \int_0^{T_b} [-A + w(t)] dt$

$$= \int_0^{T_b} (-A) dt + \int_0^{T_b} w(t) dt$$

$$= -A[T_b - 0] + \int_0^{T_b} w(t) dt$$

$$= -AT_b + \int_0^{T_b} w(t) dt$$

$$= T_b \left[-A + \frac{1}{T_b} \int_0^{T_b} w(t) dt \right]$$

The random variable y is a gaussian distributed with a mean of $-A$ then the variance of the random variable y is

$$\sigma_y^2 = E[y + A]^2$$

$$= \frac{1}{T_b^2} E \left[\int_0^{T_b} w(t) dt \int_0^{T_b} w(u) du \right]$$

$$= \frac{1}{T_b^2} \int_0^{T_b} \int_0^{T_b} w(t) w(u) dt du$$

$$\therefore E[w(t) w(u)] = R_w(t, u)$$

$$R_w(t, u) = \frac{N_0}{2} \delta(t-u)$$

$$\sigma_y^2 = \frac{1}{T_b^2} \int_0^{T_b} \int_0^{T_b} \frac{N_0}{2} \cdot \delta(t-u) dt du$$

$$= \frac{N_0}{2T_b^2} \int_0^{T_b} (1) du$$

$$= \frac{N_0}{2T_b^2} \cdot [T_b - 0] = \frac{N_0}{2T_b}$$

• The conditional probability density function of random variable y given that symbol 0 has sent.

$$f_y(y|0) = \frac{1}{\sqrt{2\pi\sigma_y^2}} e^{-\frac{(y-\mu_y)^2}{2\sigma_y^2}}$$

$$\sigma_y^2 = \frac{N_0}{2T_b}, \mu_y = -A$$

$$f(y|0) = \frac{1}{\sqrt{2\pi \cdot \frac{N_0}{2T_b}}} e^{-\frac{(-y+A)^2}{2 \cdot \frac{N_0}{2T_b}}}$$

$$f(y|0) = \frac{1}{\sqrt{\pi \frac{N_0}{T_b}}} e^{-\frac{(y+A)^2}{N_0/T_b}}$$

The conditional probability error is given by

$$P_{10} = \int_{\lambda}^{\infty} f_y(y|0) dy$$

$$= \frac{1}{\sqrt{\pi \frac{N_0}{T_b}}} \int_{\lambda}^{\infty} e^{-\frac{(y+A)^2}{N_0/T_b}} dy$$

Let us consider $z = \frac{y+A}{\sqrt{\frac{N_0}{T_b}}}$

$$y = \infty, z = \infty$$

$$y = \lambda, z = \frac{A+\lambda}{\sqrt{N_0/T_b}}$$

$$P_{10} = \frac{1}{\sqrt{\pi \frac{N_0}{T_b}}} e^{-\frac{(y+A)^2}{N_0/T_b}}$$

$$= \frac{1}{\sqrt{\pi \frac{N_0}{T_b}}} \int_{\frac{A+\lambda}{\sqrt{\frac{N_0}{T_b}}}}^{\infty} e^{-z^2} dz \sqrt{\frac{N_0}{T_b}}$$

$$P_{10} = \frac{1}{\sqrt{\pi}} \int_{\frac{A+\lambda}{\sqrt{\frac{N_0}{T_b}}}}^{\infty} e^{-z^2} dz$$

The complementary error function is given by

$$\text{erfc}(u) = \frac{2}{\sqrt{\pi}} \int_u^{\infty} e^{-z^2} dz$$

$$\therefore P_{10} = \frac{1}{2} \text{erfc} \left(\frac{A+\lambda}{\sqrt{\frac{N_0}{T_b}}} \right)$$

similarly,

$$P_{01} = \frac{1}{2} \text{erfc} \left(\frac{-A+\lambda}{\sqrt{\frac{N_0}{T_b}}} \right)$$

