3. ANGLE MODULATION

Introduction :- In Amplitude modulation Amplitude of carrier wave is varied according to the amplitude of Information signal.

→ The carrier wave has 3 characteristics i.e Amplitude, frequency and phase.

→ In Angle Modulation Instead of Amplitude, either frequency or phase of the carrier wave can be changed inaccordance with message signal keeping Amplitude constant is called Angle Modulation.

→ Frequency of carrier used to vary inaccordance with message signal is known as Frequency Modulation

→ Angle of carrier wave used to vary inaccordance with message signal/ Information signal is known as phase Modulation.

Frequency Modulation :- Frequency of carrier signal is varied inaccordance with frequency of message signal.

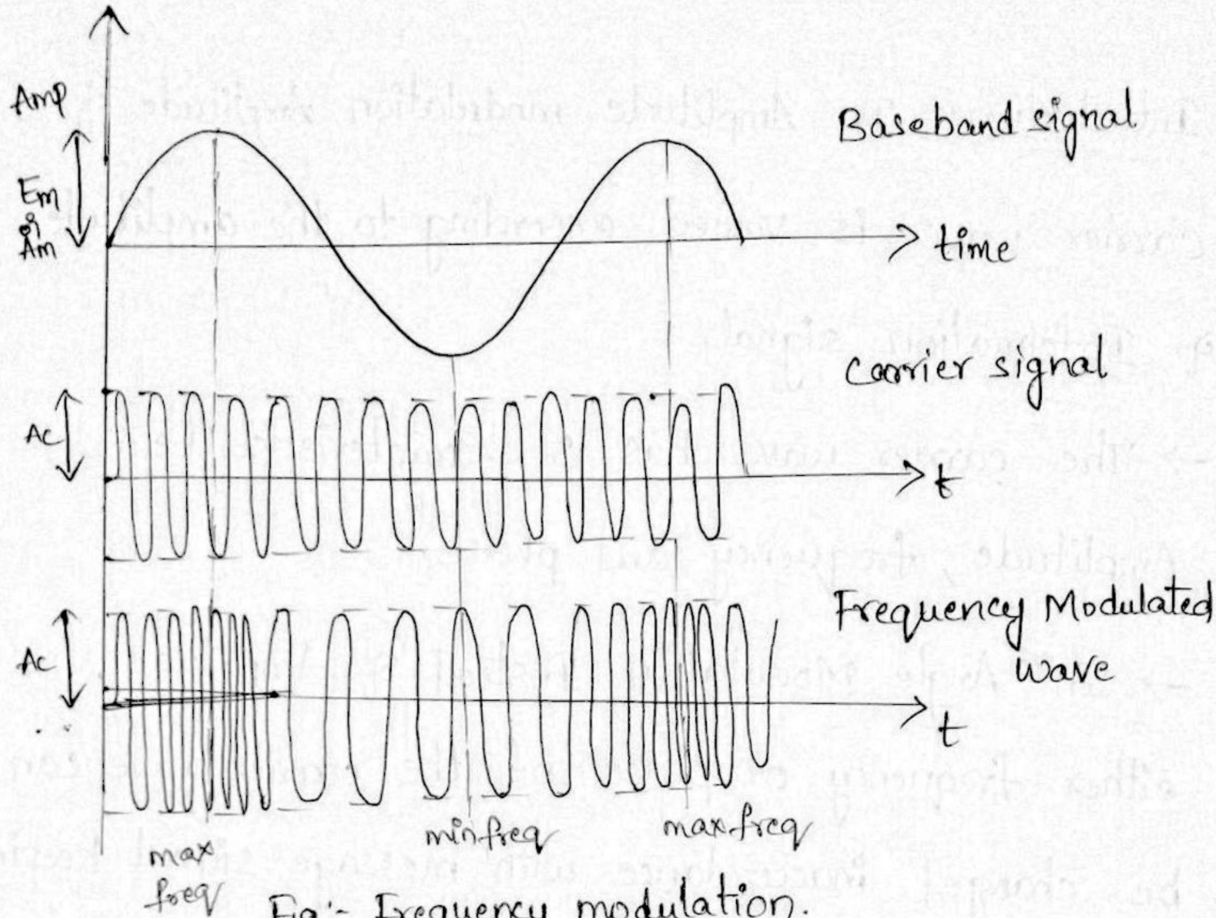

Fig:- Frequency modulation.

→ Instantaneous frequency $f_i(t)$ Varied linearly with message signal as expressed as

$$f_i(t) = f_c + k_f m(t) \quad \text{--- (1)}$$

f_c → frequency of unmodulated carrier

k_f → frequency sensitivity of modulator

Assuming that $m(t)$ is a voltage waveform

$$f_i(t) = \frac{1}{2\pi}\frac{d\theta}{dt} \Rightarrow \frac{d\theta}{dt} = 2\pi f_i(t)$$

Integration on both sides we get

$$\theta(t) = \int_0^t 2\pi f_i(t)\,dt$$

From eq ① we get $f_i(t)$

$$\theta(t) = \int_0^t 2\pi(f_c + k_f\, m(t))\,dt$$

$$= 2\pi f_c \int_0^t dt + 2\pi k_f \int_0^t m(t)\,dt$$

$$\theta(t) = 2\pi f_c t + 2\pi k_f \int_0^t m(t)\,dt$$

In time domain frequency modulated wave can be written as $s(t) = A_c \cos[\theta(t)]$

$$s(t) = A_c \cos\left[2\pi f_c t + 2\pi k_f \int_0^t m(t)\,dt\right]$$

<u>phase Modulation</u> :- phase of high frequency carrier signal is varied inaccordance with the phase of message signal.

Angular arguement $\theta(t)$ is varied linearly with the message signal is $\theta(t) = 2\pi f_c t + k_p\, m(t)$

$k_p \rightarrow$ phase sensitivity. Expressed radians per volt phase modulated wave in time domain

$$s(t) = A_c \cos[\theta(t)]$$
$$= A_c \cos(2\pi f_c t + K_p m(t))$$

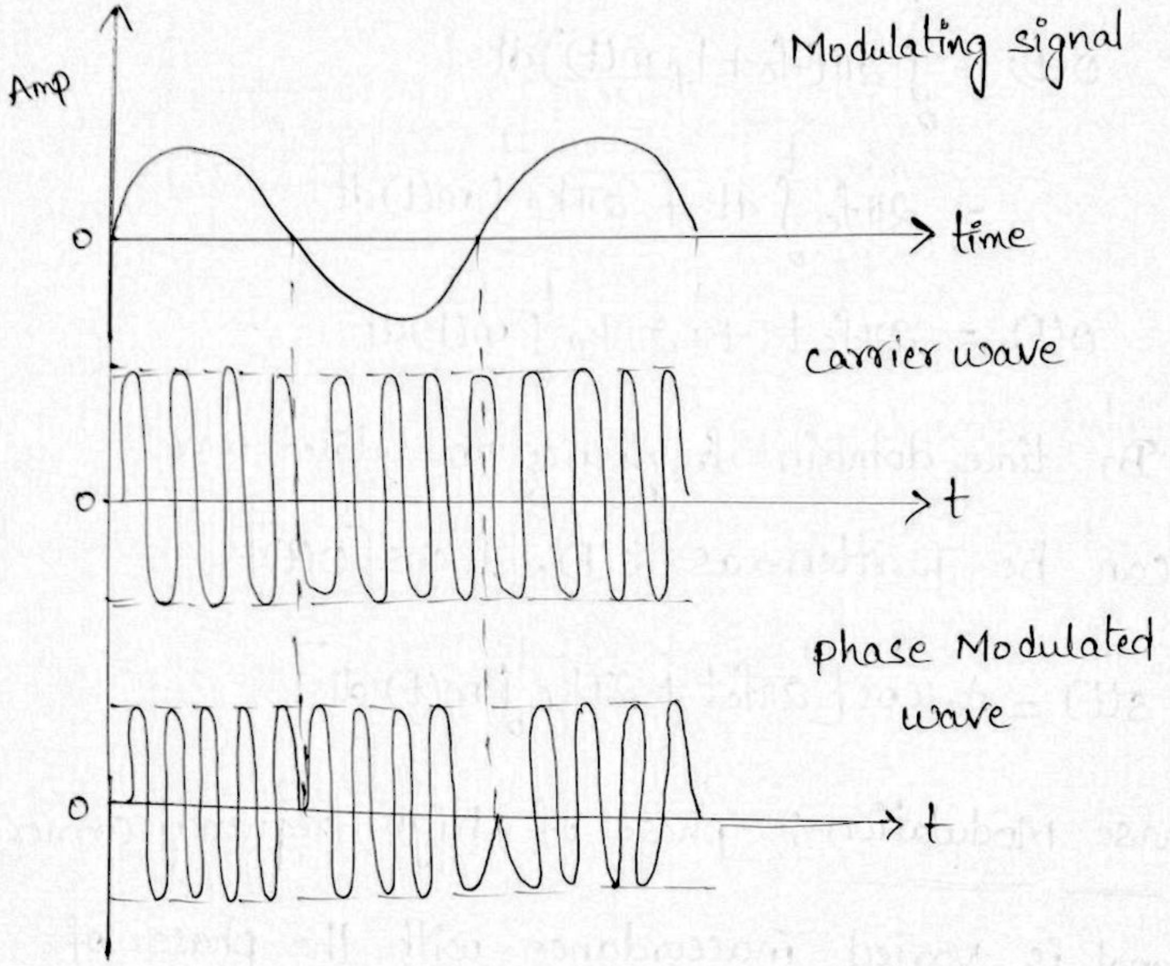

Fig:- phase Modulation.

Relation between frequency and phase Modulation :-

PM $s(t) = A_c \cos(2\pi f_c t + K_p m(t))$

FM $s(t) = A_c \cos(2\pi f_c t + 2\pi K_f \int_0^t m(t) dt)$

→ The 2 eq PM & FM not only very similar but are inseparable

1- FM using PM :-

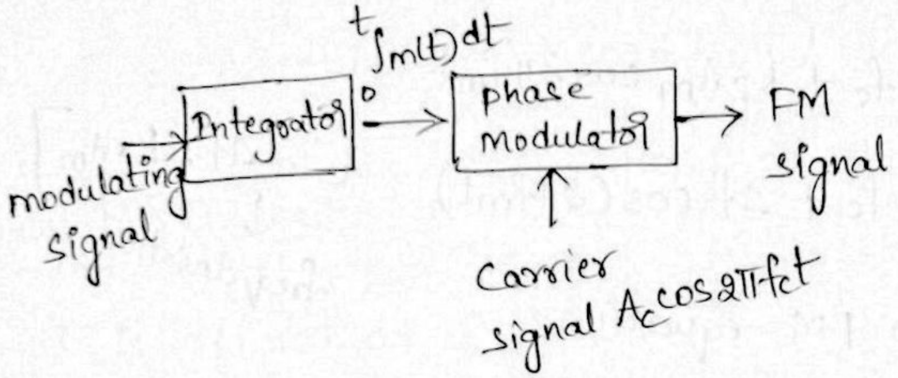

Fig:- Generation of FM using phase modulator

→ To Get FM integrate the modulating/baseband signal and then apply to phase modulator

2 PM using FM :-

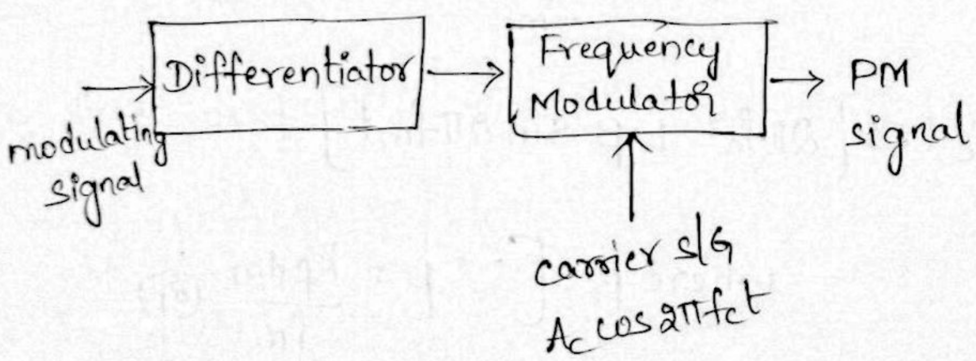

Fig:- Generation of PM using Frequency modulator

Single Tone frequency Modulation :-

Instantaneous equation of Resulting FM wave

$$f_i(t) = f_c + k_f\, m(t) \quad \text{where } m(t) = A_m \cos 2\pi f_m t$$

$$= f_c + k_f A_m \cos 2\pi f_m t$$

$$= f_c + \Delta f \cos(2\pi f_m t) \qquad \left[\because \underset{\text{freq deviation}}{\underbrace{\Delta f}} = k_f A_m\right]$$

we know FM equation

$$s(t) = A_c \cos\left[2\pi f_c t + 2\pi k_f \int_0^t m(t)\,dt\right]$$

$$= A_c \cos\left[2\pi f_c t + 2\pi k_f \int_0^t A_m \cos 2\pi f_m t\right]$$

$$= A_c \cos\left[2\pi f_c t + \cancel{2\pi} k_f \cdot A_m \left(\frac{\sin 2\pi f_m t}{\cancel{2\pi} f_m}\right)_0^t\right]$$

$$= A_c \cos\left[2\pi f_c t + \frac{k_f A_m}{f_m} \sin 2\pi f_m t\right]$$

$$= A_c \cos\left[2\pi f_c t + \beta \sin 2\pi f_m t\right]$$

where β $\left[\because \beta = \frac{k_f A_m}{f_m} \text{ (or) } \beta = \frac{\Delta f}{f_m}\right]$

β = modulation Index

④

Single Tone phase Modulation :-

we know PM eq $s(t) = A_c \cos(2\pi f_c t + k_p m(t))$

$$= A_c \cos(2\pi f_c t + k_p A_m \cos 2\pi f_m t)$$

$$= A_c \cos(2\pi f_c t + \beta \cos 2\pi f_m t)$$

$$\left[\because \beta = k_p A_m \text{ or } \Delta\phi = k_p \cdot A_m\right]$$

Modulation Index :-

→ It is defined as Ratio of frequency deviation to the modulating frequency

$\beta = \frac{\Delta f}{f_m}$ Interms of FM

$\beta = k_p A_m$ Interms of PM

Spectrum Analysis of sinusoidal FM wave :-

→ consider sinusoidal modulating voltage signal at a single frequency, frequency modulating carrier having a frequency f_c

Eq of FM wave is given as

$$s(t) = A_c \cos\left[2\pi f_c t + 2\pi k_f \int_0^t m(t)\,dt\right]$$

In order to define Bandwidth of FM, let us

define $a(t) = \int_0^t 2\pi m(t)\,dt$ —— ①

$$\hat{s}(t) = A_c e^{j(2\pi f_c t + k_f a(t))}$$

$$= A_c \cdot e^{j2\pi f_c t} \cdot e^{jk_f a(t)}$$

$$s(t) = R_e \hat{s}(t)$$

Expanding exponential in power series i.e

$$\hat{s}(t) = A_c\left[1 + \frac{j k_f a(t)}{1} + \frac{j^2 (k_f a(t))^2}{2!} + \cdots + \frac{j^n k_f^n a^n(t)}{n!}\right] \cdot e^{j2\pi f_c t}$$

→ modulated wave consists of Unwanted carrier and Various amplitude modulated terms

→ Theoretically FM wave consist of infinite no. of sideband, occupying infinite Bandwidth

→ Eventhough theoretical BW of FM wave is infinite, most of the modulated signal power is in a finite Bandwidth

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Let us obtain the Bandwidth for a specific case of single tone modulation

consider message signal $m(t) = A_m \cos 2\pi f_m t$

from eq (1) $a(t) = \int_0^t 2\pi\, m(t)\, dt$

$$= \int_0^t 2\pi A_m \cos 2\pi f_m t \, dt$$

$$= 2\pi A_m \int_0^t \cos 2\pi f_m t \, dt$$

$$a(t) = \cancel{2\pi} \cdot A_m \cdot \frac{\sin 2\pi f_m t}{\cancel{2\pi} f_m} = \frac{A_m}{f_m} \sin 2\pi f_m t$$

we know that $\hat{s}(t) = A_c e^{j[2\pi f_c t + k_f a(t)]}$

$$\hat{s}(t) = A_c e^{j(2\pi f_c t + k_f \frac{A_m}{A_f} \sin 2\pi f_m t)}$$

$$= A_c e^{j(2\pi f_c t + \beta \sin 2\pi f_m t)}$$

$$e^{j\beta \sin 2\pi f_m t} = \sum_{n=-\infty}^{\infty} C_n e^{j 2\pi f m t}$$

using Bessel function, the equation for FM is

$$s(t) = A_c \left[J_0(\beta) \sin w_c t + J_1(\beta) \sin(w_c + w_m)t - \sin(w_c - w_m)t + J_2(\beta) \sin(w_c + 2w_m)t - \sin(w_c - 2w_m)t \right] + \cdots \Big]$$

→ Thus modulated signal has a carrier component and infinite no. of side frequencies

From above equation $J_n(\beta)$ is negligible for $n > \beta+1$ where $\beta+1$ is no. of significant sideband

$$\text{Bandwidth of FM} = 2nf_m$$

$$= 2(\beta+1)f_m$$

$$= 2\beta f_m + 2f_m$$

$$\text{w.k.T } \beta = \frac{\Delta f}{f_m} \Rightarrow 2\frac{\Delta f}{\cancel{f_m}} \times \cancel{f_m} + 2f_m$$

$$= 2\Delta f + 2f_m = 2(\Delta f + f_m)$$

$$= 2\Delta f + \text{BW of msg s/G}$$

Types of FM :- Interms of Bandwidth. It is 2 types

1. Narrow band FM
2. wide band FM

Narrow band FM :- In this frequency Modulation, frequency band is narrow i.e Bandwidth is limited as same as AM wave $\beta < 1 \rightarrow$ NBFM we know FM eq from single tone frequency

$$s(t) = A_c \cos(2\pi f_c t + \beta \sin 2\pi f_m t)$$

$$\cos(A+B) = \cos A \cos B - \sin A \sin B$$

$$= A_c[\cos 2\pi f_c t \cdot \cos\beta \sin 2\pi f_m t - \sin 2\pi f_c t \cdot \sin\beta \sin 2\pi f_m t]$$

here $\cos\beta \sin 2\pi f_m t = 1 \rightarrow \cos\theta = 1$

$\sin\beta \sin 2\pi f_m t = \beta \sin 2\pi f_m t \rightarrow \sin\theta = \theta$

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

$$s(t) = A_c \cos 2\pi f_c t - A_c \sin 2\pi f_c t \cdot \beta \sin 2\pi f_m t$$

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

$$= A_c \cos 2\pi f_c t - \frac{\beta A_c}{2}[\cos(2\pi f_c t - 2\pi f_m t) - \cos(2\pi f_c + 2\pi f_m)t]$$

$$s(t) = A_c \cos 2\pi f_c t - \frac{\beta A_c}{2} \cos 2\pi (f_c - f_m)t + \frac{\beta A_c}{2} \cos 2\pi (f_c + f_m)t$$

The above expression is similar to that of Am wave i.e Am equation is

$$s(t) = A_c \cos 2\pi f_c t + \frac{\mu A_c}{2} \cos 2\pi (f_c - f_m)t + \frac{\mu A_c}{2} \cos 2\pi (f_c + f_m)t$$

The difference in the NBFM has sign reversal of lower side band when compared to AM

→ In AM total power $P_T = P_c\left(1+\frac{\mu^2}{2}\right)$

lly in NBFM total power $P_T = P_c\left(1+\frac{\beta^2}{2}\right)$

→ comparing NBFM with AM similarities are Bandwidth, sideband and power calculation freq spectrum. NBFM $BW = 2f_m$

→ Taking Fourier Transform of our NBFM we get

$$s(f) = \frac{A_c}{2}\left[\delta(f-f_c)+\delta(f+f_c)\right] - \frac{\beta A_c}{4}\left\{\delta(f-(f_c-f_m)) + \delta(f+(f_c-f_m))\right\} + \frac{\beta A_c}{4}\left\{\delta(f-(f_c+f_m)) + \delta(f+(f_c+f_m))\right\}$$

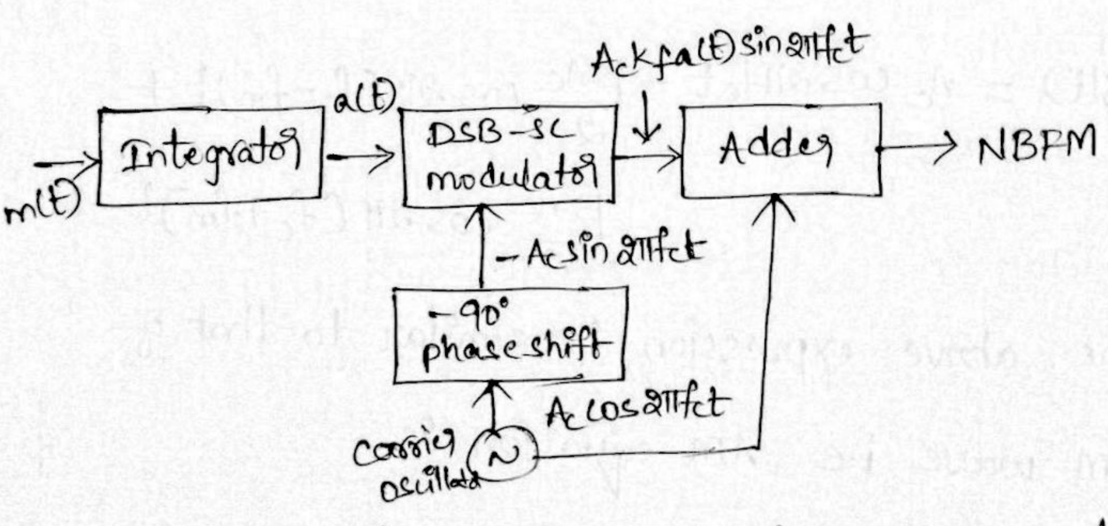

Fig:- Generation of NBFM using DSBSC modulator

→ The Narrowband PM wave is also given by $s(t) = A_c\left[\cos 2\pi f_c t - k_p m(t) \sin 2\pi f_c t\right]$

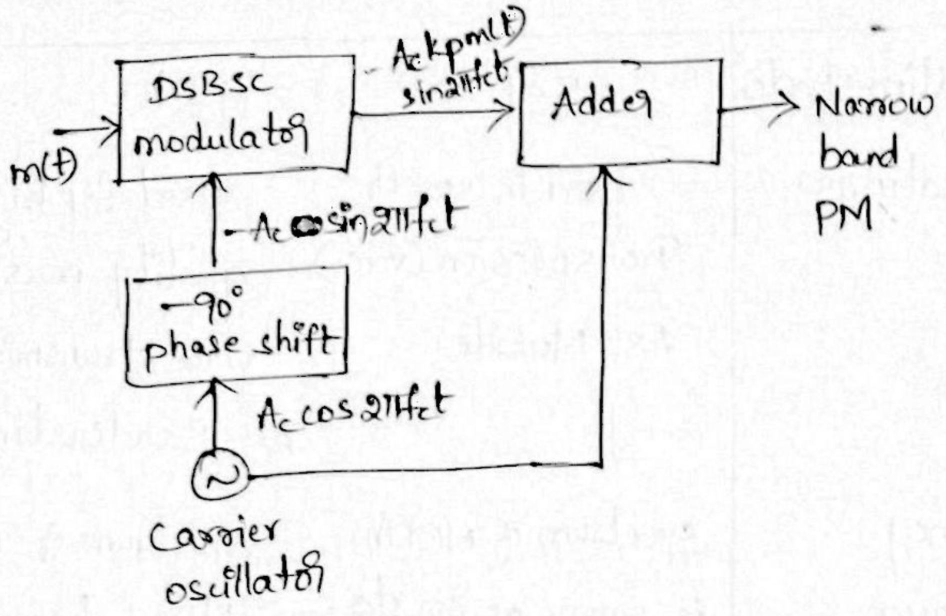

Fig:- Generation of NBPM using DSBSC modulator

Wideband FM :-

→ In this FM, frequency band is wide i.e Bandwidth is infinity. using this frequency band Multiple no. of signals are transmitted to the same channel.

→ Bessel Functions are suitable for the WBFM

→ write down the equation part of spectrum Analysis of sinusoidal FM wave

Comparision between NBFM and WBFM :-

Parameter	NBFM	WBFM
1. Modulation Index	$\beta < 1$	$\beta > 1$
2. Applications	used in speech Transmission (Voice) Ex:- Mobile	used for high quality music & Voice transmission Ex:- Broadcasting
3. Frequency spectrum	spectrum of NBFM is same as AM s/G	spectrum of WBFM differs from AM
4. Bandwidth	small ($\sim$ AM)	Very Large (i.e 15 times higher than BW of NBFM)
5. Modulating freq range	30 Hz – 3 kHz	30 Hz – 15 kHz
6. maximum duration (Δf)	5 kHz	75 kHz

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Constant Average power :-

→ The envelop of the FM wave is constant, the average power of this wave is also constant.

→ Power in Angle Modulated wave either PM or FM is always proportional to $\left(\frac{A_c}{\sqrt{2}}\right)^2$ regardless of value of k_p or k_f

$$P = \frac{A_c^2}{2}$$

Transmission Bandwidth of FM wave :-

→ Theoretically FM wave contains an infinite number of side frequencies. so that Bandwidth required for transmission is also infinite.

$$\text{Bandwidth of FM} = 2(\beta+1) f_m \qquad \beta = \frac{\Delta f}{f_m}$$

$$= 2\frac{\Delta f}{\cancel{f_m}} \times \cancel{f_m} + 2f_m$$

$$= 2(\Delta f + f_m)$$

Formulae for calculating Bandwidth of FM signal is $B_T = 2f_m + 2\Delta f$ simply called carsons rule

Generation of FM :-

→ There are two basic methods for generating frequency modulated waves

1. Direct FM
2. Indirect FM

Direct FM :- In this carrier frequency is directly varied according to the incoming message signal

→ Again in Direct FM Reactance modulators and Varactor diode modulator is used to generate our required FM.

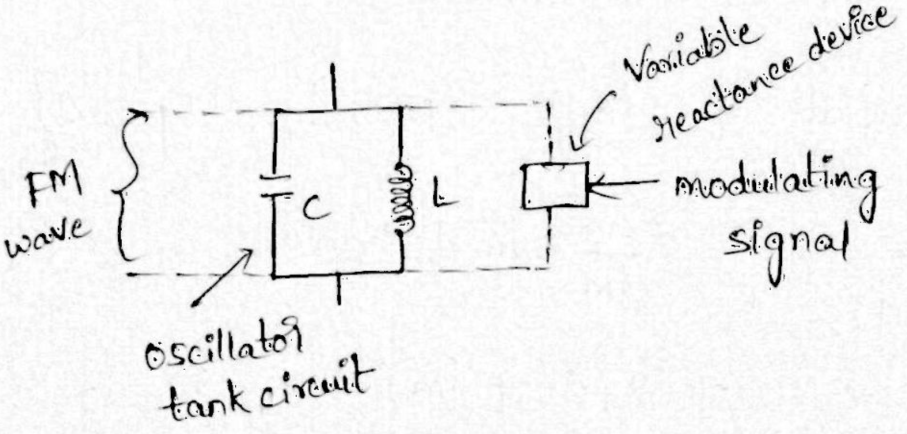


Fig:- Principle of reactance modulator.

→ Instantaneous frequency of the carrier wave is varied according to the modulating signal. For this device called voltage controlled oscillator is used.