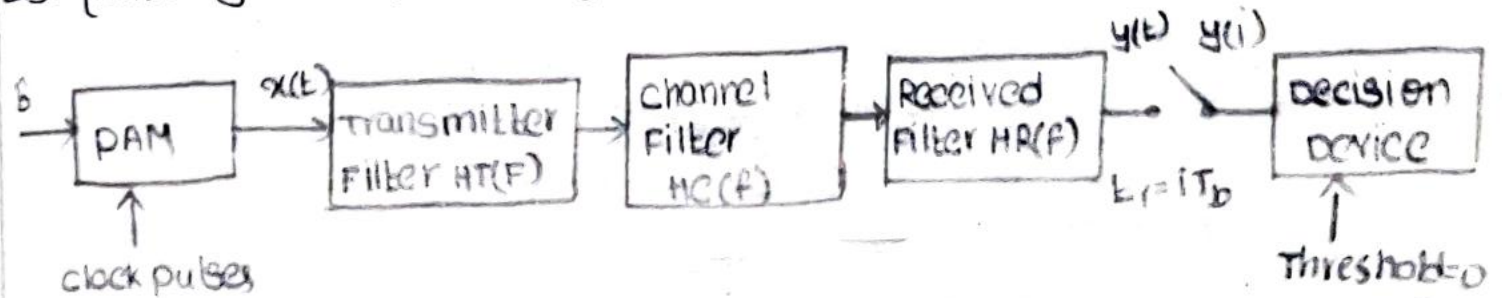
$$\text{Average probability error} = \frac{P_0}{2} \text{erfc} \left(\frac{A+\lambda}{\sqrt{\frac{N_0}{T_b}}} \right) + \frac{P_1}{2} \text{erfc} \left(\frac{-A+\lambda}{\sqrt{\frac{N_0}{T_b}}} \right)$$

$$P_1 = P_0 = \frac{1}{2}, \lambda = 0, A = -A = A$$

$$= \frac{1}{4} \text{erfc} \left(\frac{A}{\sqrt{\frac{N_0}{T_b}}} \right) + \frac{1}{4} \text{erfc} \left(\frac{A}{\sqrt{\frac{N_0}{T_b}}} \right)$$

$$\therefore P_e = \frac{1}{2} \text{erfc} \left(\frac{A}{\sqrt{\frac{N_0}{T_b}}} \right)$$

ISI [Intersymbol Interference]:-



Transmitter channel Receiver

$$x(t) = \sum_{k=-\infty}^{\infty} A_k g(t - kT_b)$$

$$y(t) = u \sum_{k=-\infty}^{\infty} A_k p(t - kT_b) + n(t)$$

Where u = scaling Function

A_k = Amplitude

$p(t)$ = shape of the pulse

$$A_k g(t) = u A_k p(t) = A_k g(t) h(t)$$

$$u A_k p(f) = A_k G(f) H(f) = A_k G(f) H_T(f) H_C(f) H_R(f)$$

Threshold = 0
choose 1 if $y(t) > \lambda$
choose 0 if $y(t) < \lambda$

$$H(f) = H_T(f) H_C(f) H_R(f)$$

$$y(t_i) = \mu \sum_{k=-\infty}^{\infty} A_k p(t_i - kT_b) + n(t_i)$$

$$y(t_i) = \mu \sum_{k=-\infty}^{\infty} A_k p(iT_b - kT_b) + n(t_i)$$

$$= \mu \sum_{k=-\infty}^{\infty} A_k p((i-k)T_b) + n(iT_b)$$

$$= \mu A_i p(0) + \mu \sum_{k=-\infty}^{\infty} A_k p((i-k)T_b) + n(iT_b)$$

$$y(t_i) = \mu A_i + \mu \sum_{k=-\infty}^{\infty} A_k p((i-k)T_b) + n(t_i)$$

1st transmitted i^{th} signal bit

$$y(t_i) = \mu A_i$$

Intersymbol Interference:-

- ISI is a form of a distortion of a signal i.e., one symbol is interfere with subsequent symbol.
- ISI is used to identify error in detected signal.

⇒

consider a binary baseband PAM system as shown in the figure. The input binary data sequence (b_k) is applied to the PAM. The PAM generates the pulse waveform $x(t)$. This waveform can be represented mathematically as

$$x(t) = \sum_{k=-\infty}^{\infty} A_k g(t - kT_b)$$

Here T_b is the duration of each 1/p binary bit, $g(t)$ is the shaping pulse and

$$A_k = \begin{cases} +a & \text{if } b_k = 1 \\ -a & \text{if } b_k = 0 \end{cases}$$

Since polar NRZ signaling scheme is considered. The signal $x(t)$ is then passed through the transmitting filter. The transmitting filter combines all the necessary transmitting circuits & systems. The combined transfer function of transmitting filter is $H_T(f)$. The signal is then passed through the channel having transfer function $H_C(f)$.