→ we use a circuit that converts a modulating voltage into a corresponding change in capacitance or Inductance

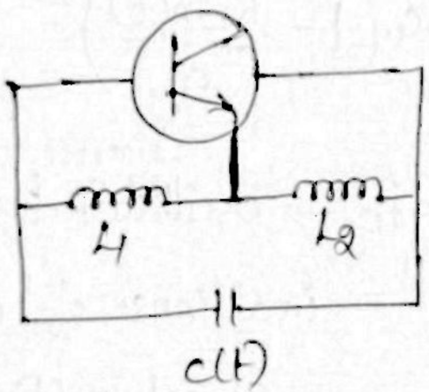


Fig:- Hartley oscillator

frequency of oscillation of hartly oscillator

$$f_i(t) = \frac{1}{2\pi\sqrt{(L_1+L_2)\,c(t)}}$$

where $c(t)$ = total capacitance of fixed capacitor

L_1+L_2 = 2 Inductance in the tuned circuit

→ For modulating wave the capacitance is given by $C(t) = C_0 - K_c m(t)$ $K_c \rightarrow$ Variable capacitor sensitivity

$$f_i(t) = \frac{1}{2\pi\sqrt{(L_1+L_2)(C_0-K_c m(t))}}$$

$$= \frac{1}{2\pi\sqrt{C_0(L_1+L_2) - K_c m(t)(L_1+L_2)}}$$

$$= \frac{1}{2\pi\sqrt{(L_1+L_2)C_0}\left(1-\frac{K_c m(t)}{C_0}\right)^{1/2}}$$

Let $\frac{1}{2\pi\sqrt{(L_1+L_2)C_0}} = f_0 \rightarrow$ oscillator frequency in absense of modulating signal $m(t)=0$

$$\therefore f_i(t) = f_0\left[1 - \frac{K_c}{C_0} m(t)\right]^{-1/2}$$

→ maximum change in capacitance produced by modulating wave is small with the unmodulated capacitance.

$$f_i(t) = f_0\left[1 + \frac{K_c}{2C_0} m(t)\right]$$

$$= f_0 + \frac{f_0 k_c}{2C_0} m(t) \quad \left[\because \frac{f_0 k_c}{2C_0} = k_f\right]$$

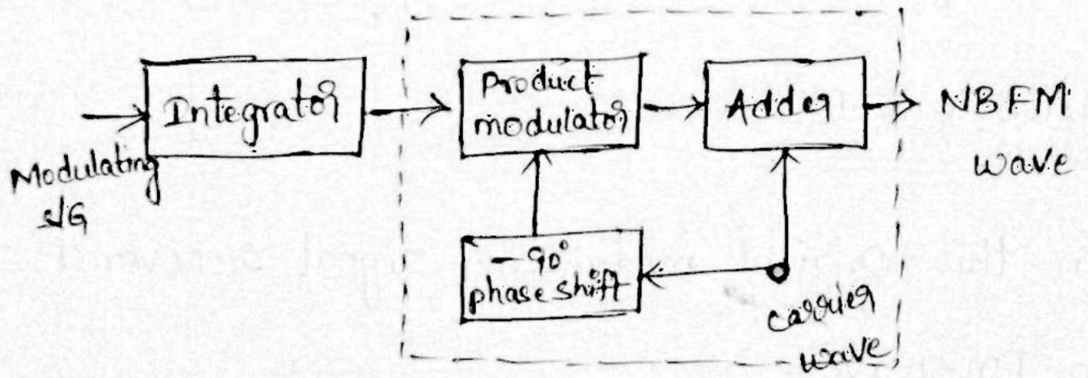
$$f_i(t) = f_0 + k_f m(t) \qquad k_f \rightarrow \text{frequency sensitivity}$$

<u>Disadvantage</u>:- carrier frequency is not obtained from a highly stable oscillator.

→ In practical, problem solved by stable frequency generated by a crystal will be able to control carrier frequency.

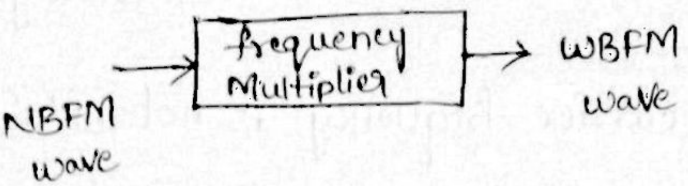
<u>Indirect FM</u>:-

→ In this FM, modulating signal first produces a narrow band FM wave.



Narrow band phase modulator

→ This Narrowband FM wave have no. of drawbacks, so NBFM is given to input of another device to get WBFM.



$$s(t) = A_c \cos\left(2\pi f_c t + 2\pi k_f \int_0^t m(t)dt\right)$$

$$= A_c\left[\cos 2\pi f_c t \cdot \cos 2\pi k_f \int_0^t m(t)dt - \sin 2\pi f_c t \cdot \sin 2\pi k_f \int_0^t m(t)dt\right]$$

for NBFM $\cos 2\pi k_f \int_0^t m(t)\,dt << 1$

$\cos(A+B) = \cos A \cos B - \sin A \sin B$

$\sin\theta = \theta,\ \cos\theta = 1$

$$s(t) = A_c\left[\cos 2\pi f_c t - \sin 2\pi f_c t \cdot 2\pi k_f \int_0^t m(t)dt\right]$$

Detection of FM :-

→ In this original modulating signal recovered from FM wave

→ Two methods used for frequency demodulation.

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1. Direct Method
   - 1. Frequency Discriminator
   - 2. Zero crossing detectors

2. Indirect Method → phase Locked Loop.

Frequency Discriminators / simple slope detector :-

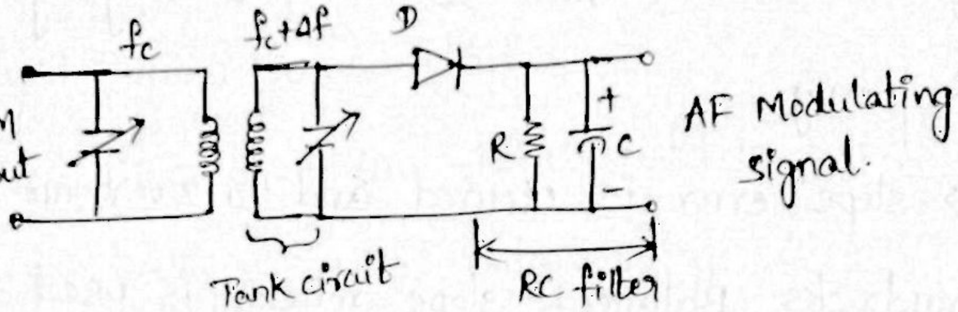


Fig :- Simple slope detector

→ The output voltage of tank circuit is then Applied to a simple diode detector with RC Load with proper time constant.

→ This detector is identical to AM diode detector

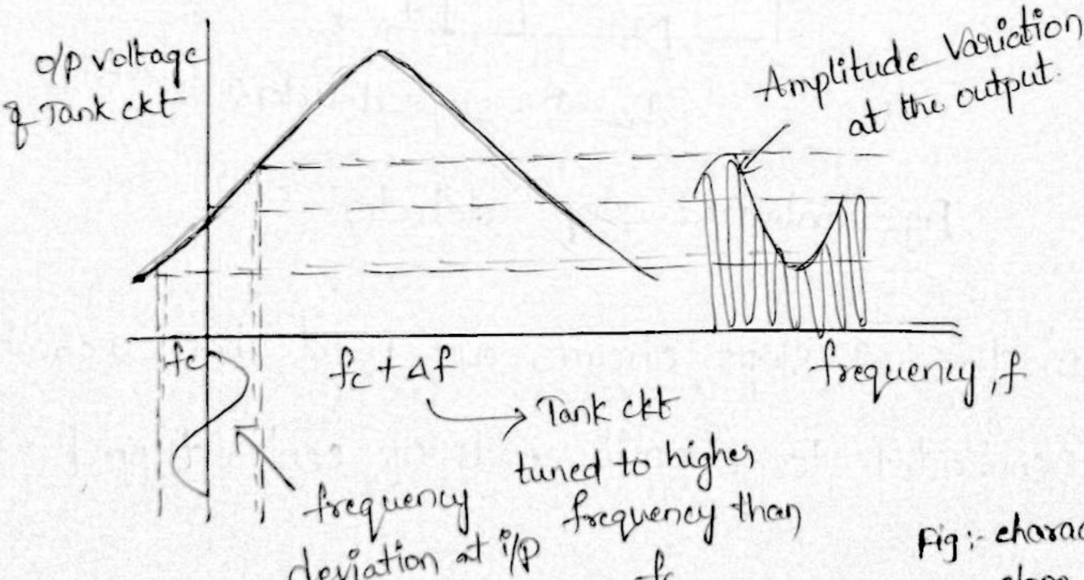


Fig :- characteristics of slope detector

Disadvantages :-

1. It is Inefficient

2. Linear over a limited frequency range.

3. Difficult to adjust primary and secondary winding of Transformer must be tuned to slightly different frequency

→ slope error is occured and to overcome these drawbacks Balanced slope detector is used.

Balanced slope detector :-

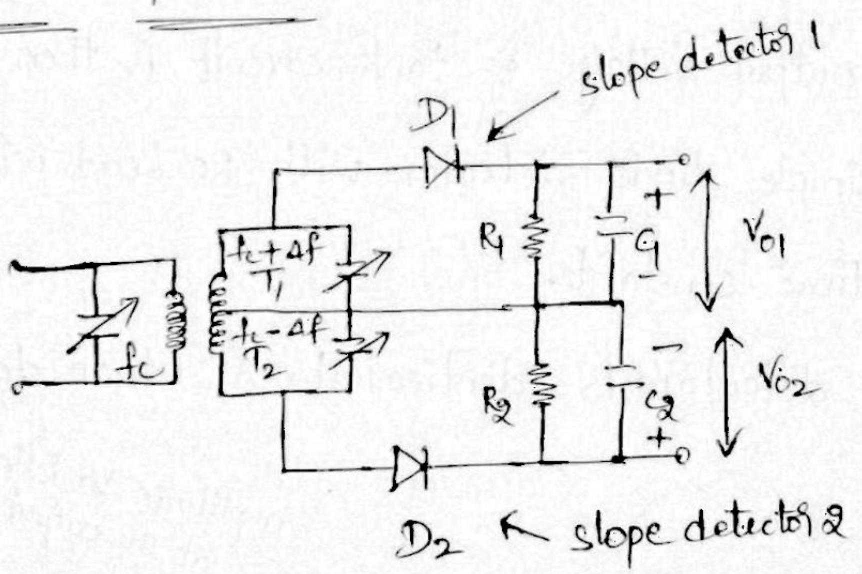


Fig:- Balanced slope detector

→ In this 2 slope circuits are used. These 2 circuits are connected to opposite ends of center tapped transformer. Due to this inputs to these circuits are 180° out of phase.

→ There are 3 tuned circuits.

→ Primary tuned to IF (f_c). upper tuned circuit T_1 tuned above f_c by Δf i.e $f_c + \Delta f$. Lower tuned circuit T_2 tuned below f_c i.e $f_c - \Delta f$

→ R_1C_1 and R_2C_2 are the filters used to bypass the RF ripple voltages. Final output voltage is given

$$V_0 = V_{01} - V_{02}$$

<u>working</u>:- To understand circuit operation input frequency into 3 Ranges

(i) $f_{in} = f_c$ induced voltage in T_1 winding = induced voltage in T_2 winding. Input voltages to both the diodes D_1 and D_2 will be same

Net output voltage is zero because V_{01} and V_{02} have opposite polarities.

(ii) $f_c < f_{in} < f_c + \Delta f$ induced voltage in $T_1 > T_2$. Therefore input to D_1 > input to D_2. Hence positive output V_{01} of

D_1 is higher than negative output of D_2. Therefore output is positive

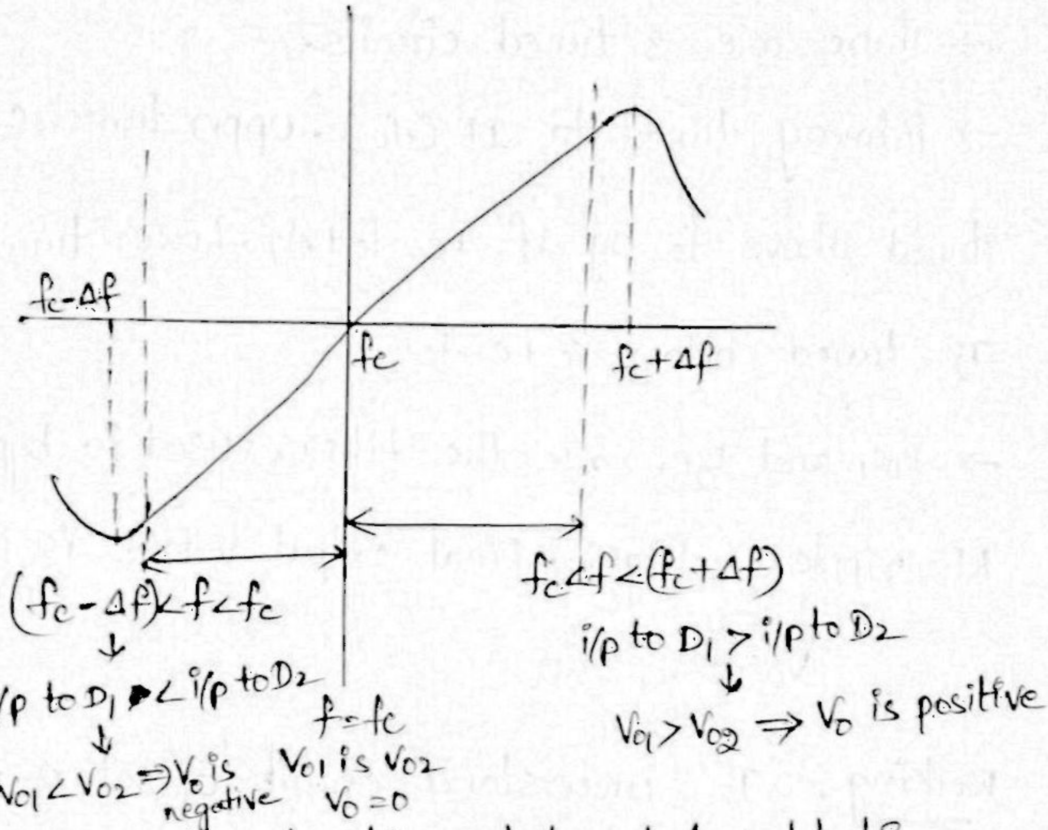


Fig:- characteristics of balanced slope detector

Advantages :-

1. circuit more efficient than simple slope
2. It has better linearity than simple slope

Drawbacks :-

1. Eventhough good linearity it is not enough
2. Amplitude Limiting is not provided
3. circuit difficult to tune . circuit has to tuned

at different frequency f_c, $f_c+\Delta f$, $f_c-\Delta f$

Zero crossing detector :-

→ It detects FM signal by counting the no. of zero crossing of the input waveform.

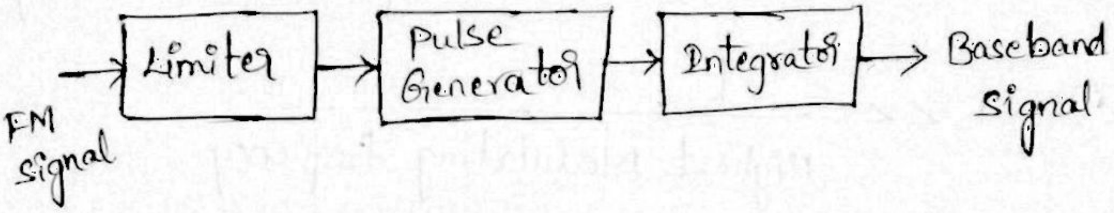
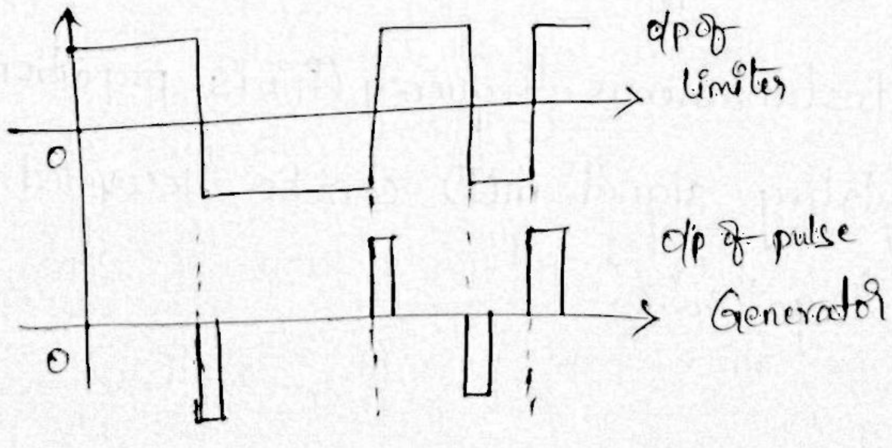


Fig:- Block diagram of zero crossing detector

→ Limiter converts the input FM signal into corresponding square wave

→ pulse Generator produces short pulses at the positive going and as well as negative going edges of the limiter output.



→ Integrator → Averaging of these short pulses over an interval 'T' and gives the output original modulating signal

→ Instataneous frequency of FM wave is

$$f_i = \frac{1}{2\Delta t}$$ Δt → Time difference between two adjacent zero crossing.

(i) $$T << \frac{1}{\text{Highest Modulating frequency}}$$

(ii) $$T >> \frac{1}{\text{carrier frequency } f_c \text{ of FM wave}}$$

$$\Delta t = \frac{T}{n_0}$$ where n_0 = no of zero crossing inside a given time interval T. as choosen above

$$f_i = \frac{1}{2 \cdot \frac{T}{n_0}} = \frac{n_0}{2T}$$

→ Instantaneous frequency (f_i) is proportional to modulating signal m(t) can be recovered by counting "n_0".

Phase Locked Loop (PLL) :-

→ phase Locked Loop is negative feedback system used to track phase and frequency of carrier component of incoming signal

→ It is useful for demodulation of angle-modulated signal, especially when (SNR) signal to Noise Ratio is poor

→ PLL has 3 components

1. Voltage controlled oscillator
2. Multiplier serving as a phase detector
3. Loop filter which is low pass filter

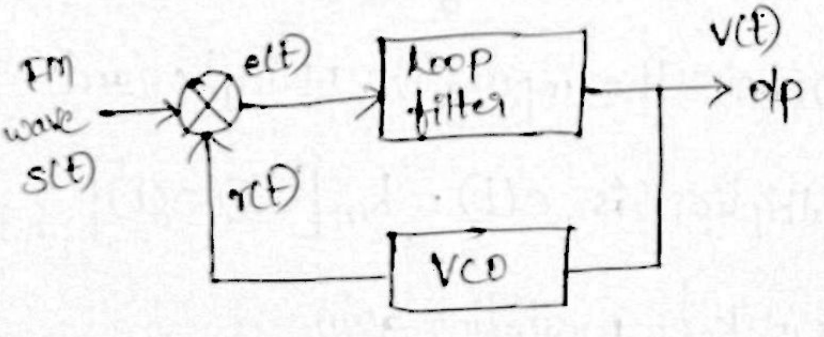


Fig:- Block diagram of PLL

→ Frequency of VCO controlled by external voltage

→ In VCO oscillator frequency varies linearly with its input voltage. output of Multiplier is lowpass filter and then applied to input of VCO.

→ Initially voltage control to VCO is zero, then VCO adjusted frequency of VCO exactly made equal to unmodulated carrier frequency.

→ output of VCO has a phase shift of 90° with respect to unmodulated carrier.

→ suppose input signal to phase Locked Loop is FM wave $s(t) = A_c \cos\left[2\pi f_c t + 2\pi k_f \int_0^t m(t)dt\right]$

→ r(t) denotes the output of VCO and is given

$$r(t) = A_v \cos\left[2\pi f_c t + 2\pi k_v \int_0^t v(t)dt\right]$$

→ s(t) and r(t) are the inputs of Multiplier and output of multiplier is $e(t) = k_m\left[s(t) \cdot r(t)\right]$

where k_m = multiplier gain

$$e(t) = k_m\left\{\left[A_c \cos\left(2\pi f_c t + 2\pi k_f \int_0^t m(t)dt\right)\right] \cdot \left[A_v \cos\left(2\pi f_c t + 2\pi k_v \int_0^t v(t)dt\right)\right]\right.$$

$$e(t) = K_m \left\{ A_c \cos(2\pi f_c t + \phi_1(t)) \cdot A_v \cos(2\pi f_c t + \phi_2(t)) \right\}$$

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

$$\phi_2(t) = 2\pi k_v \int_0^t v(t)\,dt$$

$$\cos a \cos b = \frac{1}{2} \cos(a+b) + \cos(a-b)$$

$$e(t) = \frac{K_m}{2} A_c A_v \left\{ \cos(2\pi f_c t + \phi_1(t) + 2\pi f_c t + \phi_2(t)) + \cos(\cancel{2\pi f_c t} + \phi_1(t) - \cancel{2\pi f_c t} - \phi_2(t)) \right\}$$

$$\therefore \; e(t) = \frac{K_m}{2} A_c A_v \left\{ \cos(4\pi f_c t + \phi_1(t) + \phi_2(t)) + \cos(\phi_1(t) - \phi_2(t)) \right\}$$

→ The output of multiplier $e(t)$ is passed through Low pass filter. 1st term consist of high frequency component and 2nd term consist of Low frequency component.

→ High frequency component removed by LPF

$$e(t) = \frac{1}{2} K_m A_c A_v \cos(\phi_1(t) - \phi_2(t))$$

$$= \frac{1}{2} K_m A_c A_v \cos \phi_e(t) \quad [\because \phi_e(t) \rightarrow \text{phase error}]$$

$$\phi_e(t) = \phi_1(t) - \phi_2(t)$$

$$= \phi_1(t) - 2\pi k_v \int_0^t v(t)\,dt$$

The output of Loop filter is $v(t)$ given by

$$v(t) = \int_{\infty}^{-\infty} e(\tau)\, h(t-\tau)\, d\tau \quad [\because h(\tau) \rightarrow \text{Impulse response of filter}]$$

$$\phi_e(t) = \phi_1(t) - 2\pi k_v \int_0^{t} \int_{-\infty}^{\infty} e(\tau)\, h(t-\tau)\, d\tau \cdot dt$$

differentiating above equation w.r.t to time

$$\frac{d\,\phi_e(t)}{dt} = \frac{d}{dt}\phi_1(t) - 2\pi k_0 \int_{-\infty}^{\infty} \cos \phi_e(t)\, h(t-\tau)\, d\tau$$

$$k_0 = \text{loop parameter} = k_m k_v A_c A_v$$

we already know that $\phi_e(t) = \phi_1(t) - 2\pi k_v \int_0^t v(t)\,dt$

$\phi_e(t)$ = small error so it is neglected the

above eq becomes $\phi_1(t) = 2\pi k_v \int_0^t v(t)\,dt$

$$\Rightarrow \cancel{2\pi} k_f \int_0^t m(t)\,dt = \cancel{2\pi} k_v \int_0^t v(t)\,dt$$

By differentiating we get $k_f(m(t)) = k_v(v(t))$

$$\therefore v(t) = \frac{k_f}{k_v} m(t)$$

→ Thus the output of LPF is proportional to the original modulating signal.

Comparision of FM and AM :-

	FM	AM
1.	Amplitude of FM wave is constant. It is independent of the modulation Index	Amplitude of AM wave will change with the modulating voltage
2.	It is possible to operate several transmitters on same frequency	Not possible to operate more channels on the same frequency
3.	used for short distance communication	used for long distance communication
4.	Equation of Fm is $s(t) = A_c \cos 2\pi f_c t + \beta \sin 2\pi f_m t$	equation for Am is $A_c \cos 2\pi f_c t + \mu \cos 2\pi f_m t$
5.	Bandwidth is large, Hence wide channel is required.	Bandwidth is much less than FM.
6.	All the transmitted power is useful	carrier power and one sideband power are useless

7. Bandwidth $2(\Delta f + f_m)$. The BW depends on modulation Index	$BW = 2f_m$. It is not dependent on the modulation Index
8 FM transmission and reception equipment are more complex	AM equipments are less complex
9 FM is applicable for wireless communication	AM is applicable for wired communication.

2. DSB & SSB MODULATION

Introduction

when carrier is Amplitude modulated by a single sine wave, the resulting signal consist of 3 frequencies i.e original carrier and 2 sidebands ($f_c \pm f_m$). The system is commonly known as Double sideband full carrier system (DSBFC)

→ The carrier wave is completely independent of information carrying signal and base band signal m(t) i.e the transmission of carrier wave in DSBFC is a waste of power i.e 66% power will be occupied.

→ It is the main drawback of DSBFC. To overcome this drawback we may suppress the carrier component from the modulated wave

→ when carrier is removed, the remaining signal contains simply upper and Lower sidebands such a signal called Double side band Suppressed carrier DSBSC (or) DSB signal.

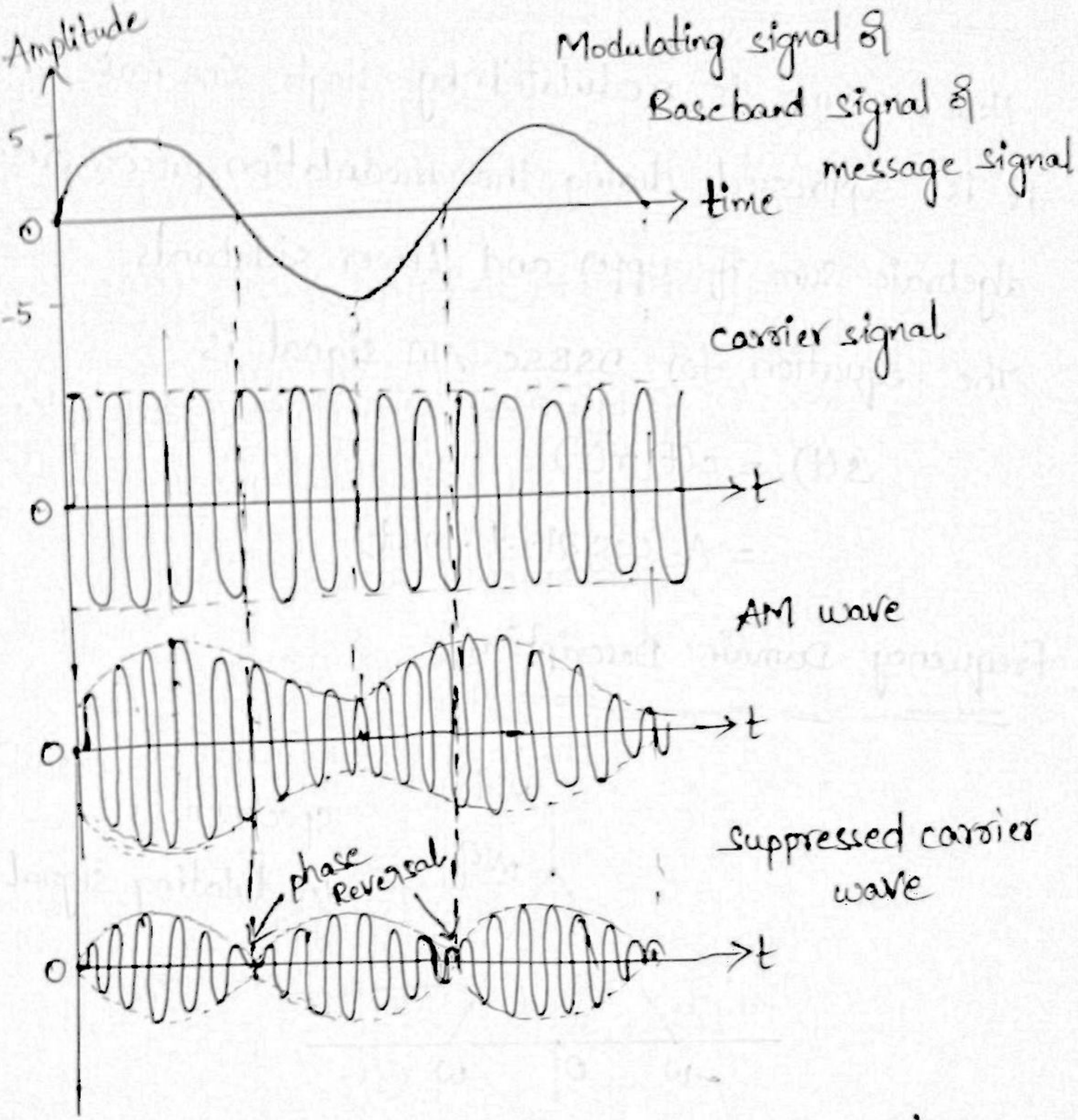
→ with this signal No power is wasted on the carrier and the saved power can be put into the sidebands for stronger signals over Long distance

→ In DSBFC the standard AM wave equation is $s(t) = (1 + k_a m(t)) \cos 2\pi f_c t \cdot A_c$

k_a = constant called Amplitude sensitivity.

→ After the supression of carrier component the DSBSC is $s(t) = c(t) \cdot m(t)$

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



→ The suppressed carrier wave undergoes a phase reversal whenever the baseband signal crores '0' line which was observed in above.

→ The DSBFC wave different than DSBSC

Time Domain Description of DSBSC :-

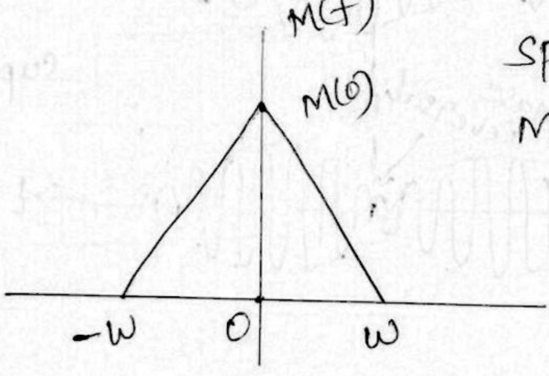
Here, carrier is modulated by single sine wave and it is suppressed during the modulation process, resulting algebraic sum of upper and Lower sidebands

The equation for DSBSC AM signal is

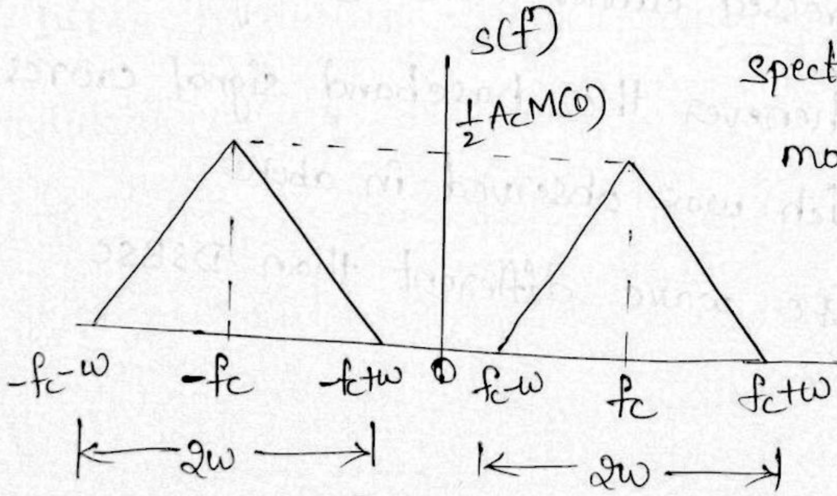
$$s(t) = c(t)\, m(t)$$

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

Frequency Domain Description :-



Spectrum of Modulating signal



Spectrum of DSBSC modulated wave

The suppression of the carrier from the modulated wave i.e $s(t) = c(t)\, m(t)$

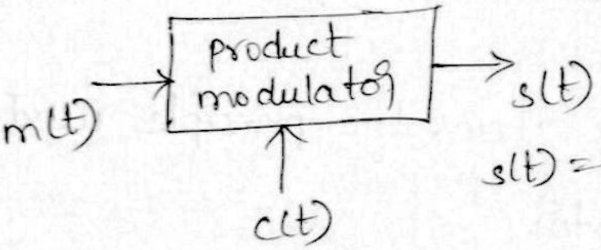
Applying Fourier Transform to the above eq

$$S(f) = \frac{1}{2} A_c \left[M(f - f_c) + (f + f_c) \right]$$

where $S(f) \rightarrow$ Fourier Transform of modulated wave $s(t)$

$M(f) \rightarrow$ Fourier Transform of modulating signal. $m(t)$

Generation of DSBSC waves :-



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

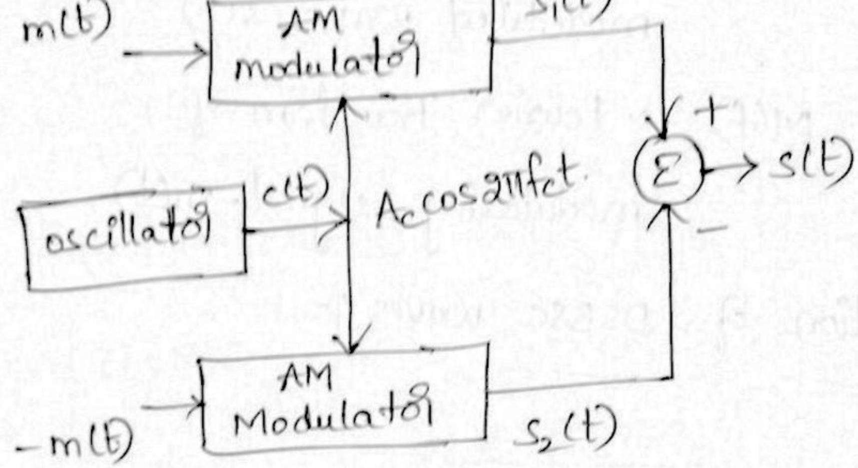
→ DSBSC modulated wave consist of product of modulating signal and carrier signal

→ The desired output $s(t)$ can be achieved by a device called a product modulator (or) Balanced modulator.

→ Balanced modulator suppress the carrier from the AM signal

→ For the generation of DSBSC we are using 2 Techniques they are 1. Balanced modulator
2. Ring modulator (or) Balanced modulator using diode

1. Balanced modulator :-



→ Block diagram shows the principle used in Balanced modulator.

→ A device having non-linear resistance such as diode, JFET or transistor can be used in Balanced modulator to generate AM signal with suppressed carrier.

→ From the diagram two modulators are identical, except the sign reversal of the message signal

(4)

applied to the input. The outputs of 2 modulators are

$$s_1(t) = A_c(1+k_a m(t)) \cos 2\pi f_c t$$

$$s_2(t) = A_c(1-k_a m(t)) \cos 2\pi f_c t$$

$$s(t) = s_1(t) - s_2(t)$$

$$= \left(A_c[1+k_a m(t)] \cos 2\pi f_c t\right) - \left(A_c[1-k_a m(t)] \cos 2\pi f_c t\right)$$

$$= \cancel{A_c \cos 2\pi f_c t} + k_a m(t) A_c \cos 2\pi f_c t - \cancel{A_c \cos 2\pi f_c t} + A_c k_a m(t) \cos 2\pi f_c t.$$

$$= 2 k_a A_c \cos 2\pi f_c t \cdot m(t)$$

where $2k_a$ = scaling factor

$m(t)$ = Modulating signal

$A_c \cos 2\pi f_c t$ = carrier signal

Ring Modulator :-

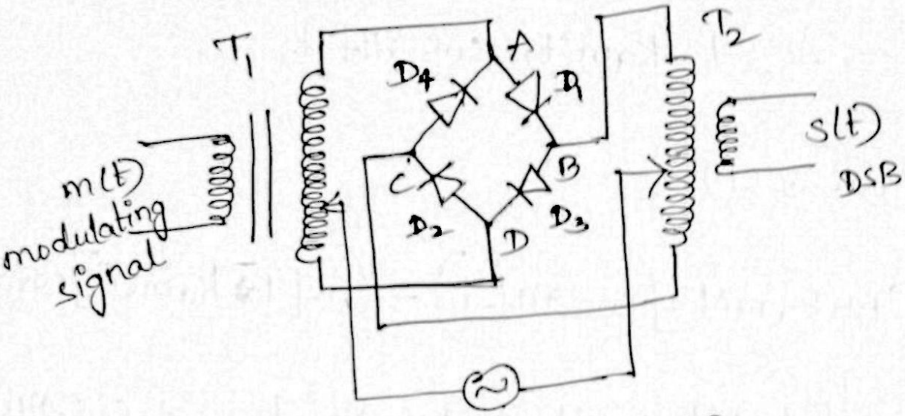


Fig:- Lattice type Balanced Modulator

→ Lattic type Balanced modulator consist of input transformer T_1, output transformer T_2 and four diodes connected in a bridge circuit.

→ Modulating signal applied to the input Transformer T_1.

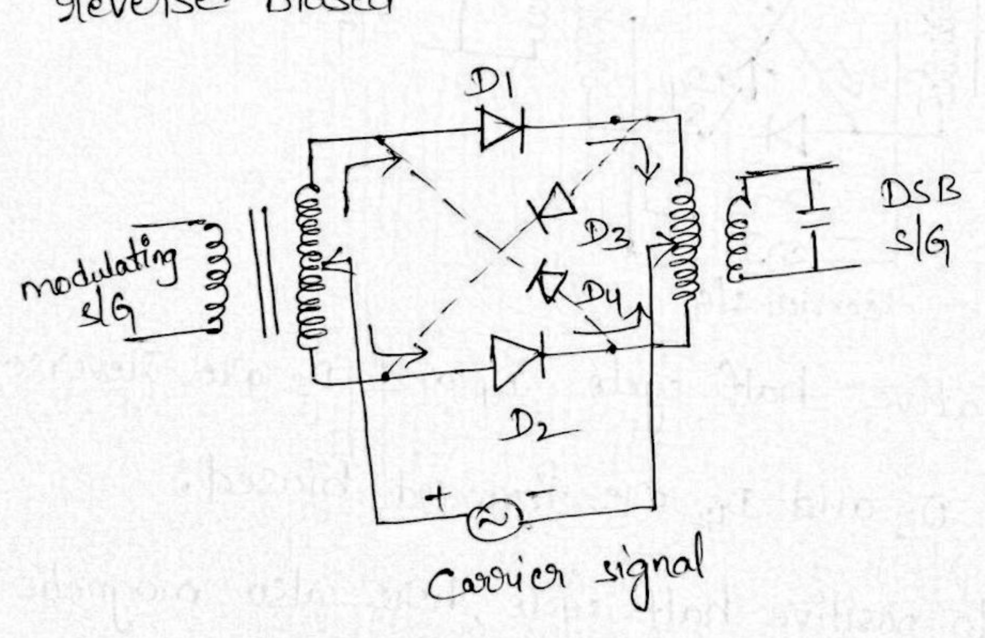
→ carrier signal is applied to the center taps of the input and output transformers. Carrier signal is higher in frequency and amplitude than modulating signal.

→ Diodes connected in the Bridge acts like switch and switching is controlled by the carrier signal.

Positive half cycle of carrier s/G :-

→ In order to understand working of the circuit assume modulating signal is zero

→ positive half cycle of the carrier signal diodes D_1 and D_2 are forward biased, D_3 and D_4 are reverse biased

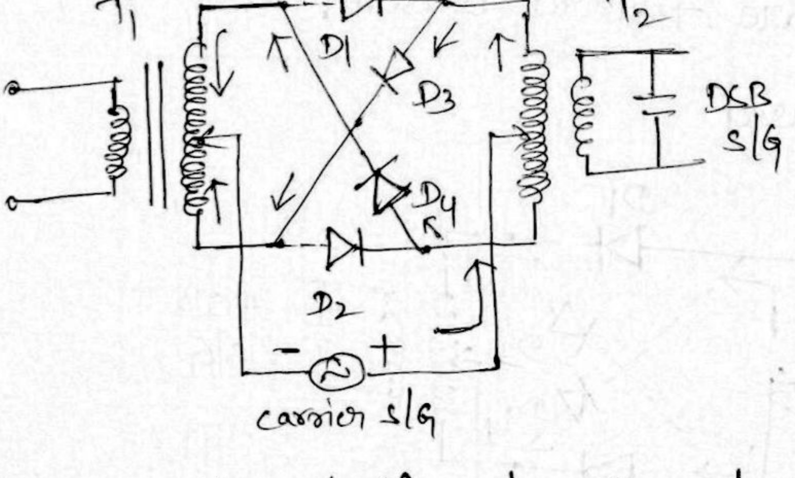


→ current divides equally in the upper and Lower of Primary winding of T_2

→ current in upper part of winding produces a magnetic field is equal and opposite to the magnetic field produced by the current in lower half of secondary

→ As a result the magnetic fields are equal and opposite they cancel each other, producing no output at the secondary of T_2. Thus, the carrier is suppressed.

<u>Negative half cycle of carrier signal</u> :-



→ In Negative half cycle D_1 and D_2 are reverse biased and D_3 and D_4 are forward biased.

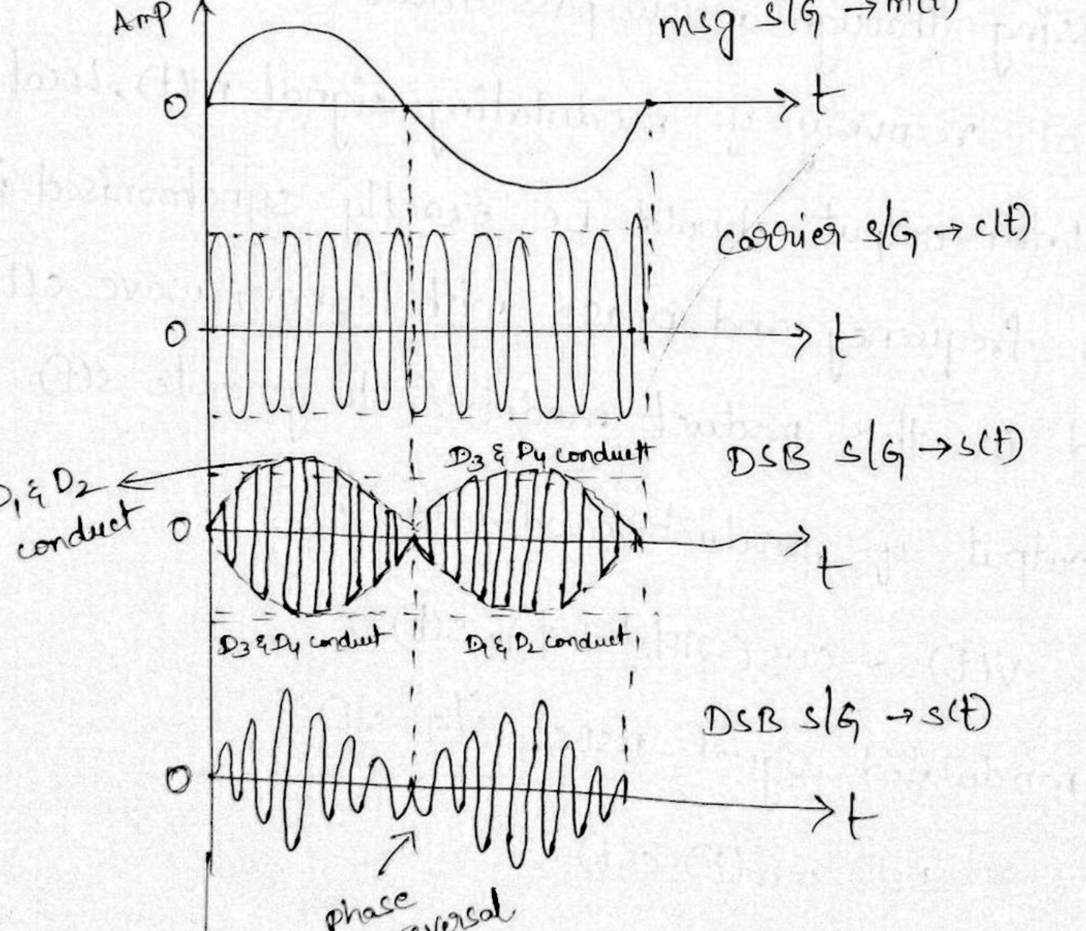
→ similarly to positive half cycle, here also magnetic fields are equal and opposite cancel each other.

with modulating signal :-

→ Assume low frequency sine wave is applied to the primary of T_1.

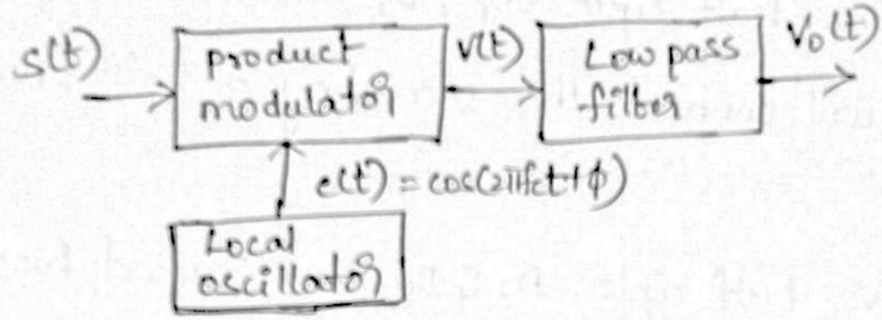
→ In positive half cycle D_1 & D_2 are forward biased and they will connect the secondary of T_1 to primary of T_2.

→ In Negative half cycle D_3 & D_4 are forward biased and they will connect the secondary of T_1 to primary of T_2 with reverse connection. This results 180° phase shift in modulating signal.



Detection of DSBSC :-

Coherent Detection of DSBSC modulated wave :-



→ The modulating signal m(t) recovered from a DSBSC wave i.e s(t) by first multiplying with a locally generated sinusoidal wave and then passing through a low pass filter

→ For recovery of modulating signal m(t), Local oscillator output should be exactly synchronised in both frequency and phase with carrier wave c(t) used in the product modulator to generate s(t).

output of product modulator v(t) is

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

modulated signal DSBSC s/G s(t) is

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

Substituting s(t) value in v(t) we get

$$v(t) = \cos(2\pi f_c t + \phi) \cdot m(t) \cdot c(t)$$

where $c(t)$ = carrier s/G = $A_c \cos 2\pi f_c t$

$m(t)$ = message s/G

$$\therefore \quad v(t) = \cos(2\pi f_c t + \phi) \cdot m(t) \cdot A_c \cos 2\pi f_c t$$

$$= A_c \cdot m(t) \cdot \cos\underbrace{(2\pi f_c t + \phi)}_{A} \cdot \cos \underbrace{2\pi f_c t}_{B}$$

$$\left[\because \cos A \cdot \cos B = \frac{1}{2}\left[\cos(A+B) + \cos(A-B)\right]\right]$$

$$= A_c\, m(t) \frac{1}{2}\left[\cos(2\pi f_c t + \phi + 2\pi f_c t) + \cos(\cancel{2\pi f_c t} + \phi - \cancel{2\pi f_c t})\right]$$

$$= \frac{A_c\, m(t)}{2}\left[\cos(4\pi f_c t + \phi) + \cos(\phi)\right]$$

$$= \underbrace{\frac{A_c\, m(t)}{2} \cos\phi}_{\text{Scaled version of msg s/G}} + \underbrace{\frac{A_c\, m(t)}{2} \cos(4\pi f_c t + \phi)}_{\text{Unwanted term}}$$

ϕ represents phase ~~error~~ angle. phase error is constant

As long as phase ~~error~~ angle is constant the detector output provides ~~its~~ undistorted modulating s/G

→ The output of product modulator consist of 2 terms = Scaled version of msg s/G + Unwanted term

→ The Unwanted term can be removed with the help of low pass filter.

→ The output of Low pass filter is $V_0(t)$

$$V_0(t) = \frac{1}{2} A_c \cos\phi \cdot m(t).$$

→ ϕ is phase angle, it is constant the demodulation signal is proportional to m(t)

→ $\phi = 0$ the amplitude of signal will be maximum

→ $\phi = \pm\frac{\pi}{2}$ the amplitude of signal will be zero This is called quadrature null effect

→ As long as phase angle will be constant then we get the original message signal.

<u>Single Tone DSBSC modulation</u> :-

we know that $s(t) = m(t) \cdot c(t)$

$s(t)$ → DSBSC modulated wave

$m(t)$ → msg s/G ; $c(t)$ = carrier s/G

$$m(t) = A_m \cos 2\pi f_m t$$

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

$$s(t) = A_m \cos 2\pi f_m t \cdot A_c \cos 2\pi f_c t$$

$$= A_m A_c \cos \underbrace{2\pi f_m t}_{A} \cdot \cos \underbrace{2\pi f_c t}_{B}$$

$$\left[\because \cos A \cos B = \tfrac{1}{2}\left[\cos(A+B) + \cos(A-B)\right]\right]$$

$$= A_m A_c \tfrac{1}{2}\left[\cos(2\pi f_m t + 2\pi f_c t) + \cos(2\pi f_c t - 2\pi f_m t)\right]$$

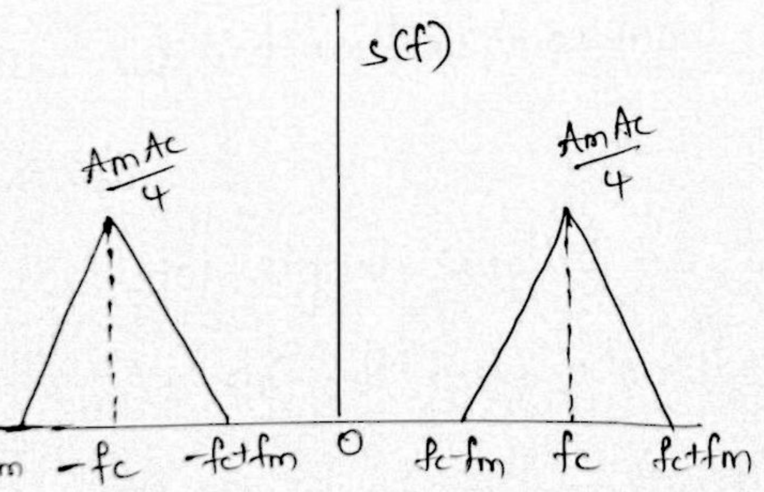
$$= \frac{A_m A_c}{2}\left[\cos 2\pi(f_m + f_c)t + \cos 2\pi(f_c - f_m)t\right]$$

$$s(t) = \frac{1}{2} A_m A_c \left(\cos 2\pi(f_m + f_c)t\right) + \frac{A_m A_c}{2} \cos 2\pi(f_c - f_m)t$$

Fourier Transform of DSBSC modulated wave is

$$S(f) = \frac{1}{4} A_m A_c \left[\delta(f - (f_m + f_c)) + \delta(f + (f_c + f_m))\right] +$$

$$\frac{A_m A_c}{4}\left[\delta(f - (f_c - f_m)) + \delta(f + (f_c - f_m))\right]$$



power calculations of DSBSC :-

$$P = \frac{V_{rms}^2}{R} \quad \text{where} \quad V_{rms} = \frac{V_m}{\sqrt{2}}$$

$$V_m = \frac{A_m A_c}{2}$$

$$\therefore \; P_{USB} = P_{LSB} = \frac{V_{rms}^2}{R} = \frac{\left(\frac{V_m}{\sqrt{2}}\right)^2}{R} = \frac{\left(\frac{A_m A_c}{2\sqrt{2}}\right)^2}{R}$$

$$= \frac{A_m^2 A_c^2}{4 \times (\sqrt{2})^2} \times \frac{1}{R} = \frac{A_m^2 A_c^2}{4 \times 2} \times \frac{1}{R}$$

$$P_{USB} = P_{LSB} = \frac{A_m^2 A_c^2}{8R}$$

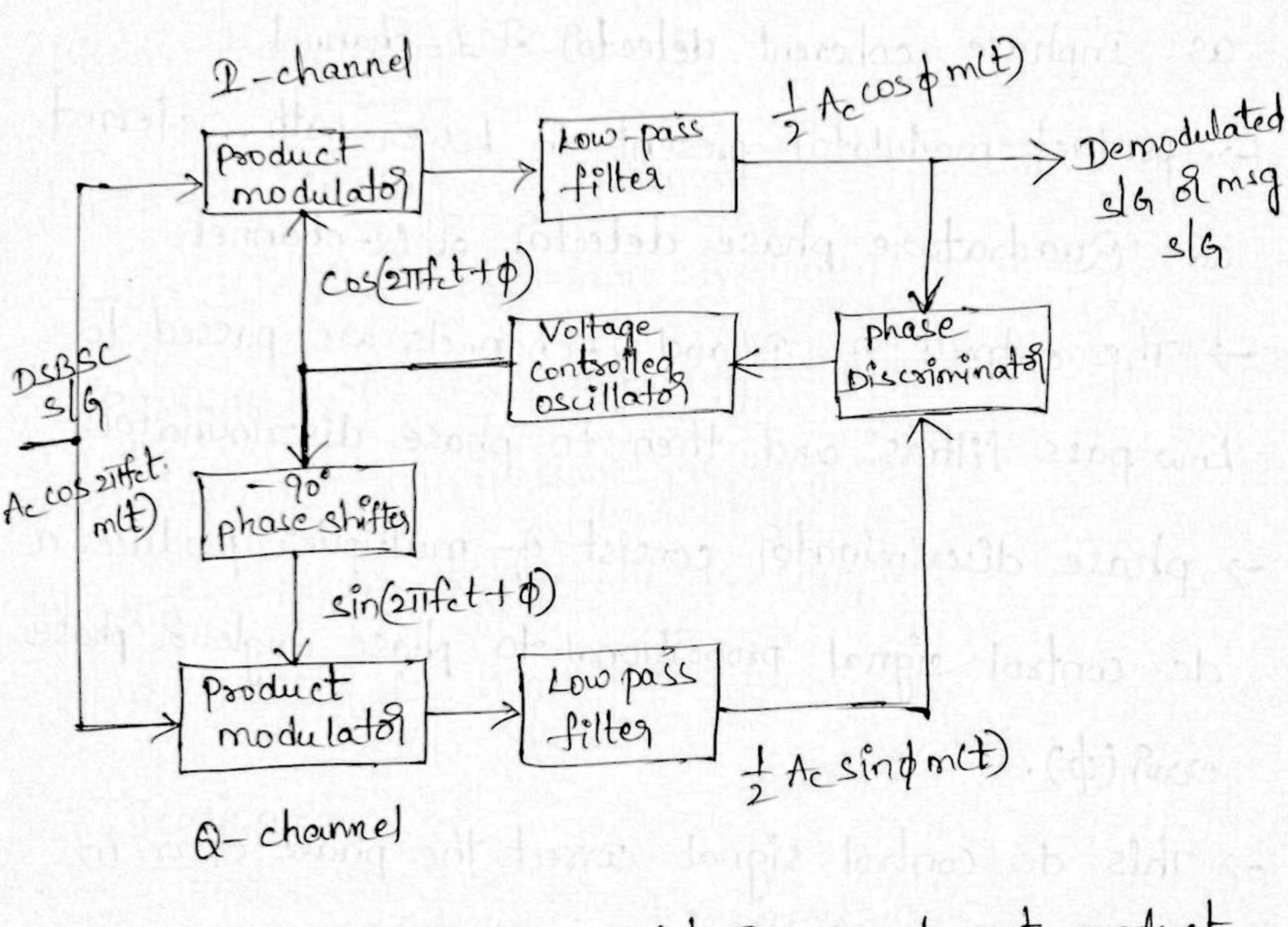
$$P_{Total} = P_{USB} + P_{LSB}$$

$$= \frac{A_m^2 A_c^2}{8R} + \frac{A_m^2 A_c^2}{8R} = \frac{\cancel{2} A_m^2 A_c^2}{\underset{4}{\cancel{8}} R}$$

$$P_{Total} = \frac{A_m^2 A_c^2}{4R}$$

Costas Loop :-

→ Costas Loop is a method of obtaining a practical synchronous receiving system.