The channel delivers the signal to the receiver filter. It consists of all the necessary receiver circuits and systems. The combined function of receiving filter is $H_R(f)$. The output of receiving filter is $y(t)$. This $y(t)$ is noisy replica of the transmitted signal $x(t)$.

- The signal $y(t)$ is sampled synchronously to the transmitter. The sampling instants are $t_i = iT_b$. These sampling instants are synchronous to the clock pulse at the transmitter. The sampled signal $y(t_i)$ is then given to the decision device.
- The decision device compares the 1/p signal with threshold ' λ '. Then the decision is taken as follows:

If $y(t_i) > \lambda$ select symbol '1'

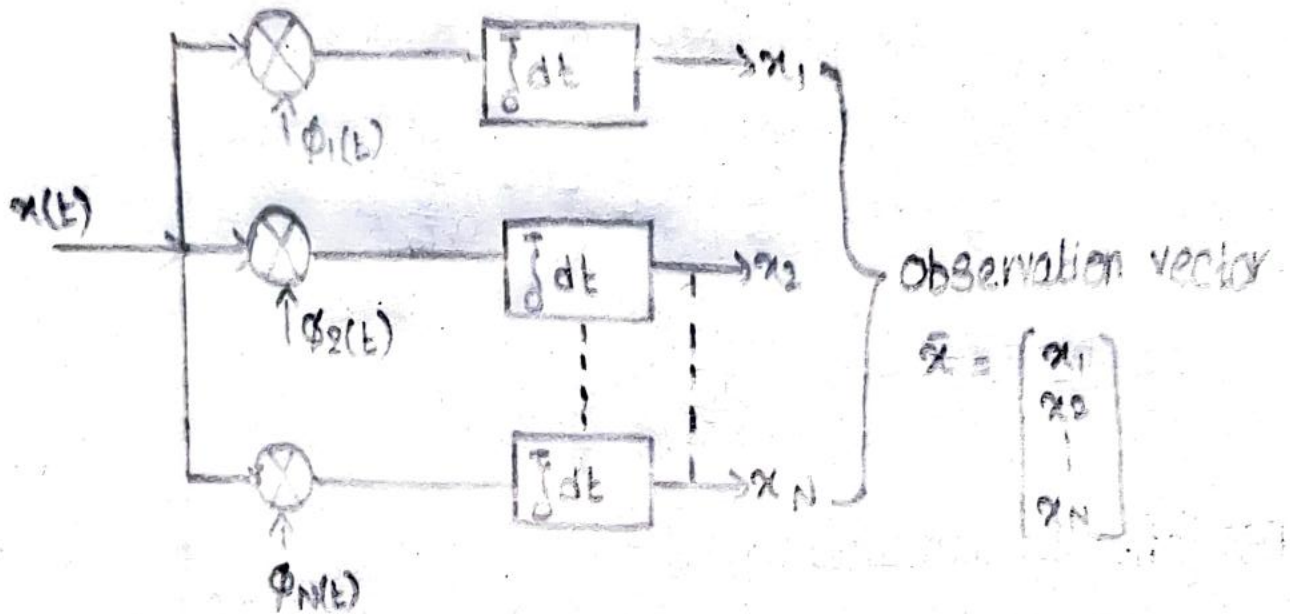
$y(t_i) < \lambda$ select symbol '0'.