→ In this, Receiver consist of 2 coherent product modulators supplied with same input signal i.e DSBSC signal

→ Local oscillator signal supplied to the product modulators are 90° out of phase. The frequency of

~~Local os~~

Local oscillator is adjusted to be same as carrier frequency fc

→ product modulator present in upper path referred as Inphase coherent detector or I-channel

→ product modulator present in Lower path referred as Quadrature phase detector or Q-channel

→ The outputs of I and Q-channels are passed to Low pass filters and then to phase discriminator.

→ phase discriminator consist of multiplier produces a dc control signal proportional to phase angle or phase error (ϕ).

→ This dc control signal correct the phase error in local oscillator

→ Costas Loop is the Negative feedback system designed to maintain the Local oscillator synchronous with carrier wave.

Advantages of DSBSC :-

1. Low power consumption compared to DSBFC
2. simple and efficient modulation system.

Disadvantages of DSBSC :-

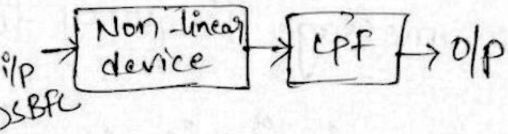
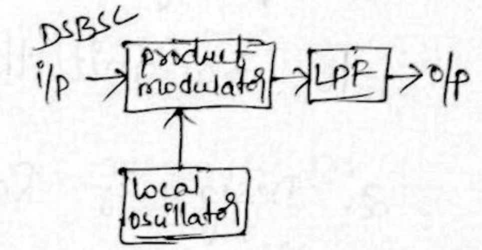
1. carrier is suppressed remaining upper side band and Lower side band has same frequency i.e message S/G frequency twice but only one side band is required to transmit the message frequency fm
2. Design of Receiver circuit is complex
3. Bandwidth is larger i.e 2fm

Application :-

→ used in point to point communication

→ used in Radio communication

Comparision between Synchronous and envelop detection

Parameter	Envelope	Synchronous
1. used in	In only DSBFC	DSBSC, SSB & VSB
2 use of Synchronous carrier	Locally generated Synchronous carrier not is used	synchronous carrier is used
3. complexity	Low	High
4. Block diagram		
5. Principle of operation	i/p s/G is rectified using diode or transistor & passes through LPF	i/p s/G is multiplied with Local oscillator i-e carrier & passes through LPF
6. Errors	Envelop distortion & diagnal clipping	Frequency and phase errors.

Comparision between coherent & Non-coherent Receivers

Coherent	Non-coherent
1. It Requires carrier signal to be generated at the Receiver.	1. Doesn't require carrier signal
2. used in DSBSC	2. used in DSBFC
3. m(t) is recovered from s(t) by multiplying with locally generated carrier and passes to LPF	3. m(t) is recovered by rectifying the incoming s/G to remove half of the envelop.
4. uses Linear Receiver	4. Uses Non-Linear Receiver
5. High cost	5. Low cost
6. Noise performance is better than Non-linear	6. Noise performance is very poor than coherent detector.

SSB Modulation :-

→ carrier signal doesnot contain any information used to just pass message signal for long distance.

→ The information carried by sidebands.

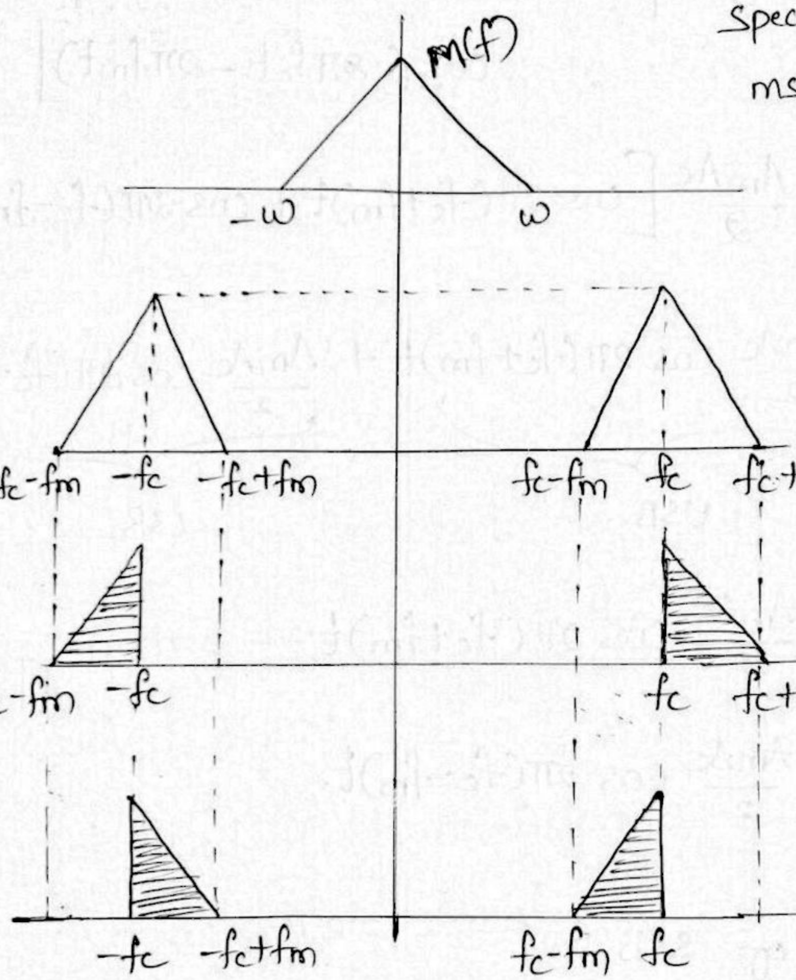
→ In DSB two sidebands carry same information but no need of two sidebands to carry same information so one sideband is suppressed

SSB modulation defination :-

→ In DSBSC the basic information is transmitted through 2 sidebands but two side bands contains the same information in order to convey the information one sideband is suppressed and one sideband is transmitted. The Resulting signal is called single side band signal (SSB signal).

Frequency Domain Description of SSB :-

The frequency domain description displays which sideband is transmitted.



Spectrum of msg S/G

Spectrum of DSBSC

Spectrum of SSB with upper sideband

Spectrum of SSB with Lower sideband

Time Domain Description of SSB :-

→ SSB S/G generated by passing DSBSC through a band pass filter

$$s(t)_{DSBSC} = m(t) \cdot A_c \cos 2\pi f_c t$$

where $m(t) = A_m \cos 2\pi f_m t$

$$s(t)_{DSBSC} = A_m \cos 2\pi f_m t \cdot A_c \cos 2\pi f_c t$$

$$\underset{DSBSC}{s(t)} = A_m A_c \cos \underbrace{2\pi f_c t}_{A} \cos \underbrace{2\pi f_m t}_{B}$$

$$\left[\because 2\cos A \cos B = \cos(A+B) + \cos(A-B)\right]$$

$$= \frac{A_m A_c}{2}\left[\cos(2\pi f_c t + 2\pi f_m t) + \cos(2\pi f_c t - 2\pi f_m t)\right]$$

$$= \frac{A_m A_c}{2}\left[\cos 2\pi(f_c+f_m)t + \cos 2\pi(f_c-f_m)t\right]$$

$$= \underbrace{\frac{A_m A_c}{2}\cos 2\pi(f_c+f_m)t}_{USB} + \underbrace{\frac{A_m A_c}{2}\cos 2\pi(f_c-f_m)t}_{LSB}$$

$$\underset{SSB-SC}{s(t)} = \frac{A_m A_c}{2}\cos 2\pi(f_c+f_m)t$$

$$\underset{SSB-SC}{s(t)} = \frac{A_m A_c}{2}\cos 2\pi(f_c-f_m)t.$$

Advantages of SSB :-

1. Due to suppression of carrier and onesideband power is saved.
2. The spectrum space occupied by the SSB signal is f_m, which is only half that of AM and DSB signals.

3. Reduced the interference of noise due to the reduced Bandwidth.

4. Fading doesnot occur in SSB Transmission. Fading means that a signal alternately increase and decrease in strength when it is received by Receiver

Disadvantages of SSB :-

1. Generation and reception of SSB signal is complex.

2. carrier is absent, the SSB Transmitter and Receiver need to have an excellent frequency stability. Little change in frequency hampers the quality of transmitted and Received signal.

3. SSB not used for transmission of good quality of signal such as music signal.

Applications of SSB :-

1. SSB used for transmission of speech signal.

2. SSB used in Application which Bandwidth requirements are low.

3. SSB used to save power in Application i.e in mobile system.

4. point to point communications, television, telemetry, military communications, Radio navigation are the greatest users of SSB in one form or another.

Generation of SSB modulated wave :-

SSB modulated waves can be generated by using 2 methods
1. Frequency Discrimination method
2. phase Discrimination method

Frequency Discrimination method for Generating SSB :-

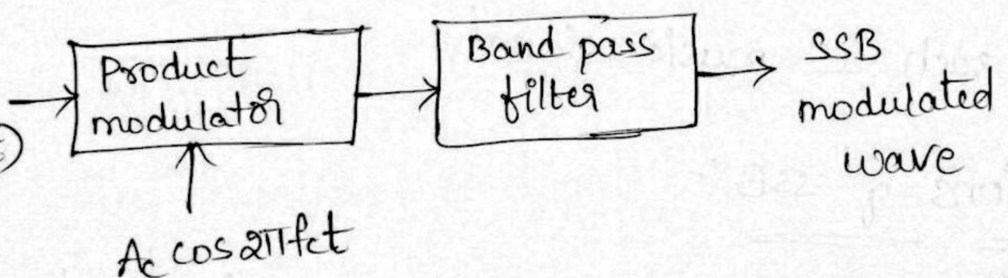


Fig:- Block diagram of freq discrimination method

→ From the block diagram product modulator or balanced modulator, band pass filter is used

→ Band pass filter is used to remove unwanted sidebands. filter must have a flat passband and extremely high attenuation outside the bandpass.

→ In order to have above response the Q of tuned circuits must be very high

→ The required Q factor increases as the difference between modulating frequency and carrier frequency increases.

→ carrier frequency usually same as Transmitter frequency.

→ It is necessary to generate an SSB modulated wave occupying frequency band higher than message signal, it becomes very difficult to design an filter that will pass the desired sideband and reject other using single stage modulation. In such cases double stage modulation is used.

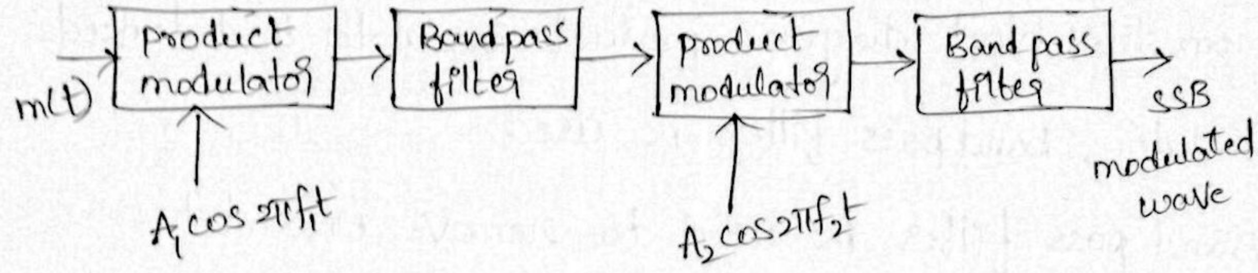


Fig:- Block diagram of two stage SSB modulation

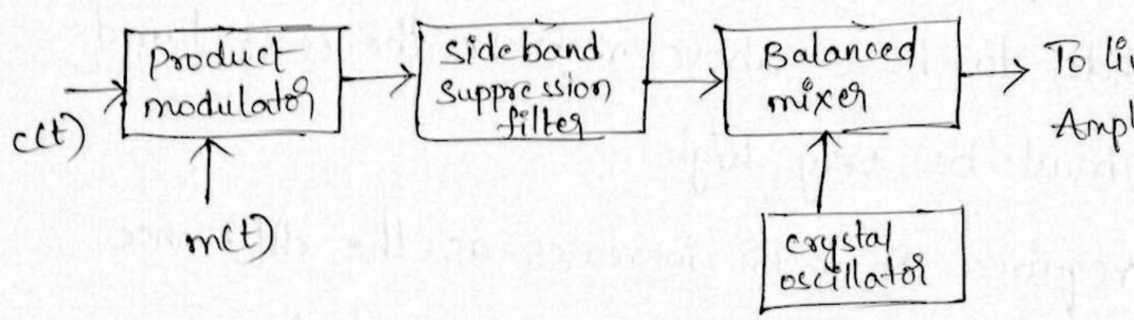


Fig:- Filter method of sideband suppression

Advantages of filter method :-

1. Filter method gives sideband suppression upto 50dB which is quite adequate.
2. Bandwidth is sufficiently flat and wide.

Disadvantages of filter method :-

1. They are bulky
2. At lower audio frequencies expensive filters are required

3. As modulation takes place at lower carrier frequency repeated mixing is required with extremely stable oscillators to generate SSB at high radio frequencies.

phase Discrimination method for Generating SSB modulated wave :-

phase shift method of SSB generation uses phase-shift technique that causes one of the sidebands to be cancelled out.

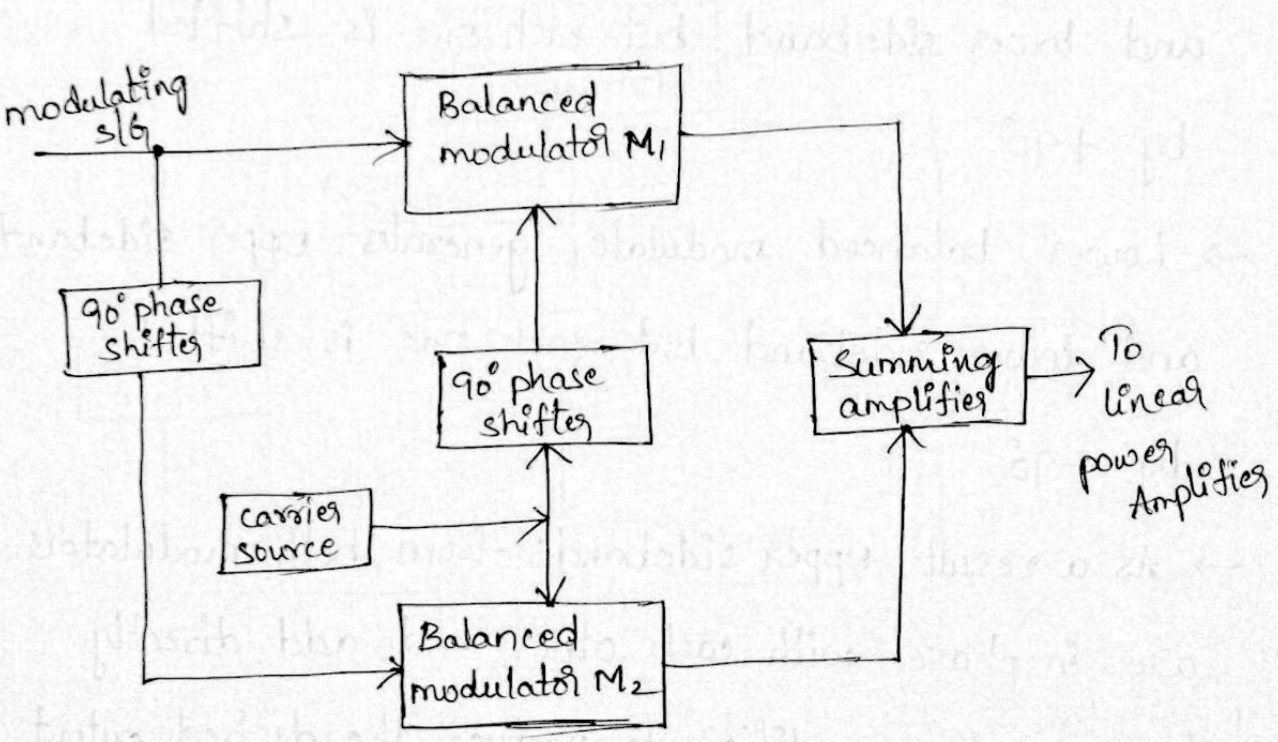


Fig:- phase shift method of lower side band of SSB

→ The above block diagram consist of 2 balanced modulator and 2 phase shifting networks.

→ The carrier signal shifted by 90° and modulating signal ~~shifted by~~ applied to the upper balanced modulator M_1 whereas

→ The carrier signal and modulating signal shifted by 90° applied to the Lower balanced modulator M_2.

→ Both modulator outputs consist of only of sidebands.

→ upper balanced modulator generates upper sideband and lower sideband but each one is shifted by +90°

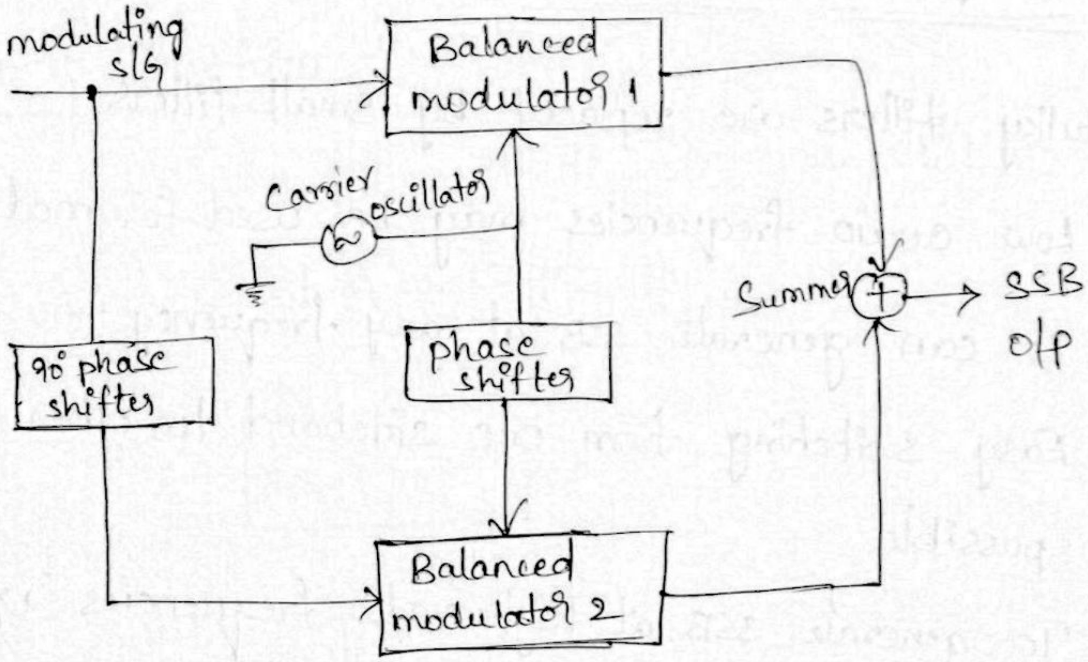
→ Lower balanced modulator generates upper sideband and lower sideband but each one is shifted by −90°

→ As a result upper sidebands from both modulators are in phase with each other and add directly in summing amplifier to produce the desired output sideband signal

→ The lower sidebands from both modulators are shifted so that they are 180° out of phase with one another and cancel when added

→ Therefore the outputs of the two balanced modulators are added algebracially we get upper sideband signal and lower sideband is suppressed.

→ To suppress upper sideband the phase shifter connection are slightly different i.e



→ here modulating & carrier signals are applied directly to the upper balanced modulator & these signals phase shifted by 90° and then applied to lower balanced modulator

At modulator 1

$$m(t) = A_m \cos 2\pi f_m t$$

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

At modulator 2

$$\hat{m}(t) = A_m \sin 2\pi f_m t$$

$$c(t) = A_c \sin 2\pi f_c t$$

$$s(t) = \text{o/p of modulator 1} + \text{o/p of modulator 2}$$

$$= m(t) \cdot A_c \cos 2\pi f_c t \pm \hat{m}(t) A_c \sin 2\pi f_c t$$

Advantages of phase shift method :-

1. Bulky filters are replaced by small filters.
2. Low audio frequencies may be used for modulation
3. It can generate SSB at any frequency
4. Easy switching from one sideband to other sideband is possible
5. To generate SSB at high radio frequencies up conversion and hence repeatitive mixing is not necessary.

Disadvantages :-

1. Requires complex AF phase-shift network to work for large frequency range
2. output of 2 balanced modulators must be exactly same otherwise cancellation will be incomplete
3. If phase shifter provides phase change other than 90° at any audio frequency that paticular frequency will not be completely removed from unwanted sideband.

Hilbert Transform :-

→ circuit used to provide the $-\pi/2$ phase shift to the incoming signal or message signal

$$m(t) \longrightarrow \boxed{\text{Hilbert Transform } \left(-\frac{\pi}{2}\right)} \longrightarrow \hat{m}(t)$$

$$m(t) = A_m \cos 2\pi f_m t$$

$$\hat{m}(t) = A_m \sin 2\pi f_m t$$

→ Need of this transform is we have to maintain the sufficient transmitting signal in the transmitting end.

→ mostly we used this transformation in communication purpose.

Single Tone SSB modulation :-

→ Hilbert Transform is bit complex and it is difficult to sketch waveforms of SSB modulated waves. To make it simpler single tone modulation is used

consider modulating signal

$$m(t) = A_m \cos 2\pi f_m t$$

Hilbert transform is means changing phase of the input signal by 90° without changing its amplitude. so

$$\hat{m}(t) = \left(A_m \cos 2\pi f_m t - \frac{\pi}{2}\right) = A_m \sin 2\pi f_m t$$

From time domain discription USB of SSB modulated wave is

$$s(t)_{SSB\text{-}SC} = \frac{A_m A_c}{2} \cos 2\pi (f_c + f_m) t$$

$$S(f) = \frac{A_m A_c}{4}\left[\delta(f-(f_c+f_m)) + \delta(f+(f_c+f_m))\right]$$

From time domain discription LSB of SSB modulated wave

(18)

is $\underset{SSB-SC}{s(t)} = \frac{A_m A_c}{2} \cos 2\pi (f_c - f_m) t$

$$s(f) = \frac{A_m A_c}{4} \left[\delta(f - (f_c - f_m)) + \delta(f + (f_c - f_m)) \right]$$

for USB $\underset{SSB-SC}{s(t)} = \frac{A_m A_c}{2} \cos 2\pi (f_c + f_m) t$

$$\left[\because \cos(a+b) = \cos a \cos b - \sin a \sin b \right]$$

$$s(t) = \frac{A_m A_c}{2} \left[\cos 2\pi f_c t \cdot \cos 2\pi f_m t - \sin 2\pi f_c t \cdot \sin 2\pi f_m t \right]$$

Ily LSB $\underset{SSB-SC}{s(t)} = \frac{A_m A_c}{2} \cos 2\pi (f_c - f_m) t$

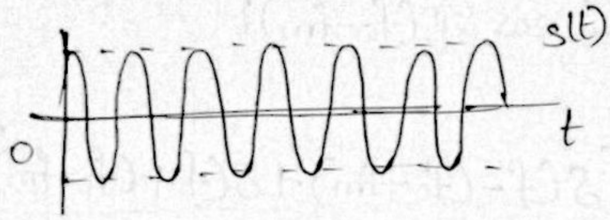
$$= \frac{A_m A_c}{2} \left[\cos 2\pi f_c t \cdot \cos 2\pi f_m t + \sin 2\pi f_c t \cdot \sin 2\pi f_m t \right]$$

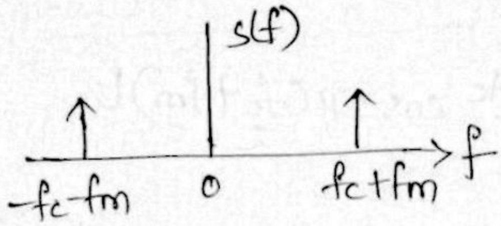
From both USB & LSB eq we can written as

$$s(t) = \frac{A_m A_c}{2} \left[\cos 2\pi f_c t \cdot \cos 2\pi f_m t \pm \sin 2\pi f_c t \cdot \sin 2\pi f_m t \right]$$

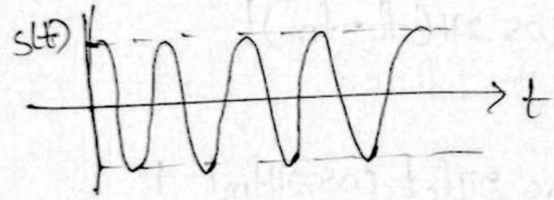
+ Indicates USB

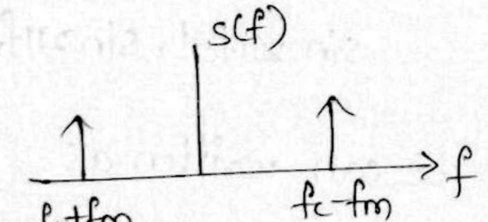
− Indicates LSB





Spectrum of SSB with lower sideband suppressed



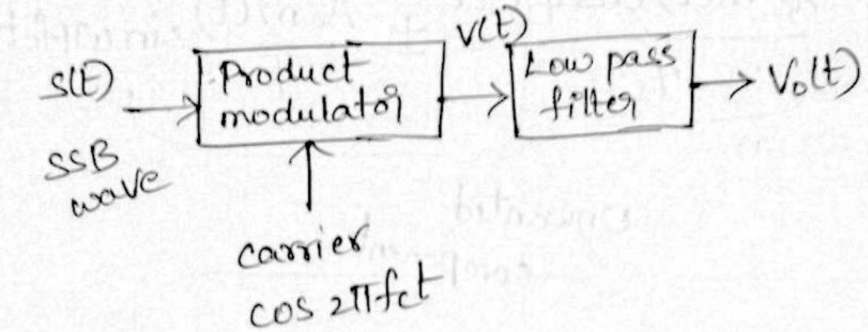


spectrum of SSB with upper sideband supressed.

Demodulation of SSB :-

→ In this we will see how to recover the message signal i.e coherent detection method.

Coherent detection of SSB :-



→ Consider a SSB wave either uppersideband or Lower sideband is applied to the product modulator

→ the output of product modulator is

$$v(t) = s(t) \cdot \cos 2\pi f_c t \quad \text{—①}$$

where $s(t) = \frac{A_c}{2}\left[m(t)\cos 2\pi f_c t \pm \hat{m}(t)\sin 2\pi f_c t\right]$ —②

Sub eq ② in eq ① we get

$$v(t) = \frac{A_c}{2}\left[m(t)\cos 2\pi f_c t \pm \hat{m}(t)\sin 2\pi f_c t\right]\cos 2\pi f_c t$$

$$= \frac{A_c}{2} m(t)\cos^2 2\pi f_c t \pm \frac{A_c}{2}\hat{m}(t)\cos 2\pi f_c t \cdot \sin 2\pi f_c t$$

$$\left[\because \cos^2 A = \frac{1+\cos 2A}{2}, \ \sin 2A = 2\sin A\cos A\right]$$

$$= \frac{A_c}{2} m(t)\left(\frac{1+\cos 2(2\pi f_c t)}{2}\right) \pm \frac{A_c}{2}\hat{m}(t)\cdot\frac{1}{2}\cdot \sin 2(2\pi f_c t)$$

$$= \frac{A_c}{4} m(t)(1+\cos 4\pi f_c t) \pm \frac{A_c}{4} \hat{m}(t) \sin 4\pi f_c t$$

$$v(t) = \underbrace{\frac{A_c\, m(t)}{4}}_{\substack{\text{Desired} \\ \text{s/G}}} + \underbrace{\frac{A_c\, m(t) \cos 4\pi f_c t}{4} \pm \frac{A_c \hat{m}(t)}{4} \sin 4\pi f_c t}_{\substack{\text{Unwanted} \\ \text{component}}}$$

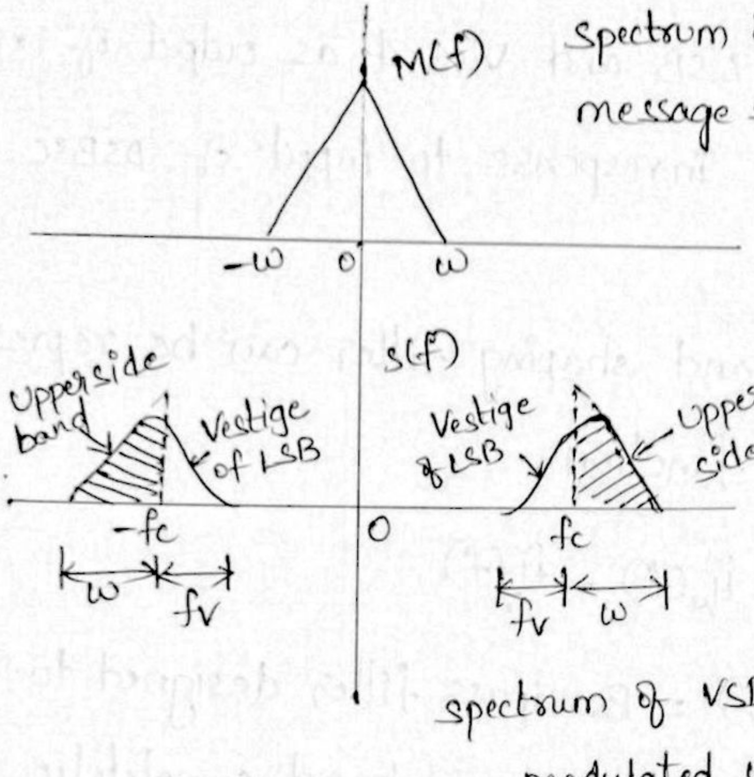
→ Unwanted component can be removed by filtering the signal $v(t)$.

Vestigial sideband modulation :- VSB

→ In SSB modulation when the message signal contains significant components at extremely low frequencies upper and lower sidebands very close to the carrier frequency and it is difficult to isolate one sideband. To overcome this difficulty we are using Vestigial sideband modulation (VSB)

→ In this VSB technique one sideband is passed almost completely and also trace or vestige of the other sideband is retained.

Frequency Domain Description of VSB :-



→ In this technique one side band is passed almost completely and also vestigal of other side band is retained.

→ For transmission of USB it occupies the unwanted sideband i.e LSB vestigal (some portion) occupied.

→ The bandwidth required by the VSB modulated wave is given by $BW = w + f_v$

where w = message Bandwidth

f_v = width of vestigial sideband.

Time Domain description of VSB waves :-

Let us consider s(t) VSB modulated wave containing a vestige of LSB and viewed as output of sideband shaping filter in response to input of DSBSC modulated wave.

→ The sideband shaping filter can be represented by its transfer function

$$\tilde{H}(f) = \tilde{H}_u(f) - \tilde{H}_v(f)$$

where $\tilde{H}_u(f)$ = Bandpass filter designed to reject the lower sideband completely

$\tilde{H}_v(f)$ = Vestige of LSB + removal portion of USB

$$\tilde{H}_u(f) = \begin{cases} \frac{1}{2}[1+\mathrm{sgn}(f)], & 0<f<w \\ 0 & \text{otherwise} \end{cases}$$

Substitute $\tilde{H}_u(f)$ in $\tilde{H}(f)$ we get

$$\tilde{H}(f) = \begin{cases} \frac{1}{2}[1+\mathrm{sgn}(f) - 2\tilde{H}_v(f)], & -f_v<f<w \\ 0, & \text{otherwise} \end{cases} \quad \text{--- (1)}$$

sgn(f) and $\tilde{H}_V(f)$ both odd function of frequency they both have purely imaginary inverse Fourier Transform and it is possible new transfer function

$$H_Q(f) = \frac{1}{j}\left[\operatorname{sgn}(f) - 2\tilde{H}_V(f)\right]$$

$h_Q(t) \rightleftharpoons H_Q(f) \rightarrow$ Real inverse fourier Transform of $H_Q(f)$

Substitute $H_Q(f)$ in equation (1) we get

$$\tilde{H}(f) = \begin{cases} \frac{1}{2}\left[1 + jH_Q(f)\right], & -f_V < f < \omega \\ 0, & \text{otherwise} \end{cases} \quad \text{—— (2)}$$

Lets determine VSB modulated wave s(t) i.e

$$s(t) = \operatorname{Re}\left[\tilde{S}(t)\, e^{j2\pi f_c t}\right]$$

$$\text{Spectrum of } \tilde{S}(f) = \tilde{H}(f)\, \underset{DSBSC}{\tilde{S}(f)}$$

$$\left[\because \underset{DSBSC}{\tilde{S}(f)} = A_c m(f)\right]$$

$$\tilde{S}(f) = \tilde{H}(f) A_c m(f)$$

From eq (2) we get $\tilde{H}(f)$ value

$$\tilde{S}(f) = \frac{A_c}{2}\left[1 + j H_Q(f)\right] M(f)$$

Apply Inverse fourier transform to above equation

$$\tilde{s}(t) = \frac{A_c}{2}\left[m(t) + j\, m_Q(t)\right]$$

VSB modulated wave $s(t) = Re\left\{\tilde{s}(t)\, e^{j2\pi f_c t}\right\}$

$$s(t) = Re\left\{\frac{A_c}{2}\left[m(t) + j m_Q(t)\right] e^{j2\pi f_c t}\right\}$$

$$= \frac{A_c}{2}\left[m(t)\cos 2\pi f_c t + j m_Q(t)\, j \sin(2\pi f_c t)\right]$$

$$= \frac{A_c}{2}\left[m(t)\cos 2\pi f_c t + j^2 m_Q(t) \sin 2\pi f_c t\right]$$

$$j^2 = -1$$

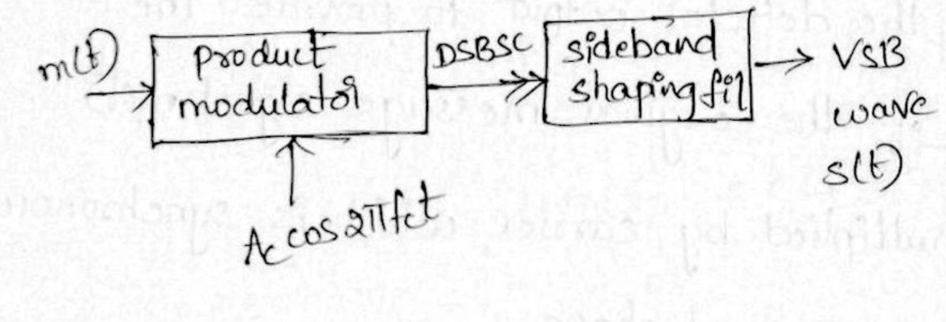
$$\therefore s(t) = \frac{A_c}{2}\left[m(t)\cos 2\pi f_c t\right] - \frac{A_c}{2} m_Q(t) \sin 2\pi f_c t$$

$\frac{A_c}{2} \rightarrow$ inphase component of VSB modulated wave

$\frac{1}{2} A_c m_Q(t) \rightarrow$ Quadrature component of VSB modulated wave.

Generation of VSB modulated wave :-

We can generate the VSB modulated wave by passing DSBSC modulated wave through a sideband shaping filter. The filter can be designed to provide desired spectrum of VSB modulated wave.

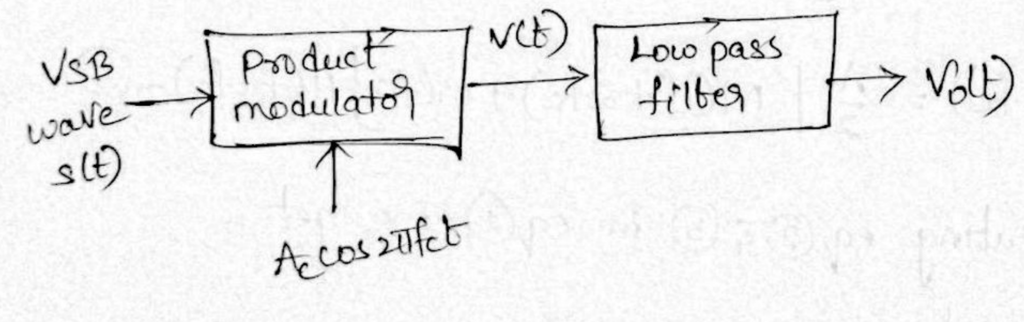


$$s(t) = m(t)\, A_c \cos 2\pi f_c t$$

The relation between the transfer function H(f) of the filter and the spectrum s(f) of VSB modulated wave is given as

$$S(f) = \frac{A_c}{2}\left[M(f-f_c)+M(f+f_c)\right]H(f)$$

<u>Demodulation of VSB modulated wave</u> :-



→ The demodulation of VSB modulated wave can be achieved by passing VSB wave $s(t)$ through a coherent detector and then determining the necessary condition for the detector output to provide the undistorted of the original message signal $m(t)$.

→ $s(t)$ is multiplied by carrier which is synchronous in both frequency and phase

$$V(t) = s(t) \cdot A_c \cos 2\pi f_c t$$

By taking Fourier transform we get

$$V(f) = \frac{1}{2}\left[S(f-f_c) + S(f+f_c)\right] \quad \text{--- (1)}$$

Generation of VSB

$$\text{where} \quad S(f) = \frac{A_c}{2}\left[M(f-f_c) + M(f+f_c)\right]H(f)$$

$$S(f-f_c) = \frac{A_c}{2}\left[M(f-f_c-f_c) + M(f+\cancel{f_c}-\cancel{f_c})\right]H(f-f_c)$$

$$S(f-f_c) = \frac{A_c}{2}\left[M(f-2f_c) + M(f)\right]H(f-f_c) \quad \text{--- (2)}$$

$$\text{Ily} \quad S(f+f_c) = \frac{A_c}{2}\left[M(f+2f_c) + M(f)\right]H(f+f_c) \quad \text{--- (3)}$$

substituting eq (2) & (3) in eq (1) we get

$$V(f) = \frac{1}{2}\left\{\left[\frac{A_c}{2}\left\{(M(f) + M(f-2f_c)\right\} H(f-f_c)\right] + \frac{A_c}{2}\left\{M(f) + M(f+2f_c)\right\} H(f+f_c)\right\}$$

$$= \frac{A_c}{4}\left[M(f) + M(f-2f_c)\right] \cdot H(f-f_c) + \frac{A_c}{4}\left[M(f) + M(f+2f_c)\right] H(f+f_c)$$

$$= \frac{A_c}{4} M(f) H(f-f_c) + \frac{A_c}{4} M(f-2f_c) \cdot H(f-f_c) + \frac{A_c}{4} M(f) H(f+f_c) + \frac{A_c}{4} M(f+2f_c) H(f+f_c)$$

$$V(f) = \underbrace{\frac{A_c}{4} M(f)\left[H(f-f_c) + H(f+f_c)\right]}_{o/p} + \frac{A_c}{4}\left[M(f-2f_c) \cdot H(f-f_c) + M(f+2f_c) \cdot H(f+f_c)\right]$$

→ The unwanted term can be removed by the low pass filter to produce an output of $V_o(t)$. The spectrum of $V_o(t)$ can be given as

$$V_o(f) = \frac{A_c}{4} M(f)\left[H(f-f_c) + H(f+f_c)\right].$$

$$2H(f_c) = H(f-f_c) + H(f+f_c)$$

where $H(f_c)$ is constant. The transfer function $H(f)$

must satisfy the condition in order to get distortionless original message signal at the coherent detector o/p.

Envelope detection of VSB wave pulse carrier:-

→ This detection consists of non-coherent. To make demodulation of VSB wave by an envelope detector at the Receiving end it is necessary to transmit a sizeable carrier together with modulated wave

$$s(t) = A_c \cos(2\pi f_c t) + \frac{A_c}{2} k_a m(t) \cos 2\pi f_c t - \frac{A_c}{2} k_a m_Q(t) \sin 2\pi f_c t$$

$$s(t) = A_c\left[1 + \frac{k_a}{2} m(t)\right] \cos 2\pi f_c t - \frac{A_c}{2} k_a m_Q(t) \sin 2\pi f_c t$$

k_a = modulation Index

→ The above signal passed through envelop detector, the detector output

$$a(t) = A_c\left[\left(1 + \frac{k_a}{2} m(t)\right)^2 + \left(\frac{k_a}{2} m_Q(t)\right)^2\right]^{1/2}$$

Advantages of VSB modulated wave :-

1. Highly efficient technique for reduction of Bandwidth
2. Filter design is very easy and it is high accuracy
3. Transmission of low frequency components is possible without difficulty

Disadvantages :-

1. when compared to Bandwidth SSB is greater
2. Demodulation process is very complex

Applications :-

→ Television signals contain significant components at extremely low frequencies and hence VSB modulation is used in television transmission.

Comparision of AM Techniques :-

Parameter	standard AM	DSBSC	SSB	VSB
1. Power	High	Medium	Less	< DSBSC > SSB
2. carrier Supression	NO	Yes	Yes	NO
3. Modulation Type	Non-Linear	Linear	Linear	Linear
4. Side band supression	NO	NO	One sided completely	one side band supressed partialy
5. Application	Radio communication	point-point communication	point-point communication for long distance	Television broadcasting
6. Transmission efficiency	min	moderate	max	moderate
7 Bandwidth	$2f_m$	$2f_m$	f_m	$f_m < B_w < 2f_m$
8. $R_x$er complexity	simple	complex	complex	simple

1. AMPLITUDE MODULATION

Introduction to communication system :-

Communication :- It is the process of exchanging information. (or) Transfer of information between two or more points. Information may be voice, television picture, computer data or some other form of electronic information.

→ The electronic equipments which are used for communication purpose are called communication equipment. different communication equipment assembled together form a communication system.

→ In beginning, communication over long distances posed the problem. The communication took a dramatic change in the late 19th century when electricity was discovered.

→ Telegraph was invented in 1844 and telephone in 1876 Television was invented in 1923, the first communication satellite was launched in 1962.

→ communication is of two types i.e

1. Analog communication and 2. Digital communication.

Elements of communication system :-

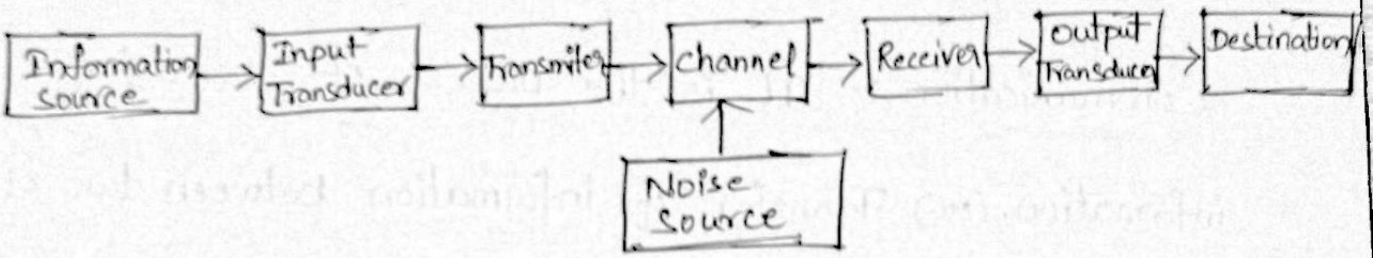


Fig :- Block diagram of communication system

→ Noise is inherently present in the channel or medium It gets added to the information being communicated

Information source :- It contains required message which has to be transmitted. message like Audio, sound, picture, data etc. Amount of any information measured in bits.

Transducer :- It is used to convert one form of energy to another form i.e convert non-electrical to electrical signals. and electrical to non-electrical signals.

Transmitter :- collection of electronic circuits designed to convert the information into a signal suitable for transmission over a given communication medium.

→ mostly message from information source is non-electrical and it is suitable for Immediate transmission such message need to be coded.

→ Most of the Transmitter have buit in amplifier circuit and these amplify the incoming signal before transmission.

channel :- communication channel is the medium by which the electrical signal is transmitted from one to another. Examples - conducting pair of wires, fiber optic cable used for light and Infrared Transmission

Noise :- During Transmission signal gets distorted. Noise cannot be completely eliminated.

Receiver :- collection of electronic circuits designed to convert the signal back to the original information

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

Baseband signal :- Original information signals are Analog or digital they are all referred as "baseband signals".

→ In a communication system, the original information signals may be transmitted over the medium. putting the original signal directly into the medium is referred to as "baseband transmission".

Limitations of Baseband Transmission :-

→ There are many instances when the baseband signals are incompatible for direct transmission over the medium

Ex:- Voice signals cannot travel longer distance in air. the signal gets attenuated rapidly.

→ Hence for transmission of baseband signals by radio, modulation technique has to be used

Modulation :- Modulation is a process a high frequency carrier signal is varied in accordance with the low frequency modulating signal.

Need for Modulation and Advantages :-

Advantages of using modulation technique

1. Reduces the height of Antenna
2. Avoids mixing of signals
3. Increases the range of communication
4. Allows multiplexing of signals
5. Allows Adjustments in the Bandwidth
6. Improves quality of Reception.

1. Reduces the height of Antenna :-

The height of the antenna required for Transmission and reception of radio waves is a function of wavelength of the frequency used. The minimum height of Antenna is given as $\lambda/4$

wavelength $\lambda = \frac{c}{f}$ → Velocity of light

for ex baseband signal with $f = 15\,kHz$

$$\text{height of antenna} = \frac{\lambda}{4} = \frac{c}{f \times 4} = \frac{3 \times 10^8}{15\,kHz \times 4}$$

$$= \frac{3 \times 10^8}{15 \times 10^3 \times 4} = 5000 \text{ meters}$$

5000 meters antenna unthinkable & unpracticable

if we consider baseband signal 1MHz the height of antenna $= \frac{\lambda}{4} = \frac{c}{f \times 4} = \frac{3 \times 10^8}{1MHz \times 4}$

$= \frac{3 \times 10^8}{1 \times 10^6 \times 4} = 75 \text{ meters}$

height of antenna is practical & such antennas can be installed

2. <u>Avoids mixing of signals</u> :- All sound signals are concentrated within range from 20Hz to 20kHz.

→ Transmission of baseband signals from various source causes the mixing of signal and it is difficult to seperate at the Receiver inorder to seperate we are passing them through a channel each must be given its own Bandwidth commonly known as channel Bandwidth

→ Once the signal transmitted, a tuned circuit at the Receiver end selects the paticular signal.

3. <u>Increases the Range of communication</u> :-

→ Baseband signal cannot directly transmitted over long distance because of low frequencies.

→ Modulation increases the frequency of signal to be radiated and signal can be transmitted over long distance.

4. Allows multiplexing of signals :-

→ multiplexing means Transmission of two or more signals simultaneously over the same channel.

→ Best Example Television channel and Radio station Broad-casting. These signals seperated at the Receiver end known as tuning the Receiver to the desired station and unwanted signal are rejected.

5. Allows Adjustments in the Bandwidth :-

→ Bandwidth may be smaller or larger than original signal.

→ signal to noise ratio in the Receiver which is a function of signal bandwidth can be improved by proper control of bandwidth at modulating stage.

6. Improves quality of reception:-

Modulation technique such as frequency modulation, pulse code modulation reduce the effect of noise

→ Reduce noise improves quality Reception

Frequency spectrum :-

Type of signal	frequency Range	Application
VLF	3 - 30kHz	Long distance voice communication
LF	30 - 300kHz	Radio Navigation, Aeronatical
MF	300kHz - 3MHz	AM radio broadcasting (535 - 1645kHz)
HF	3 - 30MHz	Radio Telephony
VHF	30 - 300MHz	FM broadcasting (88 - 108 MHz)
UHF	300MHz - 3GHz	TV broadcasting, mobile communication
SHF	3 - 30GHz	Microwave frequencies
EHF	30 - 300GHz	satellite communication
Infrared	> 300 GHz	TV remote control Application
Visible spectrum	> 300GHz	optical communication.

Communication system

- Analog communication
  - Continuous mode
    - AM (Amp modulation)
    - Angle
      - FM (Freq modu)
      - PM (phase mod)
  - pulse mode
    - PAM (Pulse Amp modulation)
    - PWM (pulse width modulation)
    - PPM (pulse position modulation)
- Digital communication
  - PCM
  - DM
  - DPCM

Types of Modulation
in Analog

- Continuous wave modulation
  - Amplitude modulation (AM)
  - Angle modulation
    - Frequency modulation (FM)
    - phase modulation (PM)
- Pulse modulation
  - Digital modulation
    - PCM - pulse code mod
  - Analog modulation
    - PAM
    - PWM
    - PPM

Multiplexing :-

→ Multiplexing transmits more than one message signal on the same carrier and on the same channel (or)

→ No of transmitters are transmitting simultaneously on the same channel is called multiplexing.

There are two types of multiplexing

1. Frequency Division Multiplexing (FDM)
2. Time Division Multiplexing (TDM)

Frequency Division Multiplexing :-

→ Multiplexing requires that the signal be kept apart so that they do not interfere with each other and they can be seperated at the Receiving end.

→ The Technique of separating the signals in frequency is referred to as frequency Division Multiplexing.

→ Any type of modulation can be used in FDM as long as the carrier spacing is sufficient to avoid spectral overlapping.

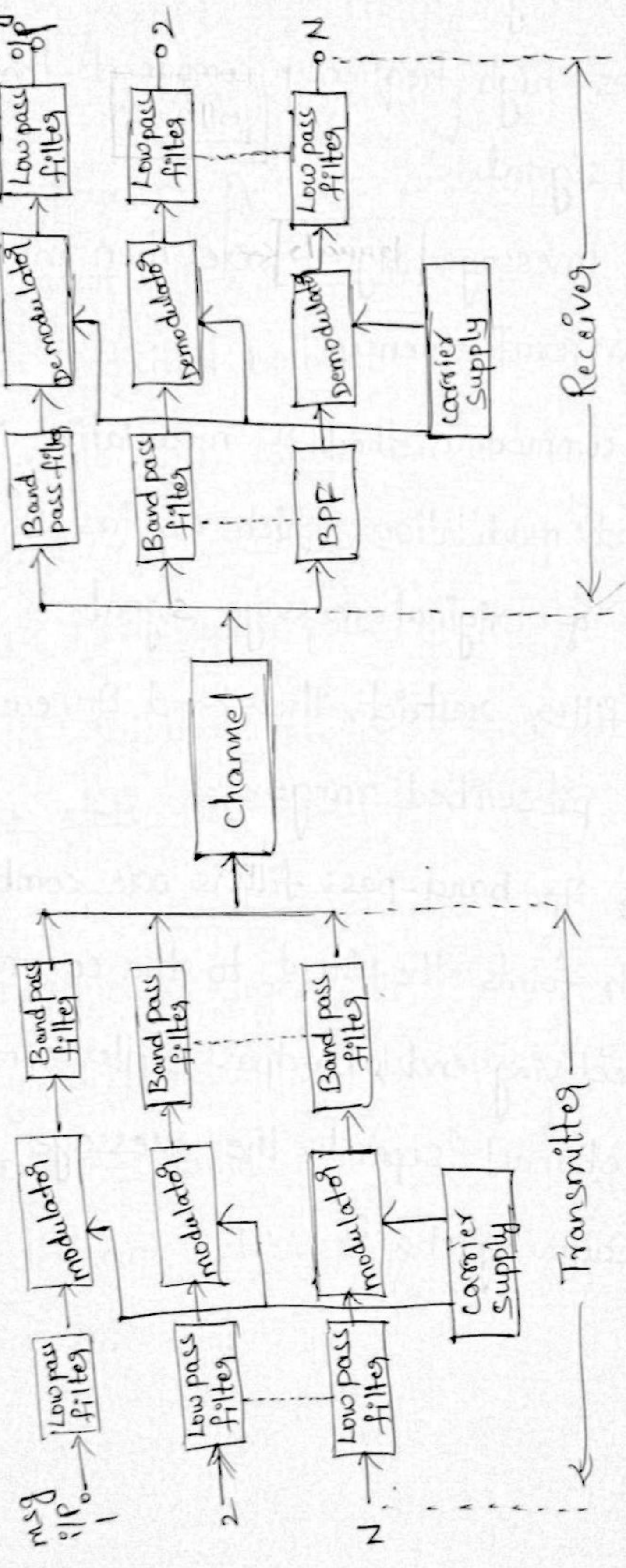


Fig:- Block diagram of FDM

→ Input message signal assumed low pass type and are passed through Low-pass filters. This filtering action removes high frequency component that do not contribute to signal

→ The filter message signal are then modulated with necessary carrier frequency

→ The most common method of modulation in FDM is single sideband modulation which requires Bandwidth equal to that of original message signal.

→ Band pass filter restrict the band of each modulated wave to its prescribed range.

→ The outputs of band-pass filters are combined in parallel which forms the input to the common channel.

→ At the Receiving end, bandpass filter connected to the common channel seperate the message signal on frequency occupancy basis.

Time Division Multiplexing (TDM) :-

→ The Technique of separating the signals in time is called time-division Multiplexing.

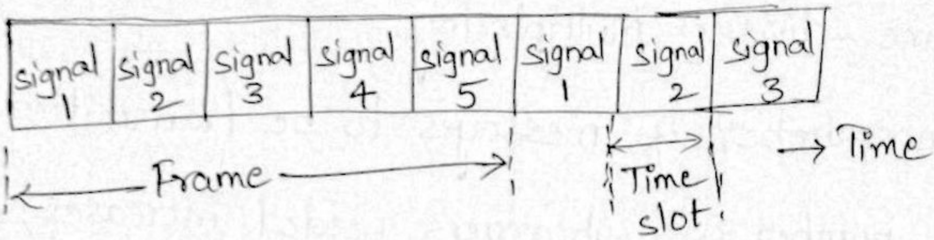
→ As the number of messages to be transmitted increases, number of Subcarriers needed increases, Additional circuitry is required both at transmitting and Receiving ends to handle each added channel.

→ These problems are eliminated by using time division Multiplexing (TDM) with pulse modulation.

→ TDM is suitable for DC. Here we are using the no. of time slots for each input whereas FDM is used for AC

→ In TDM, each signal to be transmitted is sampled sequentially and resulting pulse code used to modulate the carrier. The same carrier frequency is used to transmit different pulses sequentially, one after other, in a given time slot.

→ since only one signal modulates the carrier at any time, no added equipment and no increase in bandwidth is needed when multiplexing.