Remedy:-

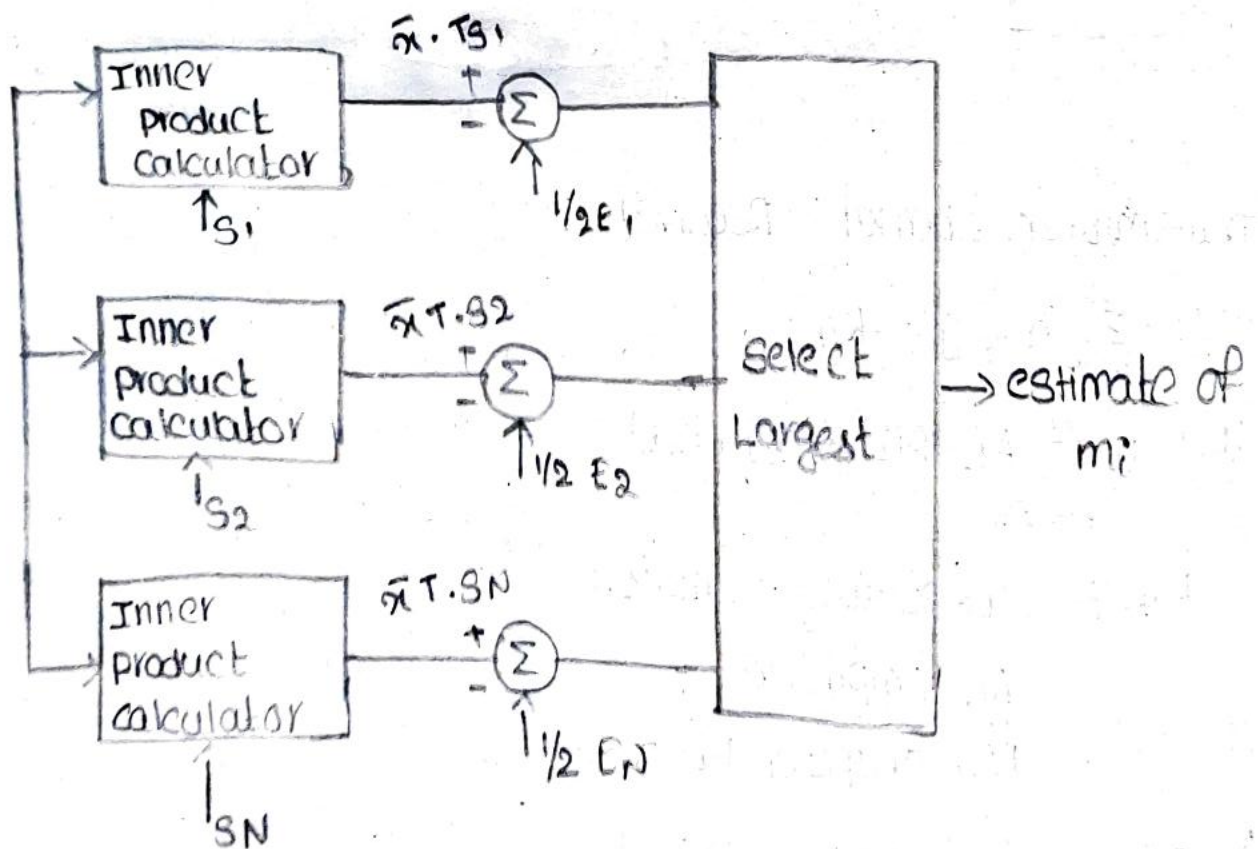
- The ISI can be reduced by proper design of pulse spectrum $G(f)$, transmit filter, receive filter $H_R(f)$ and channel $H_C(f)$.
- It has proved that the function which proves a zero ISI is a sinc function.

correlation Receiver:-

1. coherent detector



2. signal transmission Encoder:-



• The correlation receiver consists of two sub systems:-

(i) correlation receiver:-

The correlation detector is shown in below figure. It consists of a bank of M product integrators or correlators, supplied with a corresponding set of coherent orthonormal basic function $\phi_1(t), \phi_2(t), \dots, \phi_N(t)$. This bank of correlators operates on the received signal $x(t)$, $0 \leq t < 1$ to produce the observation vector $\bar{x}$.

(ii) signal transmission decoder:- It is implemented in form of a maximum-likelihood decoder that operates on the observation vector, to produce an estimate $\hat{m}$ of the transmitted signal (or symbol) m_i , $i = 1, 2, \dots, M$, in a way that would minimize the average probability of symbol error.

The " N " elements of the observation vector $\bar{x}$ are first multiplied by the corresponding N elements of each of the signal vector $s_1, s_2, \dots, s_m$ and the resulting products are successfully summed in a accumulator. The inner products are corrected for the fact that transmitted signal energies may be unequal. Finally the resulting set of numbers is selected.