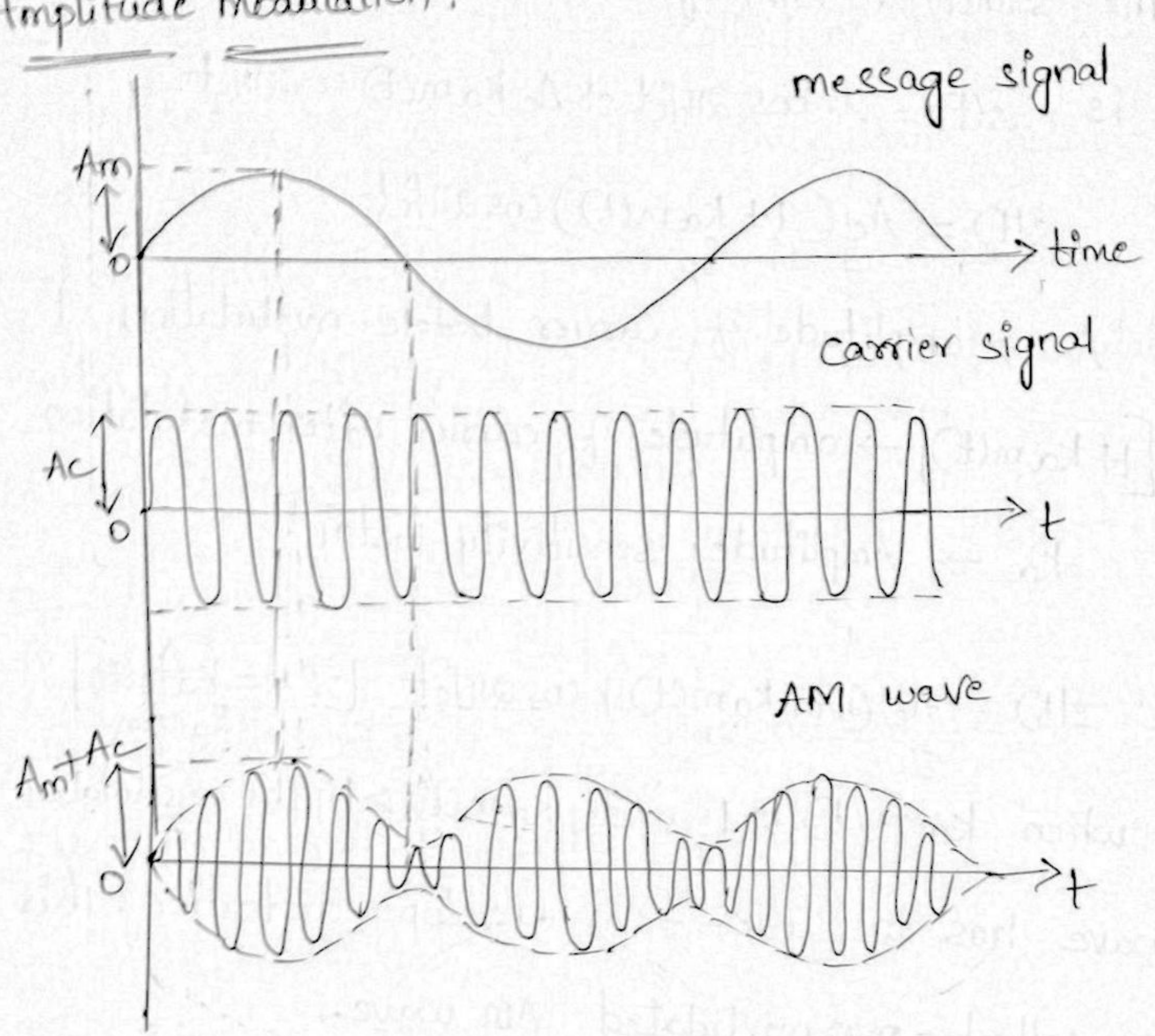
→ Five signals are time division multiplexed. each signal is allowed to use the channel for fixed interval of time called "time slot". The 5 signals use the channel sequentially one after other

→ one transmission of each channel completes one cycle of operation called "frame". once all signals have been transmitted, the cycle repeats again and again at a high rate of speed.

Amplitude Modulation :- The amplitude of the high frequency carrier signal is varied in accordance with the instantaneous message signal/ modulating signal/ base band signal is known as Amplitude modulation.

→ In AM frequency & phase same but amplitude Varies.

Amplitude modulation :-



Time Domain Description :-

Instantaneous value of modulating signal and carrier signal is given by

$$m(t) = A_m \cos 2\pi f_m t$$

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

where A_m = maximum amplitude of modulating signal

A_c = maximum amplitude of carrier signal.

The standard form of Amplitude modulated wave

is $s(t) = A_c \cos 2\pi f_c t + A_c K_a m(t) \cos 2\pi f_c t$

$s(t) = A_c(1 + K_a m(t)) \cos 2\pi f_c t$

$A_c \rightarrow$ amplitude of carrier before modulation

$[1 + K_a m(t)] \rightarrow$ amplitude of carrier after modulation

$K_a \rightarrow$ Amplitude sensitivity factor

$\therefore s(t) = A_c(1 + K_a m(t)) \cos 2\pi f_c t \quad \left[\because \mu = \frac{K_a A_m}{K_a \cdot A_m}\right]$

→ when $K_a \cdot m(t) < 1$ and $K_a m(t) > 1$ the modulated wave has to suffer from envelope distortion. This is called overmodulated AM wave.

Modulation Index:-

Modulation Index is defined as the ratio of amplitude of message signal to the amplitude of carrier signal. It is denoted by μ.

$$\mu = \frac{A_m}{A_c}$$

$$\% \mu = \frac{A_m}{A_c} \times 100$$

→ Modulation Index is also called as degree of modulation / Depth of modulation / modulation factor

→ Modulation Index (μ) < 1 it is said to be under modulation, $\mu = 1$ is called critical modulation, $\mu > 1$ is called overmodulation.

frequency domain discription :-

→ The modulated carrier has new signals at different frequencies called sidebands

$$f_{USB} = f_c + f_m$$

$$f_{LSB} = f_c - f_m$$

we know that $s(t) = A_c(1 + k_a m(t)) \cos 2\pi f_c t$

$$s(t) = A_c(1 + k_a A_m \cos 2\pi f_m t) \cos 2\pi f_c t$$

$$= A_c(1 + \mu \cos 2\pi f_m t) \cos 2\pi f_c t$$

$$= A_c \cos 2\pi f_c t + A_c \cdot \mu \cdot \cos 2\pi f_c t \cdot \cos 2\pi f_m t \quad [\because \mu = k_a A_m]$$

$$\therefore s(t) = A_c \cos 2\pi f_c t + \mu A_c \cos 2\pi f_c t \cdot \cos 2\pi f_m t$$

$$[\because \cos a \cos b = \tfrac{1}{2} \cos(a+b) + \cos(a-b)]$$

$$s(t) = A_c \cos 2\pi f_c t + \frac{1}{2}\left[\mu A_c \cos(2\pi f_c + 2\pi f_m)t + \mu A_c \cos(2\pi f_c - 2\pi f_m)t \right]$$

$$s(t) = A_c \cos 2\pi f_c t + \frac{\mu A_c}{2} \cos(2\pi f_c + 2\pi f_m)t + \frac{\mu A_c}{2} \cos(2\pi f_c - 2\pi f_m)t$$

From the above eq 1^{st} term represents unmodulated carrier and 2 additional term represent two sidebands The freq of lower sideband is $f_c - f_m$ and freq of upper sideband is $f_c + f_m$.

→ Bandwidth of Am wave is defined as freq can be measured by substracting lowest frequency of signal from the highest frequency of signal

$$BW = f_{USB} - f_{LSB}$$

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

$$= \cancel{f_c} + f_m - \cancel{f_c} + f_m = 2f_m = 2w$$

w is the message bandwidth.

Bandwidth required for AM is twice the frequency of the modulating signal.

Single-Tone Amplitude Modulation :-

Amplitude Modulation in which the modulating signal consist of only one frequency i.e modulation is done by a single frequency. This type of modulation is known as single tone modulation

Let us consider a single tone modulating signal

$$x(t) = V_m \cos \omega_m t$$

we know that general expression of AM modulation

$$s(t) = \cos\omega_c t \left(A + x(t)\right)$$

$$s(t) = \cos\omega_c t \left(A + V_m \cos\omega_m t\right)$$

$$= A\cos\omega_c t + V_m \cos\omega_c t \cos\omega_m t$$

$$= A\cos\omega_c t\left[1 + \frac{V_m}{A}\cos\omega_m t\right]$$

$$= A\cos\omega_c t\left[1 + m_a\cos\omega_m t\right] \quad \left[\because \frac{V_m}{A} = m_a\right]$$

$$= A\cos\omega_c t + A m_a \cos\omega_m t \cdot \cos\omega_c t$$

$$\left[2\cos a\cos b = \cos(A+B) + \cos(A-B)\right]$$

$$= A\cos\omega_c t + \frac{A m_a}{2}\left[2\cos\omega_m t\cos\omega_c t\right]$$

$$= A\cos\omega_c t + A\frac{m_a}{2}\left[\cos(\omega_c+\omega_m)t + \cos(\omega_c-\omega_m)t\right]$$

$$= A\cos\omega_c t + \frac{A m_a}{2}\cos(\omega_c+\omega_m)t + A\frac{m_a}{2}\cos(\omega_c-\omega_m)t$$

$$s(t) = A\cos 2\pi f_c t + \frac{A m_a}{2}\cos(2\pi f_c + 2\pi f_m)t + \frac{A m_a}{2}\cos(2\pi f_c - 2\pi f_m)t$$

Applying Fourier Transform we get

$$S(f) = \frac{1}{2}A\left[\delta(f-f_c)+\delta(f+f_c)\right] + \frac{A m_a}{4}\left[\delta(f-(f_c+f_m)) + \delta(f+f_c+f_m)\right] + \frac{A m_a}{4}\left[\delta(f-(f_c-f_m)) + \delta(f+(f_c-f_m))\right]$$

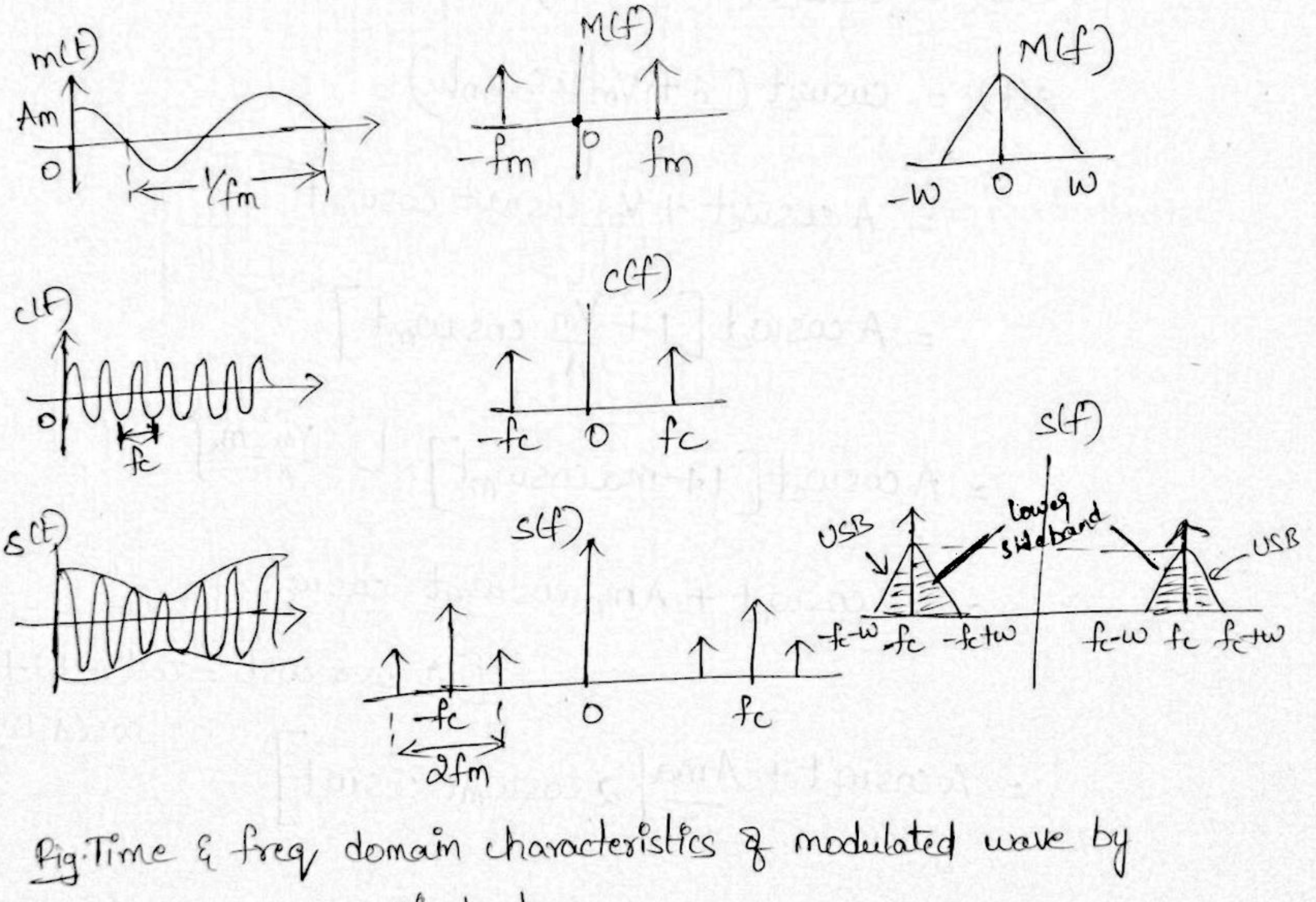
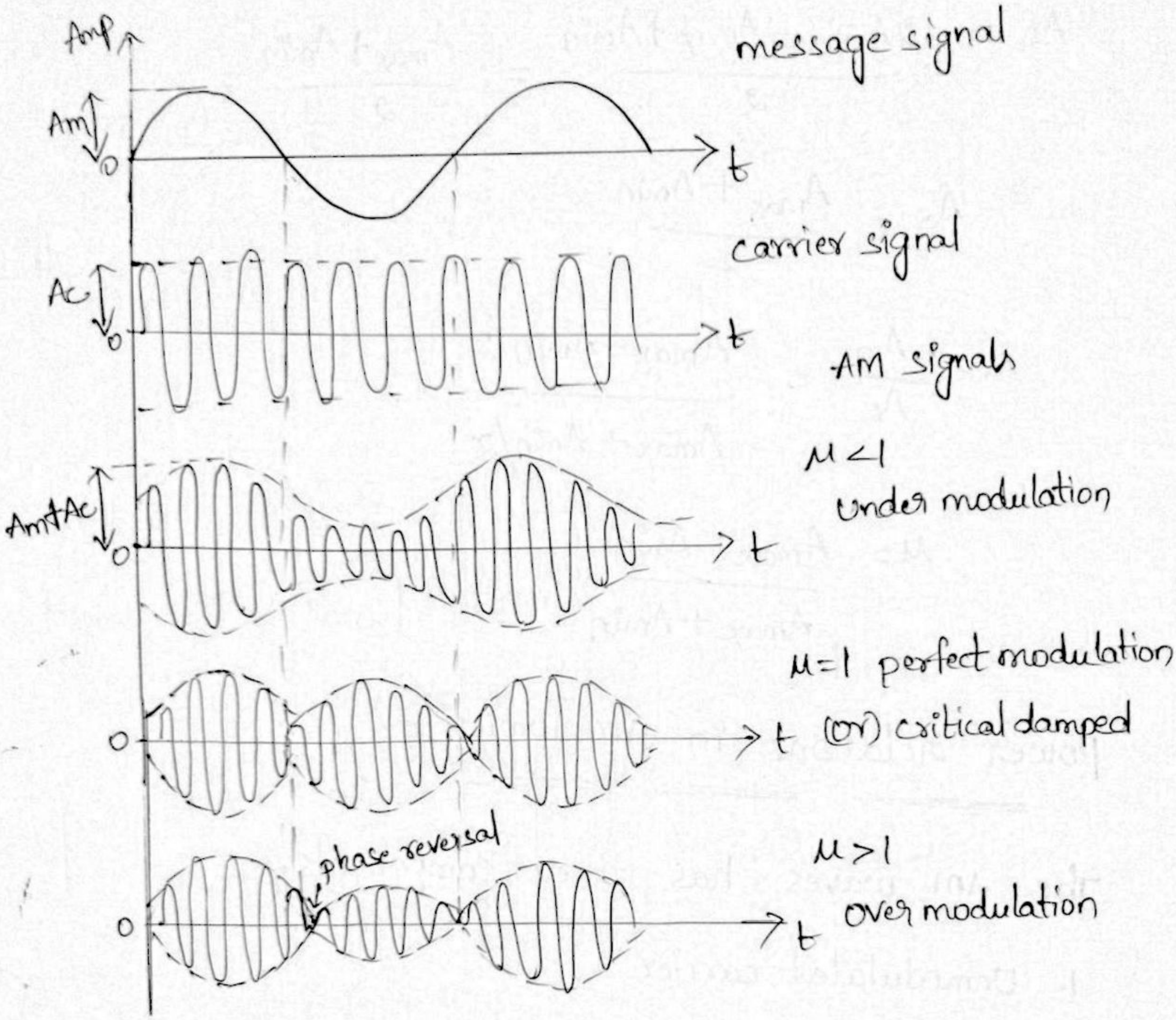


Fig. Time & freq domain characteristics of modulated wave by single tone.

Modulation Index :- Different waves for AM



Calculation of modulation Index :-

$$\mu = \frac{A_m}{A_c}$$ where A_m is given as

$$A_m = \frac{A_{max} - A_{min}}{2}$$

$$A_c = A_{max} - A_m$$

$$A_c = A_{max} - \left(\frac{A_{max} - A_{min}}{2}\right)$$

$$A_c = \frac{2A_{max} - A_{max} + A_{min}}{2} = \frac{A_{max} + A_{min}}{2}$$

$$A_c = \frac{A_{max} + A_{min}}{2}$$

$$\mu = \frac{A_m}{A_c} = \frac{A_{max} - A_{min}/\cancel{2}}{A_{max} + A_{min}/\cancel{2}}$$

$$\mu = \frac{A_{max} - A_{min}}{A_{max} + A_{min}}$$

Power relations in AM waves :-

The AM waves has three components

1. Unmodulated carrier
2. Lower side band
3. Upper side band

→ The total power of AM wave is the sum of carrier power (P_c), power in 2 sidebands (P_{LSB}, P_{USB})

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

carrier power $P_c = \dfrac{\left(\frac{A_c}{\sqrt{2}}\right)^2}{R}$ $\left[P = \dfrac{V_{rms}^2}{R} \right.$

$V_{rms} = \dfrac{V_m}{\sqrt{2}}$

$= \dfrac{A_c^2}{2} \times \dfrac{1}{R} = \dfrac{A_c^2}{2R}$ $\left. V_m = \dfrac{A_c}{\sqrt{2}} \right]$

$$P_{USB} = P_{LSB} = \frac{\left(\frac{\mu A_c}{2\sqrt{2}}\right)^2}{R}$$

$$= \frac{\mu^2 A_c^2}{4 \times 2} \times \frac{1}{R} = \frac{\mu^2 A_c^2}{8R}$$

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

$$= \frac{A_c^2}{2R} + \frac{\mu^2 A_c^2}{8R} + \frac{\mu^2 A_c^2}{8R}$$

$$= \frac{A_c^2}{2R}\left[1 + \frac{\mu^2}{4} + \frac{\mu^2}{4}\right] = \frac{A_c^2}{2R}\left[1 + \frac{\mu^2}{2}\right]$$

$$P_{Total} = P_c\left(1 + \frac{\mu^2}{2}\right)$$

modulation Index of power is given by

$$1 + \frac{\mu^2}{2} = \frac{P_{Total}}{P_c} \Rightarrow \frac{\mu^2}{2} = \frac{P_{Total}}{P_c} - 1$$

$$\mu^2 = 2\left(\frac{P_{Total}}{P_c} - 1\right) \Rightarrow \mu = \sqrt{2\left(\frac{P_{Total}}{P_c} - 1\right)}$$

Effective voltage and current for AM waves :-

$$P_{Total} = (I_{total})^2 \times R$$

$$= I_c^2 \times R$$

we known that $\frac{P_{Total}}{P_c} = 1 + \frac{\mu^2}{2}$

$$\frac{I_{total}^2 \times \cancel{R}}{I_c^2 \times \cancel{R}} = 1 + \frac{\mu^2}{2}$$

$$\frac{I_{total}}{I_c} = \sqrt{1 + \frac{\mu^2}{2}} \Rightarrow I_{total} = I_c\sqrt{1 + \frac{\mu^2}{2}}$$

Modulation Index interms of current :-

$$1 + \frac{\mu^2}{2} = \frac{I_{total}^2}{I_c^2} \Rightarrow \frac{\mu^2}{2} = \frac{I_{total}^2}{I_c^2} - 1$$

$$\mu^2 = \left(\frac{I_{total}^2 - I_c^2}{I_c^2}\right) 2$$

$$\mu = \sqrt{\left(\frac{I_{total}^2 - I_c^2}{I_c^2}\right) \cdot 2}$$

Generation of AM waves :-

Technique to generate Amplitude modulated waves are 1. Low Level Modulation
2. High Level Modulation.

→ In Low Level modulation technique, the generation of AM wave takes place in the Initial stage of Amplification i.e. low power level. The generated AM signal is then Amplified using no. of amplifier stages.

→ In High Level Modulation, modulation takes place in the final stage of amplification and therefore modulation circuitry has to handle high power

→ In this we study how to generate a standard amplitude modulated wave suited for low level modulation 1. square law modulator
2. switching modulator

→ The above mentioned modulators use non-linear elements for their implementation & well suited for Low Level Modulation

Low Level Amplitude Modulation :-

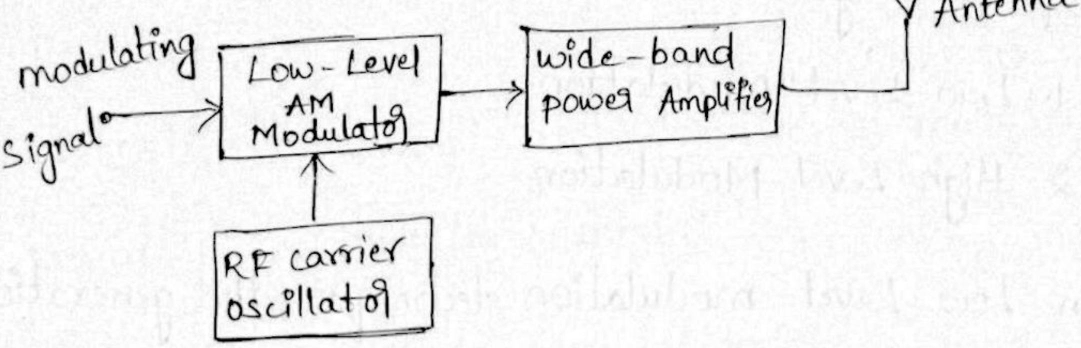


Fig:- Low Level AM modulation block diagram

→ In Low Level AM modulation system, the modulation is done at low level.

→ At low power levels, a very small power is associated with the carrier signal and modulating signal because of this output power of modulation is low. so power Amplifiers are required to boost the amplitude modulated signal upto the desired output level

→ From block diagram the amplitude modulated signal is applied to the wide band power Amplifier

→ A wide band power amplifier used to just preserve the sidebands of the modulated signal

→ Amplitude modulated system employing modulation at low level are called low level amplitude modulation Transmitter.

→ square law modulator & switching modulator are examples of low level modulation.

High Level Amplitude Modulation:-

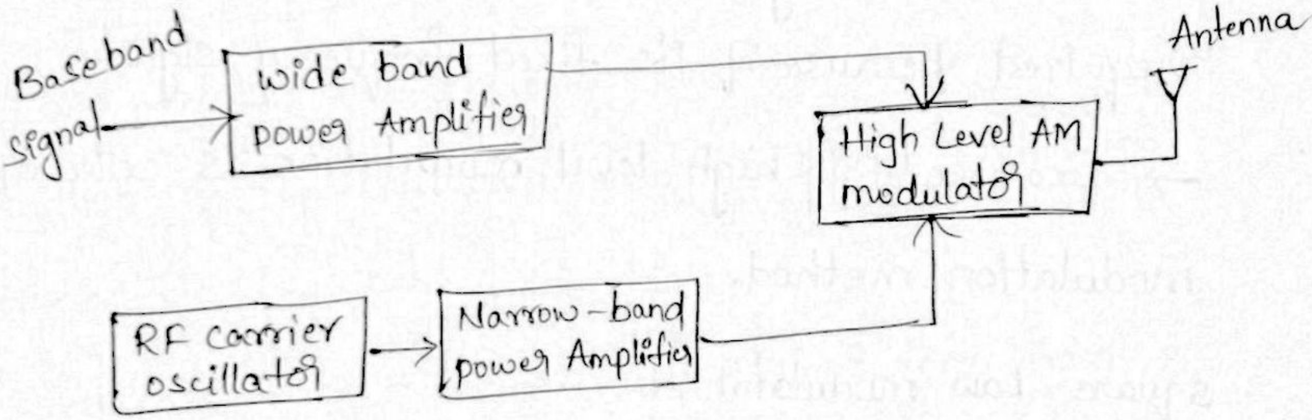


Fig:- Block diagram of high level AM.

→ In high-level amplitude modulation system, the modulation done at high power level

→ Therefore to produce amplitude modulation at these high power levels base band signal and carrier signal must be at high power level

→ from block diagram the modulating signal and carrier signal are first power amplified and then applied to AM-high level modulator.

→ For modulating signal wide band power Amplifier required just to preserve all the frequency components present in the modulating signal

→ For carrier signal the narrow band power amplitude required because of its fixed frequency signal

→ Example for high level modulation is collector modulation method.

Square-Law modulator :-

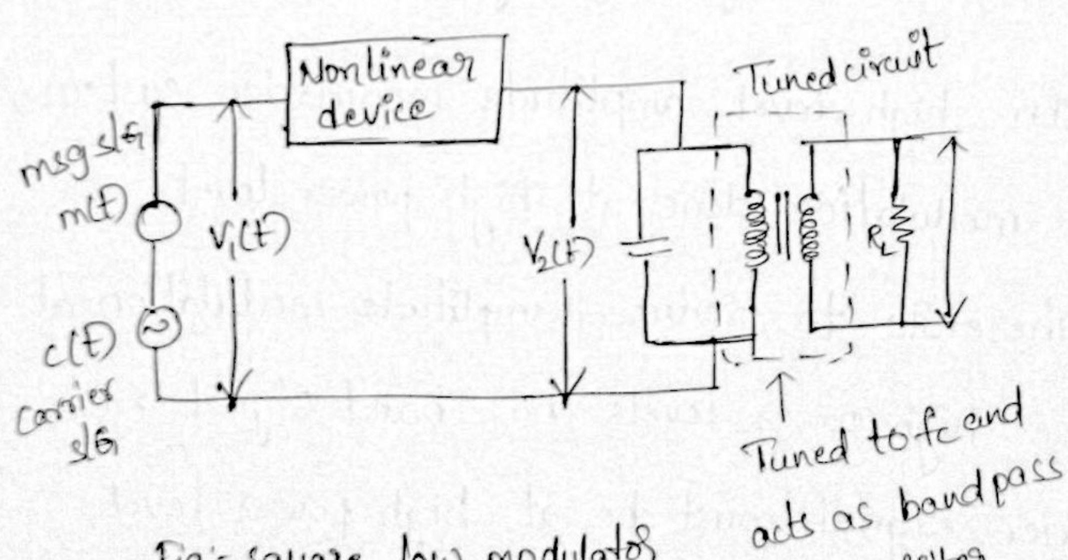


Fig:- square law modulator

→ This modulator consist of 3 elements.

1. Summer or mixer
2. Non-linear device
3. Band pass filter

Summer or Mixer :- It adds the carrier and modulating signal modulator.

Non-linear device :- used to implement square law

~~Band~~ → Non linear device such as diode is suitably portion of its characteristics, the signal applied to the diode is weak.

Band pass filter :- It extracts desired term from modulator products. Bandpass filter is implemented using a single or double tuned frequency.

The mathematical representation of input & output voltage is $V_2(t) = a_1 V_1(t) + a_2[V_1(t)]^2$ —— ①

where $V_1(t) \rightarrow$ input and $V_2(t) \rightarrow$ output

$V_1(t)$ consist of carrier wave and modulating wave

$$V_1(t) = A_c \cos 2\pi f_c t + m(t) \quad \text{--- } ②$$

substitute eq② in eq① we get

$$V_2(t) = a_1\left[A_c \cos 2\pi f_c t + m(t)\right] + a_2\Big[\underbrace{A_c \cos 2\pi f_c t}_{a} + \underbrace{m(t)}_{b}\Big]^2$$

$$= a_1\left[A_c \cos 2\pi f_c t + m(t)\right] + \left[\because (a+b)^2 = a^2+b^2+2ab\right]$$

$$a_2\left[A_c^2 \cos^2 2\pi f_c t + m^2(t) + 2A_c \cos 2\pi f_c t\, m(t)\right]$$

$$= a_1 A_c \cos 2\pi f_c t + a_1 m(t) + a_2 A_c^2 \cos^2 2\pi f_c t + a_2 m^2(t)$$

$$+ a_2 2 A_c \cos 2\pi f_c t \cdot m(t)$$

$$V_2(t) = a_1 A_c \cos 2\pi f_c t\left[1 + \frac{2a_2}{a_1} m(t)\right] + a_1 m(t) + a_2 m^2(t)$$

$$+ a_2 A_c^2 \cos^2 2\pi f_c t$$

$$\left[\because \frac{2a_2}{a_1} = k_a\right]$$

$$V_2(t) = a_1 A_c \cos 2\pi f_c t\left[1 + k_a m(t)\right] + a_1 m(t) +$$

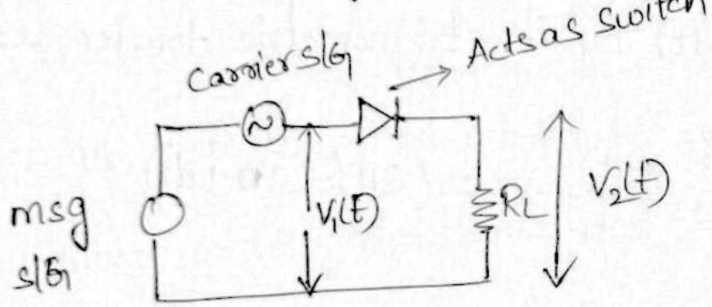
$$a_2 m^2(t) + a_2 A_c^2 \cos^2 2\pi f_c t.$$

→ The first term of equation is standard AM wave with amplitude sensitivity ka and remaining terms are unwanted removed by filter circuit.

Switching Modulator:-

→ In this modulator, multiplication operation is replaced by a simpler switching operation based on ideal diode.

→ positive half cycle of the carrier signal is applied to diode, it is in forward bias. In negative half cycle it is in Reverse bias. This operation will be done in accurately.



$c(t) > 0$ Diode Forwardbias

$c(t) < 0$ Diode Reversebias

→ we know $V_1(t)$ contains message & carrier signal

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

$$m(t) << A_c$$

output $V_2(t)$ can be represented as

$$V_2(t) = \begin{cases} V_1(t) & c(t) > 0 \\ 0 & c(t) < 0 \end{cases}$$

$$V_2(t) = V_1(t) \cdot g_p(t)$$

$$V_2(t) \cong \left[A_c \cos 2\pi f_c t + m(t)\right] g_p(t)$$

→ Load voltage varies periodically between the values $V_1(t)$ and zero at a rate equal to carrier freq

$g_p(t)$ is the periodic pulse train of duty cycle equal to one half.

→ Representing $g_p(t)$ by its trignometric fourier series

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

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

$$\therefore V_2(t) = V_1(t) \cdot g_p(t)$$

$$= \left(A_c \cos 2\pi f_c t + m(t)\right)\left(\frac{1}{2} + \frac{2}{\pi} \cos 2\pi f_c t\right)$$

$$= \underline{\frac{A_c}{2} \cos 2\pi f_c t} + \frac{2}{\pi} A_c \cos^2 2\pi f_c t + \frac{1}{2} m(t)$$

$$+ \underline{\frac{2}{\pi} m(t) \cos 2\pi f_c t}.$$

$$= \frac{A_c}{2} \cos 2\pi f_c t \left[1 + \frac{2}{\pi} m(t) \cdot \frac{2}{A_c}\right] + \frac{1}{2} m(t) + \frac{2}{\pi} \cdot A_c \cdot \cos^2 2\pi f_c t.$$

$$V_2(t) = \frac{A_c}{2}\cos 2\pi f_c t\left[1 + \frac{4}{\pi \cdot A_c} m(t)\right] + \text{Unwanted terms}$$

$$V_2(t) = \frac{A_c}{2}\cos 2\pi f_c t\left[1 + k_a m(t)\right] + \text{unwanted terms}$$

→ The unwanted term can $\left[\because \frac{4}{\pi A_c} = k_a\right]$

be removed by using Bandpass filter.

Detection of AM waves :-

→ A demodulator circuit accepts the modulated signal and recovers the original modulating signal. These circuits Known as detector or demodulator.

→ The most widely used AM demodulators are

1. square law detector
2. Envelope detector

square law detector :-

→ An AM signal can be demodulated by squaring it and then passing the squared signal through a low pass filter.

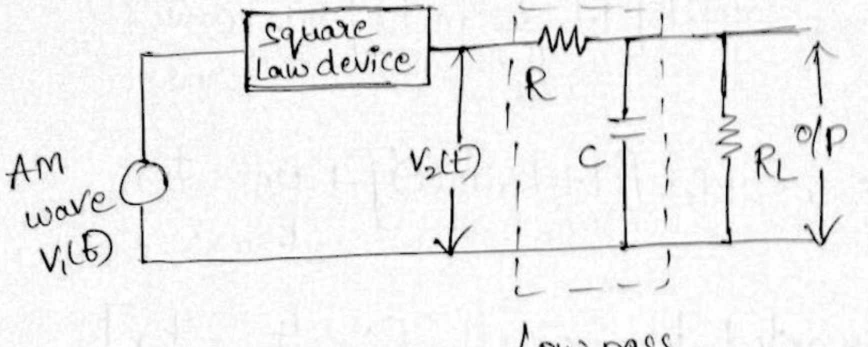


Fig:- square law detector.

→ Transfer characteristics of non-linear device

$$V_2(t) = a_1 V_1(t) + a_2 V_1^2(t) \quad \text{---} (1)$$

$V_1(t)$ and $V_2(t)$ → inputs and output respectively

a_1 and a_2 = constants

$V_1(t)$ represents AM wave & hence it is given as

$$V_1(t) = A_c[1+K_a m(t)]\cos 2\pi f_c t \quad \text{---} (2)$$

Substituting $\overset{eq(1)}{V_1(t)}$ in $\overset{eq(2)}{V_2(t)}$ we get.

$$V_2(t) = a_1[A_c(1+K_a m(t)]\cos 2\pi f_c t + a_2[A_c(1+K_a m(t)]^2 \cos^2 2\pi f_c t$$

$$= a_1 A_c \cos 2\pi f_c t + a_1 K_a m(t)\cos 2\pi f_c t \cdot A_c + a_2[A_c^2(\underset{a}{1}+\underset{b}{K_a m(t)})^2] \cos^2 2\pi f_c t$$

$$= a_1 A_c \cos 2\pi f_c t + a_1 K_a m(t)\cos 2\pi f_c t \cdot A_c +$$

$$a_2[A_c^2(1+K_a^2 m^2(t) + 2K_a m(t))]\cos^2 2\pi f_c t$$

$$= a_1 A_c \cos 2\pi f_c t + a_1 A_c k_a m(t) \cos 2\pi f_c t +$$
$$\left[a_2 A_c^2 + a_2 A_c^2 k_a^2 m^2(t) + 2 a_2 A_c k_a m(t)\right] \cos^2 2\pi f_c t$$

$$= a_1 A_c \cos 2\pi f_c t + a_1 A_c k_a m(t) \cos 2\pi f_c t + \left[a_2 A_c^2 + a_2 A_c^2 k_a^2 m^2(t) + 2 a_2 A_c k_a m(t)\right]\left[\frac{1 + \cos 2(2\pi f_c t)}{2}\right]$$

$$V_2(t) = a_1 A_c \cos 2\pi f_c t + a_1 A_c k_a m(t) \cos 2\pi f_c t +$$
$$\frac{1}{2} a_2 A_c^2 + \frac{a_2 A_c^2 k_a^2 m^2(t)}{2} + \frac{\cancel{2} a_2 A_c k_a m(t)}{\cancel{2}} +$$
$$\frac{a_2 A_c^2 \cos 4\pi f_c t}{2} + \frac{a_2 A_c^2 k_a^2 m^2(t)}{2} \cdot \cos 4\pi f_c t +$$
$$\frac{\cancel{2} a_2 A_c k_a m(t)}{\cancel{2}} \cdot \cos 4\pi f_c t$$

→ From the above eq $\frac{1}{2} a_2 A_c^2 k_a m(t)$ represents the desired signal.

Envelope Detector :-

→ Envelope detector is a simple and highly effective device that is well-suitable for demodulation of narrow-band AM wave.

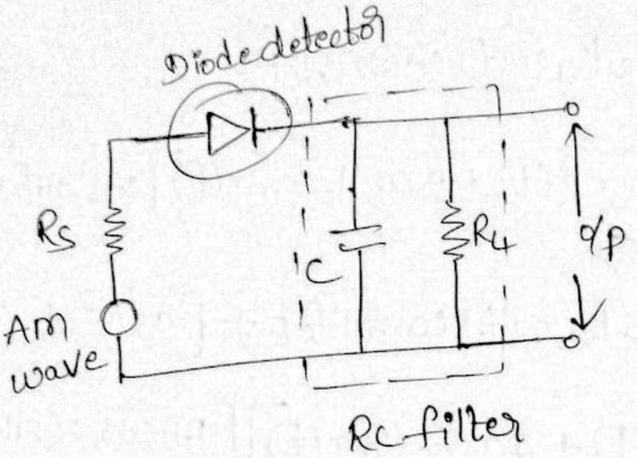
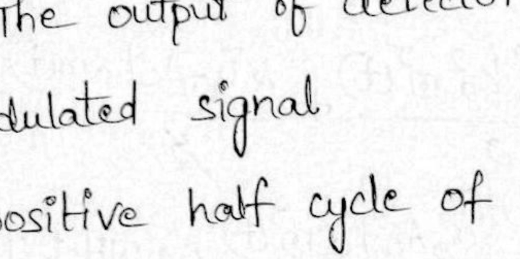


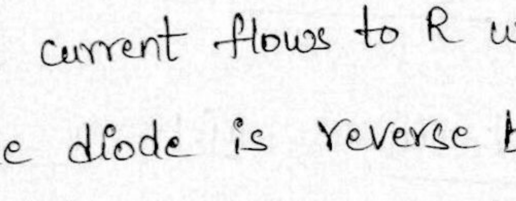
Fig :- Envelope detector

→ The output of detector follows envelope of the modulated signal

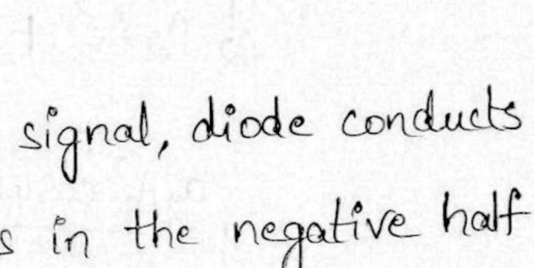
→ positive half cycle of AM signal, diode conducts and current flows to R whereas in the negative half cycle diode is reverse biased and no current flows through R. As a Result only positive half cycle of AM appears across RC.



AM signal



positive half of AM wave



Envelope and voltage across capacitor

other types of AM :-

AM

- DSB-FC
- DSBSC
- SSB
- VSB

Advantages of AM :-

1. These transmitters are less complex
2. AM receivers are simple detection and easy
3. AM receivers are cost efficient
4. These waves can travel a longer distances
5. Low bandwidth

Disadvantages :-

1. Power wastage in DSBFC transmission
2. Bandwidth inefficient system.
3. These waves affected to noise in transmission.

Applications :-

1. Radio broadcasting
2. Video transmission in TV systems.

Power wastage :-

we know $s(t) = A_c(1 + K_a m(t)) \cos 2\pi f_c t$

and $P_{Total} = P_c\left[1 + \frac{\mu^2}{2}\right]$ if $\mu = 1$

$$P_T = P_c\left[1 + \frac{1}{2}\right] = \frac{3}{2} P_c$$

$$P_T = \frac{3}{2} P_c \Rightarrow P_c = P_T \cdot \frac{2}{3}$$

$$P_c = 0.66 \, P_T$$

→ The carrier signal in DSB-FC does not contain any information, the information contain only in modulating signal but the total power 66% of Power will be utilized by carrier signal & remaining power will be utilizing the modulating signal

Bandwidth :-

→ The BW of DSBFC is $2f_m$. Out of 2 sideband only one frequency band is sufficient to convey all the information of message signal.

→ Due to this one sideband is transmitted, lot of power can be saved & BW can be reduced.

Transmitters and Recievers

AM Transmitter

AM transmitter are classified as

* Low power level AM transmitter
* High power level AM tansmitter.

Low-power level AM transmitter

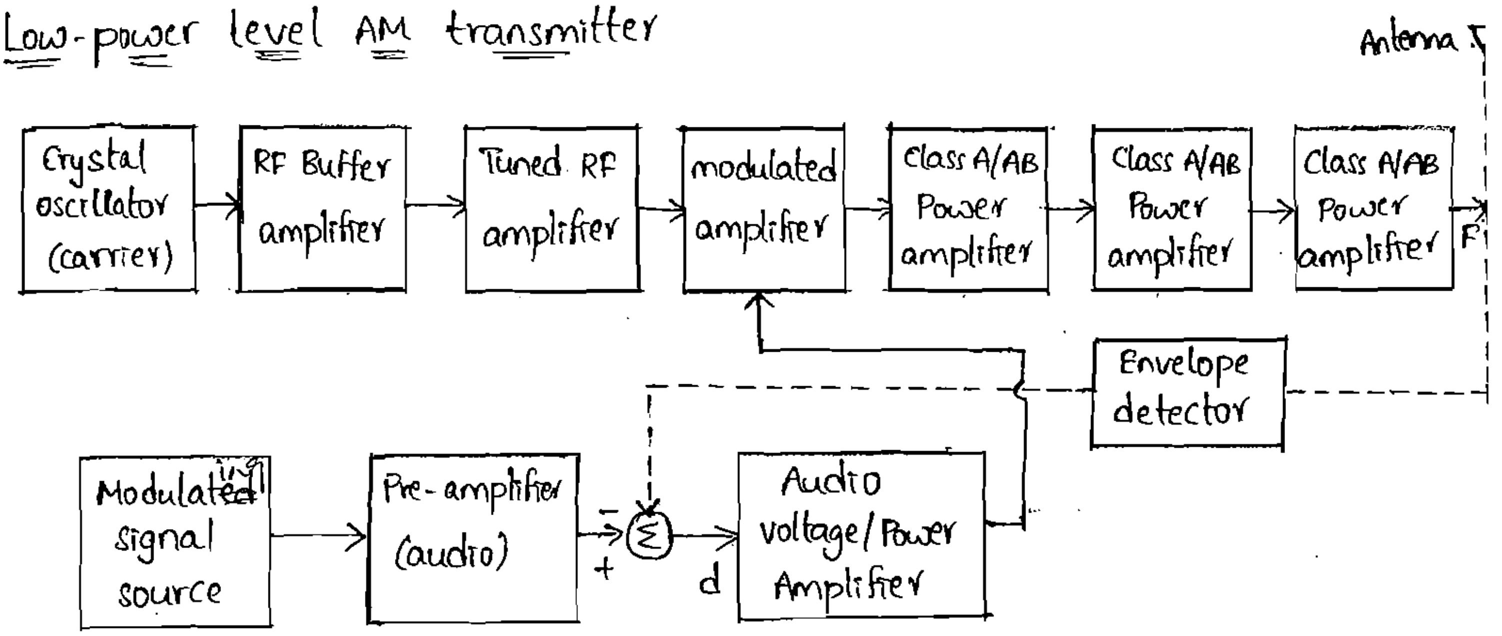


Fig : AM Transmitter with low-level

In the low level transmitters, the modulation process is done at a lower power level and then the modulating signal is passed through a high level power amplifier.

The modulating signal is obtained from a microphone. The pre amplifier is a typically sensitive class A linear voltage amplifier. This amplifier must have a high input impedence. The purpose of the pre amplifier is to bring the source signal to such a level so that the input to the driver amplifier is noise and distortion free. The driver of the modulating signal is also a linear amplifier which amplifies the modulating signal to an adequate level to sufficiently drive the modulator.

The RF Carrier signal is from an oscillator. This can be used to generate the carrier whose frequency stability is quite high. The buffer amplifier is a low-gain, high input impedence linear amplifier that isolates the oscillator from the high power amplifiers. The emitter followers or of late operational amplifiers are used as buffers. Modulators can be either emitter or collector modulation type. The intermediate and Final power amplifiers are generally class A or B push-pull type. This helps to maintain symmetry in the AM envelope. The output impedence of the final power amplifier is matched to antenna by using a antenna matching network.

High power level AM transmitter

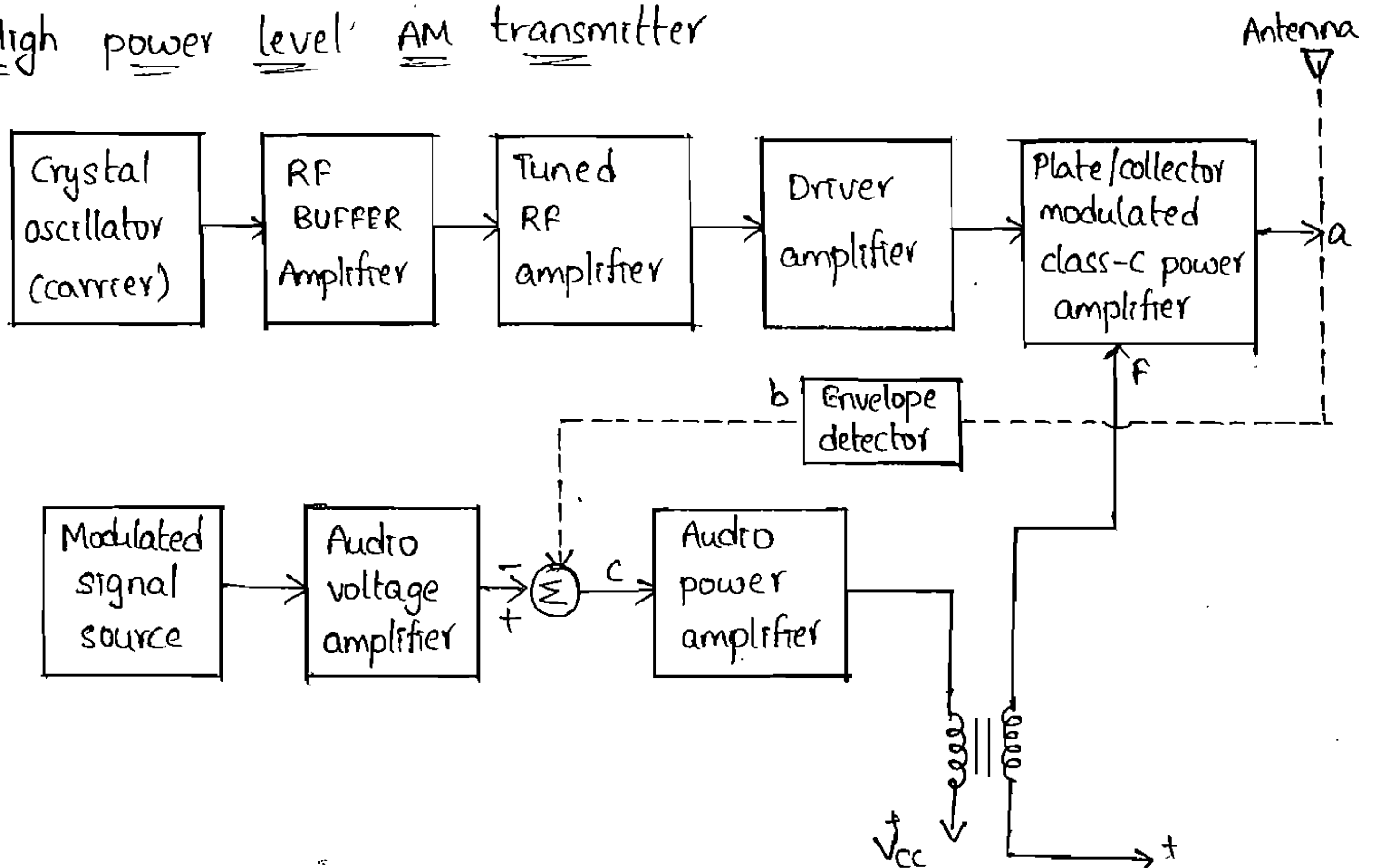


fig : High level modulator

In a high level transmitter, modulation and power amplification are done at a higher level. This requires the modulating signal and carrier signal to be brought to a certain power level before modulation is effected.

The modulating signal goes through the same stage as in the case of low power transmitter except for the addition of a power amplifier. This is due to the fact that for high-level transmitters, the modulating signal should be brought to a higher level before modulating. The carrier will also be at its full power and hence power of the modulating signal be quite high enough to achieve 100% modulation.

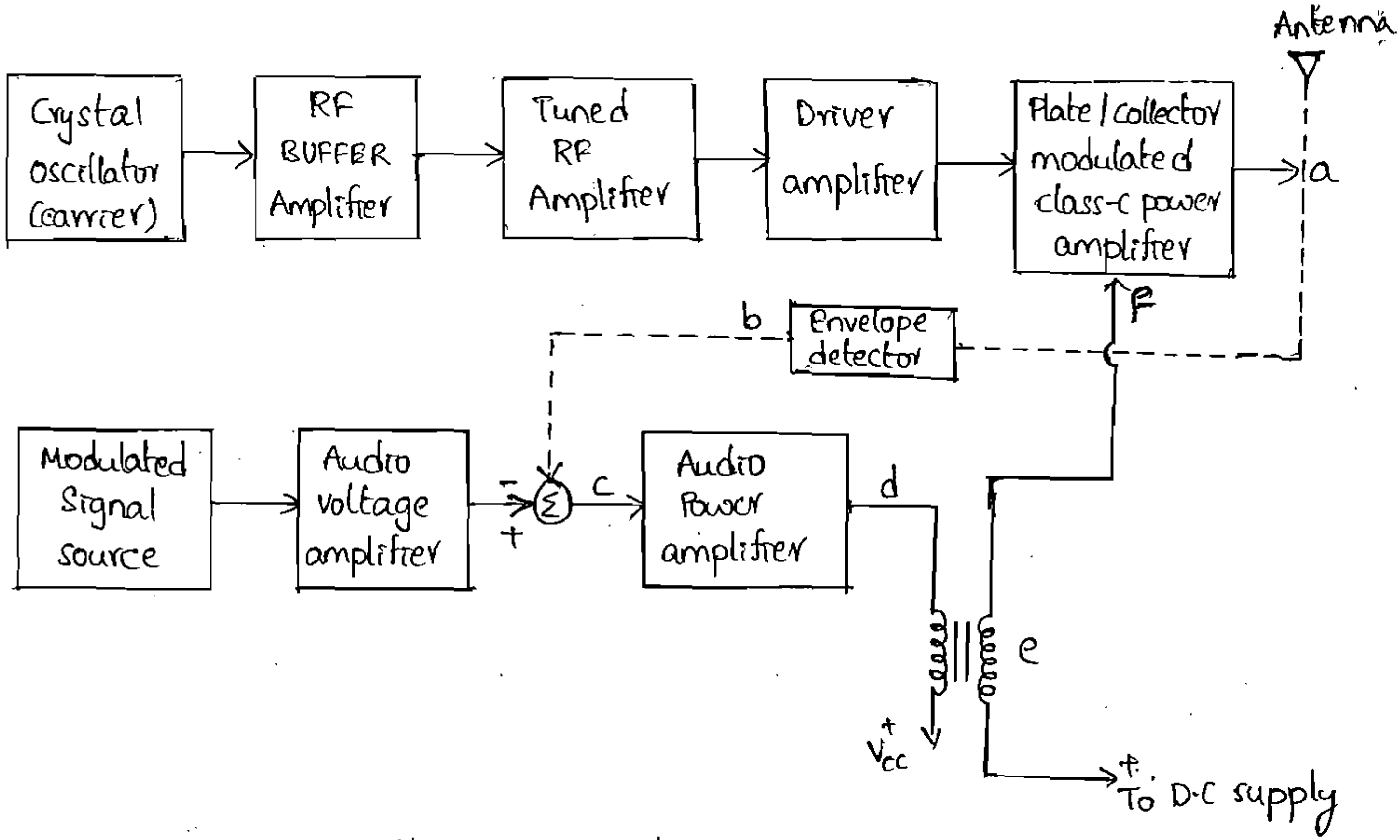
An RF oscillator and its circuit are similar to that of a low-level transmitter. The carrier signal also requires an additional power amplifier before it is given to the modulator. The final power amplifier is the actual modulator. Collector modulator class C type has a very good efficiency.

Advantage

The advantage of high-level modulator transmitter is that all the RF power amplifier can be class C power amplifier, which can be designed to have very high power efficiency of the order of 80 to 90%.

Effect of Feedback on performance of AM transmitters

Generally Negative feedback is provided in AM transmitters with ~~a~~ to improve their performance. The AM signal fed to the antenna should have, its envelope, as the message signal available at the output of the audio voltage amplifier.



This will be the case only if there is no distortion produced in the audio power amplifiers.

The AM signal to be radiated is picked up at the point 'a' its envelope is extracted. This is then subtracted from the voltage amplifier output. The loop a-b-c-d-e-f thus acts as the feedback loop.

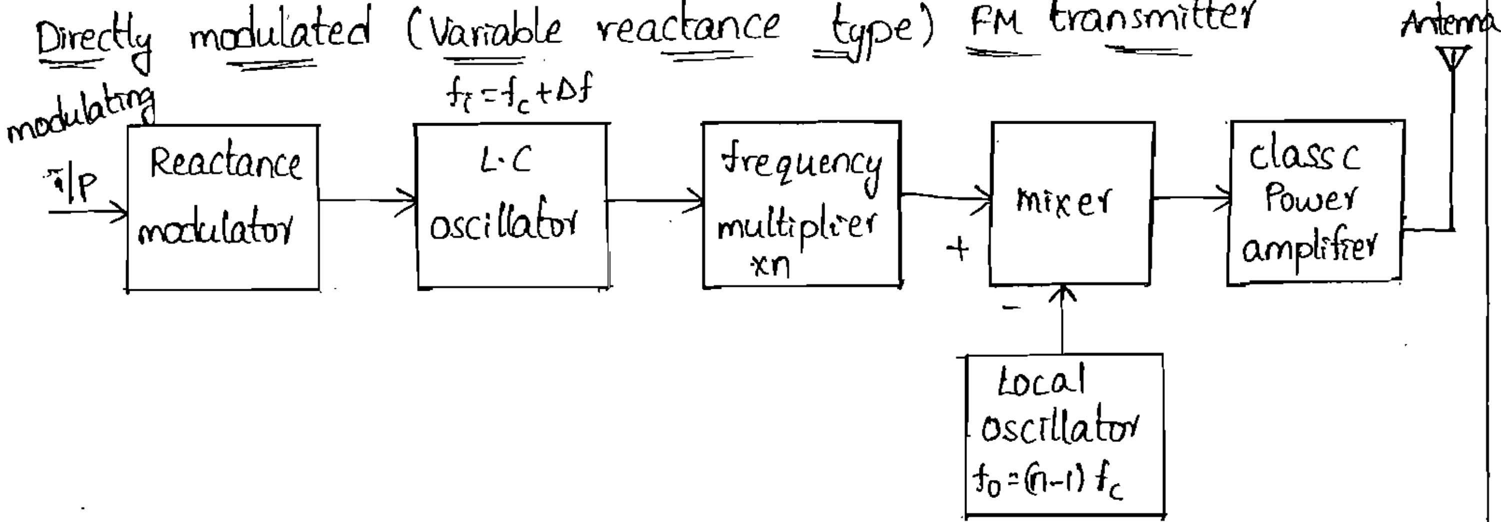
This negative feedback improves the performance of the transmitter as it reduces the distortion of the envelope of the radiated signal by making it closely resemble the message signal. It reduces the noise and power frequency also

→ FM transmitter's

FM signals can be generated either directly, by varying the frequency of the carrier oscillator, or indirectly by converting phase modulation to frequency modulation. According to modulation method employed there are two types of FM transmitters.

* Directly modulated / variable reactance type FM transmitter.
* Indirectly / Phase modulated FM transmitter.

Directly modulated (Variable reactance type) FM transmitter



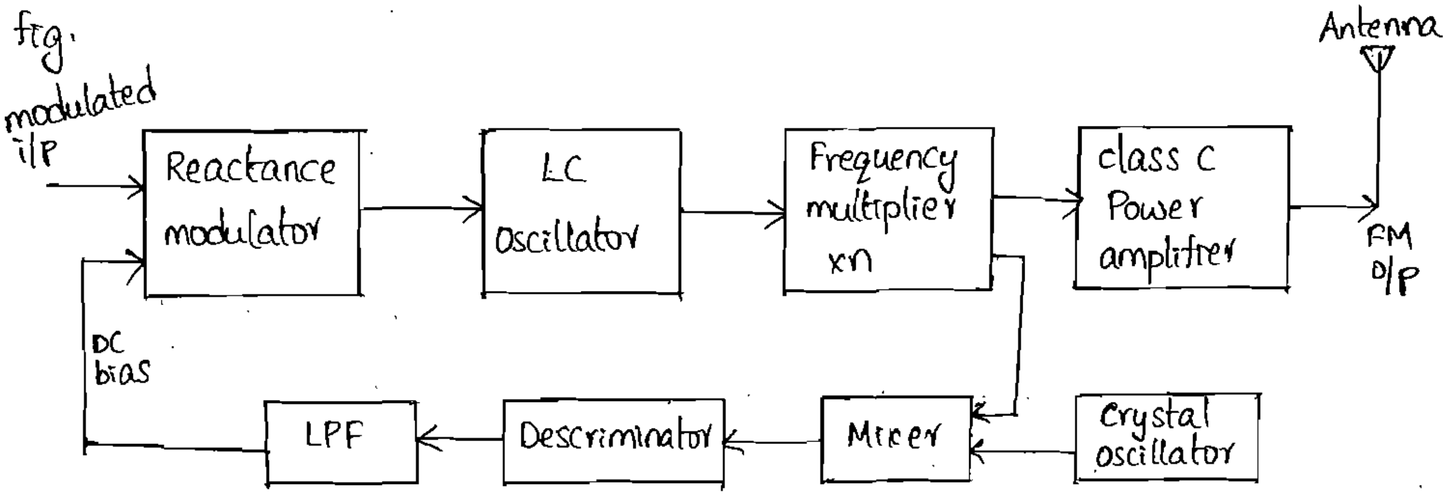
In directly modulated FM transmitters, the frequency modulation is carried out at a lower frequency and with a smaller frequency deviation. Then passing this frequency modulated wave through frequency multiplier circuit, the desired carrier frequency and desired frequency deviation is achieved.

With frequency multiplication, the instantaneous frequency is multiplied. For example, if the instantaneous frequency of an FM oscillator is $f_i = f_c + \Delta f_c$, when passed through a frequency multiplier this becomes $nf_i = nf_c + n\Delta f$, where n is the multiplying factor. The frequency multiplication can be achieved by passing the signal through a class C amplifier and tuning the output to the desired harmonic.

With frequency mixing, the deviation will not change. For example, if a signal with frequency $nf_c + n\Delta f$ is passed through a mixer, which is also fed by a local oscillator f_0, the output can be tuned to difference frequency $nf_c + n\Delta f - f_0$

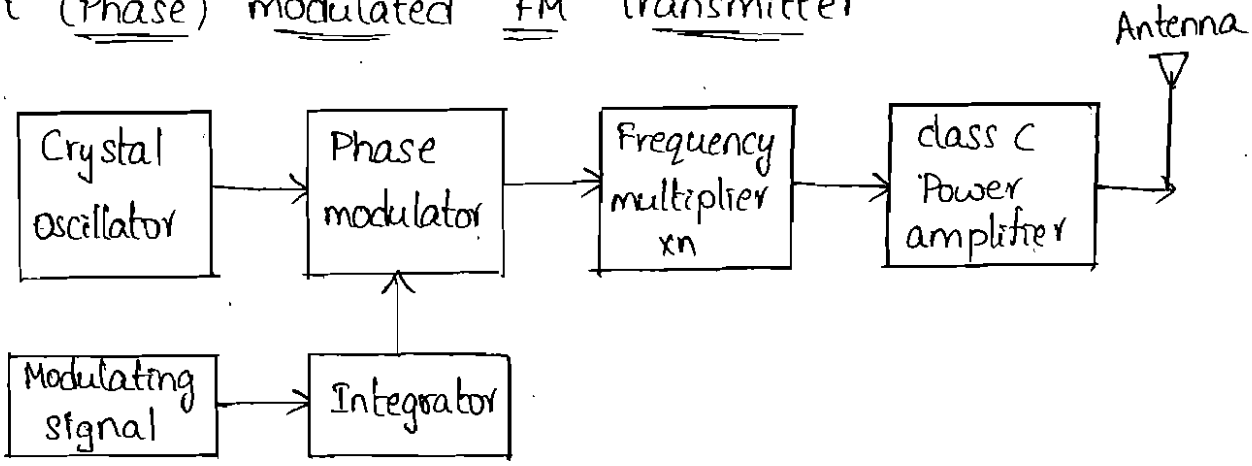
Frequency stability using AFC

It is difficult to maintain stability of unmodulated carrier frequency when LC oscillator is directly frequency modulated to produce relatively large deviation. The unmodulated carrier frequency can be kept stable using an automatic frequency control (AFC) ckt. The block diagram of a typical AFC circuit is shown in below fig.



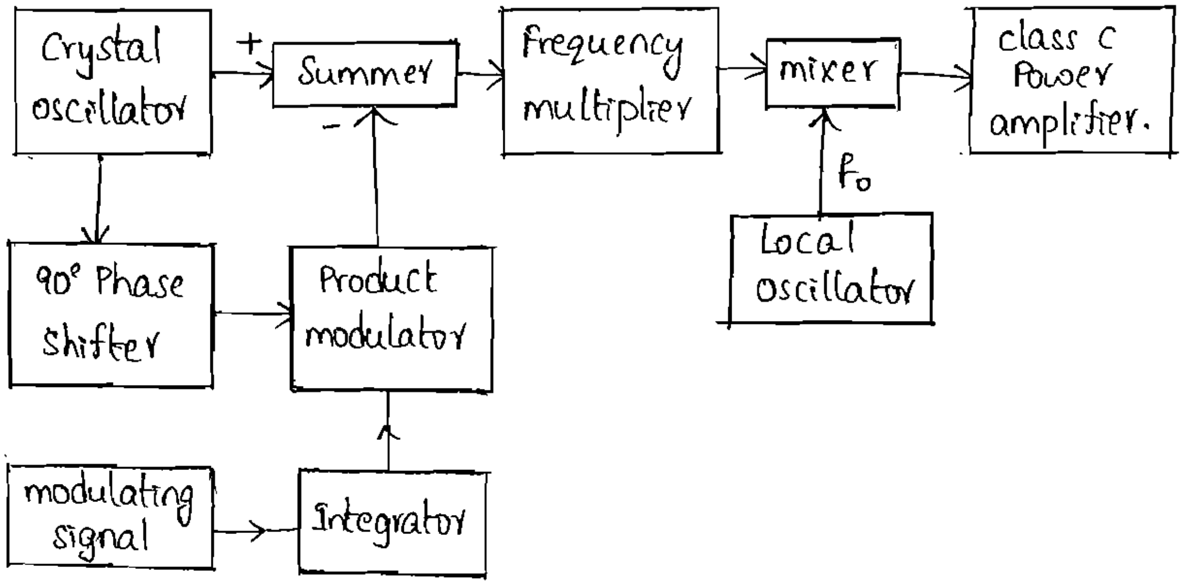
Suppose frequency of carrier increases. This higher frequency is fed to the mixer for which the other i/p frequency is from the stable crystal oscillator. A some what higher frequency will be fed to the descriminator. The descriminator will develop a positive dc voltage. The low pass filter (LPF) removes the signal component and leaves only dc voltage. The output of LPF i.e., the positive dc voltage is applied to the reactance modulator whose transconductance is increased by the positive dc voltage. This increase the equivalent capacitance of the reactance modulator there by decreasing the oscillator frequency. The frequency increase in the carrier frequency is thus lowered and brought to the correct value. Exactly, opposite action takes place when carrier frequency decreases.

Indirect (Phase) modulated FM transmitter



In this technique the phase angle is made to vary while holding the frequency constant. By this technique, a phase modulated signal is generated. With some minor processing this Phase modulated signal can be passed off as an FM as shown in fig.

A very popular indirect method of achieving FM is known as the "Armstrong method"



FM transmitter: Armstrong method.

In this method, the initial modulation takes place as an amplitude modulated DSBSC signal so that a crystal oscillator can be used in desired.

Here the crystal oscillator generates the sub carrier, which can be low, say on the order of 100KHz. One o/p from the oscillator is phase shifted by 90° to produce the sine term, which is then DSBSC modulated in the balanced modulator by $V_m(t)$ This is combined with the direct o/p from the oscillator in the summing amplifier, the result then being the phase modulated signal. The modulating signal is passed through an integrator to the modulator to get the frequency modulated signal. At this stage, the equivalent frequency deviation will be low, so the arrangement shown in fig. is used to increase the peak deviation.

RECEIVERS

Introduction

The primary requirement of any communication reciever is that it should have the ability to select the desired signal from among thousands of others present and to provide sufficient amplification to recover the modulating signal. To provide this primary requirement receiver has to carry out different functions. as given below.

1. Collect the electromagnetic waves transmitted by the transmitter.
2. Select the desired signal and reject all others.
3. Amplify the selected modulated carrier signal.
4. Detect the modulating signal from the modulated RF signal.
5. Amplify the modulating signal to operate the loud speaker.

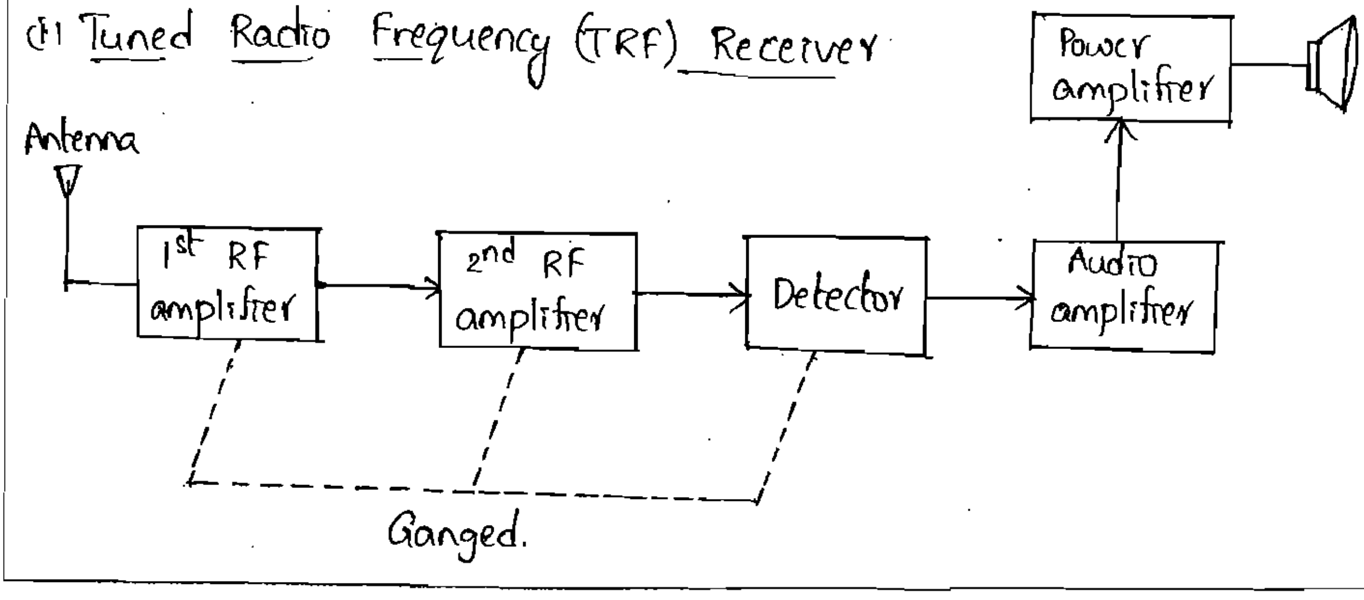
Receiver types

1. AM Receiver   2. FM Receiver

1. AM Receiver

(i) The Tuned Radio frequency Receiver and
(ii) Superheterodyne Receiver.

(i) Tuned Radio Frequency (TRF) Receiver



The TRF R_x consists of two or three stages of RF amplifiers, detector, audio amplifier and power amplifier. The RF amplifier stages placed between the antenna and detector are used to increase the strength of the receiver signal before it is applied to the detector. These RF amplifiers are tuned to fix frequency, amplify the desired band of frequencies. Therefore, they provide amplification for selected band of frequencies and rejection for all others. As selection and amplification process is carried out in two or three stages and each stage must amplify the same band of frequencies, the ganged tuning is provided.

The amplified signal is then demodulated using detector to recover the modulating signal. The recovered signal is amplified furthur by the audio amplifier followed by power amplifier which provides sufficient gain to operate a loud speaker. The TRF receivers suffered from number of annoying problems. These are listed in the next section.

Problems in TRF Receivers

1. Tracking of Tuned circuit

In TRF receiver tuned circuits are made variable so that they can be set to the frequency of the desired signal. In most of the receivers, the capacitors in the tuned circuits are made variable. These capacitors are 'ganged' b/w the stages so that they are can be changed simultaneously when the tuning knob is rotated. To have perfect tuning the capacitor values b/w the stages must be exactly same but this is not the case. The difference in the capacitors cause the resonant frequency of each tuned circuit to be slightly different, thereby increasing the pass band.

2. Instability

As high gain is achieved at one frequency by a multistage amplifier, there are more chances of positive feedback through some stray path, resulting in oscillations. These oscillations are unavoidable at high frequencies.

3. Variable Bandwidth

TRF receivers suffer from a variation in bandwidth over the tuning range. Consider a medium wave receiver required to tune over 535KHz to 1640KHz and it provides the necessary bandwidth of 10KHz at 535KHz Let us calculate Q of this ckt.

$$\Rightarrow Q = \frac{f}{BW} = \frac{535k}{10k} = 53.5$$

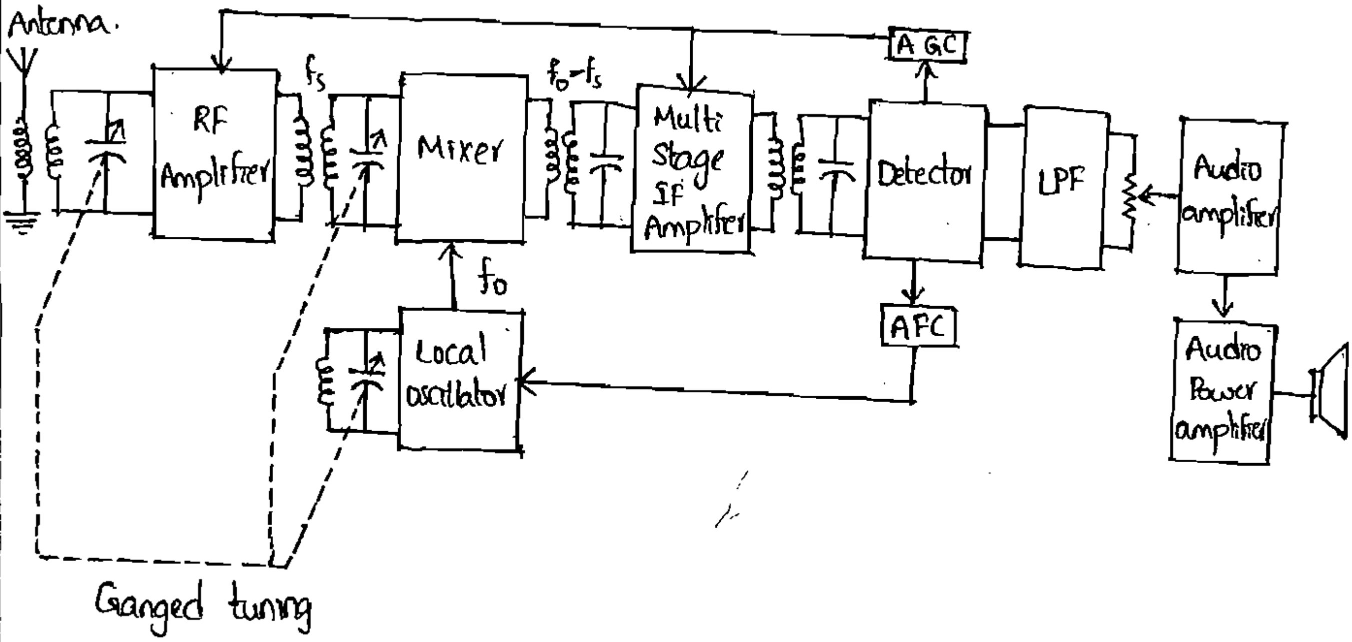
At 1640KHz Q of the coil should be 164 (1640k/10k)

However, in practice due to various losses depending on freq. we will not get so large increase in Q. let us assume that at 1640KHz frequency Q is increased to value. 100 instead of 164. With this Q of the tuned tuned circuit bandwidth can be calculated as follows $\Rightarrow BW = \frac{f}{Q} = \frac{1640k}{100} = 16.4KHz$

We know necessary BW is 10KHz we can say that in TRF R_x the BW of the tuned circuit varies over the frequency range, resulting in poor selectivity of the receiver.

Because of the problems of tracking, instability and bandwidth variation, the TRF receivers have almost been replaced by super heterodyne R_x.

Superheterodyne Receivers



In Super heterodyne Rx, first all the incoming RF frequencies are converted to a fix lower frequency called Intermediate frequency (IF). Then this fix intermediate frequency is amplified and detected to reproduce the original information since the characteristics of the IF amplifier are independent of the frequency to which the receiver is tuned, the selectivity and sensitivity of super heterodyne receivers are fairly uniform through out its tuning range.

Mixer circuit is used to produce the frequency translation of the incoming signal down to IF. The incoming signals are mixed with the local oscillator frequency signal in such a way that a constant frequency difference is maintained b/w the local oscillator and the incoming signals. This is achieved by using ganged tuning capacitors.

As shown in the figure, antenna picks up the weak radio signal and feeds it to the RF amplifier

The RF amplifier provides some initial gain and selectivity. The o/p of the RF amplifier is applied to the i/p of the mixer. The mixer also receives an i/p from local oscillator.

The o/p of the mixer circuit is difference frequency (f_o-f_s) commanly known as IF. The signal at this intermediate frequency contains the same modulation as the original carrier. This signal is amplified by one or more IF amplifier stages and most of the receiver gain is obtained in these IF stages. The highly amplified IF signal is applied to detector circuits to recover the original modulating information. Finally the o/p of detector ckt is fed to audio and power amplifier which provides a sufficient gain to operate a speaker. Another important ckt in the Superheterodyne receiver are AGC and AFC circuit. AGC is used to maintain a constant o/p voltage level over a widerange of RF i/p signallevels.

AFC circuit generates AFC signal which is used to adjust and stabilize the freq of local oscillator.

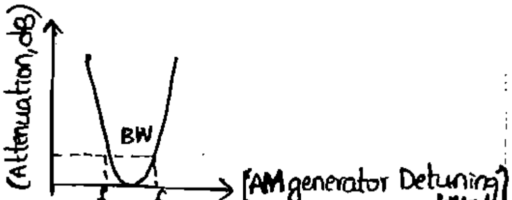
<u>Receiver characteristics</u>

The performance of the radio receiver can be measured in terms of following receiver characteristics.

1. Selectivity
2. Sensitivity
3. Fedility
4. Image frequency and its rejection
5. Double spotting

1. Selectivity

Selectivity refers to the ability of a receiver to select a signal of a desired frequency while reject all others. Selectivity in a receiver is obtained by using tuned circuits. These are LC circuits tuned to respnate at a desired signal frequency. The Q of these tuned circuits determines the selectivity. A good receiver isolates the desired signal in the RF spectrum and eliminate all other signals. we know that bandwidth of the tuned ckt is given by, $BW = \frac{f_r}{Q}$



where f_r is the resonant frequency

Narrower the bandwidth better the selectivity. The below figure shows the selectivity curve for the typical tuned ckt. As shown in the figure below, bandwidth is the difference b/w the upper f_2 and lower f_1 cutoff frequencies which are located at the 3dB or 0.707 points on the sensitivity curve.

2. Sensitivity

The sensitivity of a communication receiver refers to the receivers ability to pickup weaksignals, and amplify it. The more gain that a receiver has, the smaller the i/p signal necessary to produce desired o/p power. Therefore, sensitivity is a primary function of the overall receiver gain. It is ~~of~~ often expressed in microvolts or in decibells. The sensitivity of receiver mostly depends on the gain of the IF amplifiers. Good communication receiver has Sensitivity of 0.2 to 1μV

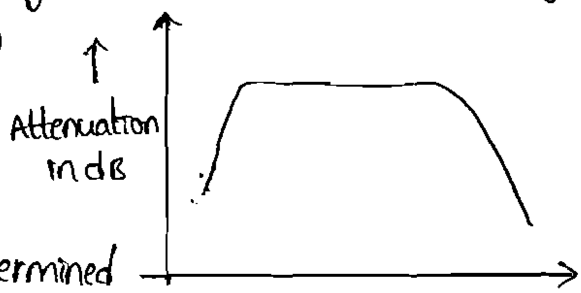
Fig. Sensitivity curve for typical receiver.

3. Fedility

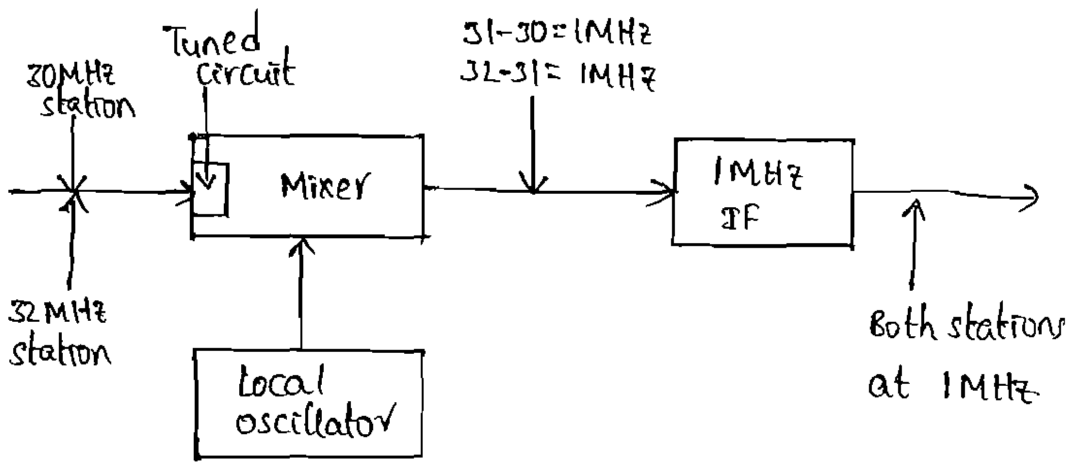
Fedility refers to the ability of the receiver to reproduce all the modulating frequencies equally. The below figure shows the typical fedility curve for radio receiver.

The fedility at the lower modulating frequencies is determined by the low frequency response of the IF amplifier and the fedility at the higher modulating frequencies is determined by the high frequency response of the IF amplifier. Fedility is difficult to obtain in AM receiver because good fedility requires more bandwidth of IF amplifier resulting in poor selectivity.

Fig: Typical Fedility curve



4. Image frequency and its Rejection



Consider a superheterodyne receiver having an intermediate frequency of 1MHz tuned to receive a 30MHz station. The local oscillator frequency necessary for the tuning is equal to 31MHz. so that it may produce an intermediate frequency of 1MHz.

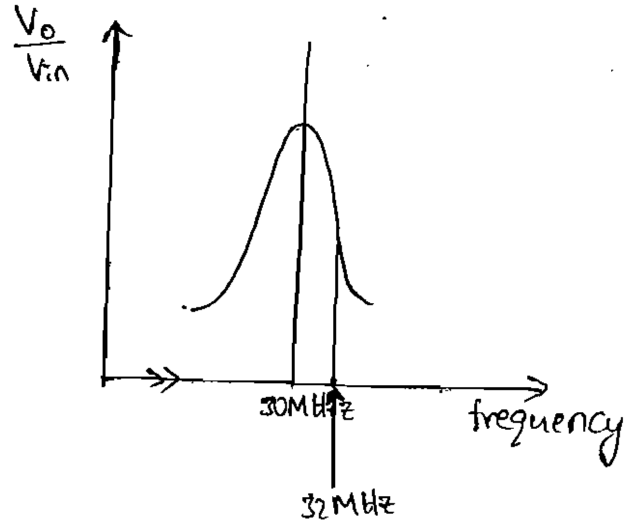


Fig: Response of the tuned.

If another station operating at 32MHz is also present in the air, it is possible for it to get into the mixer. As soon as this signal at 32MHz (undesired station) is present at the mixer input, it will produce a difference frequency with the L.O. frequency This difference frequency 1MHz (32MHz−31MHz) which is the same IF Thus we now have the undesired 32MHz station in addition to the desired 30MHz station. In the IF section both the desired and undesired signal will be amplified. The desired frequency of 32MHz in this case is called the image frequency.

A superheterodyne receiver must therefore, be designed to have a very high image frequency rejection capability to get rid of this Spurious frequency.

5. Double spotting

The phenomenon of double spotting occurs at higher frequencies due to poor front end selectivity of the receivers. In this, receiver picks up same short wave station at two nearby points on the receiver dial.

When the receiver is tuned across the band, a strong signal appears to be at two different frequencies, once at the desired frequency and again when the receiver is tuned to 2 times IF below the desired frequency. In this second case, the signal becomes the image, reduced in strength by the image rejection, thus making it appear that the signal is located at two frequencies in the band.

Receiver sections

RF Amplifier

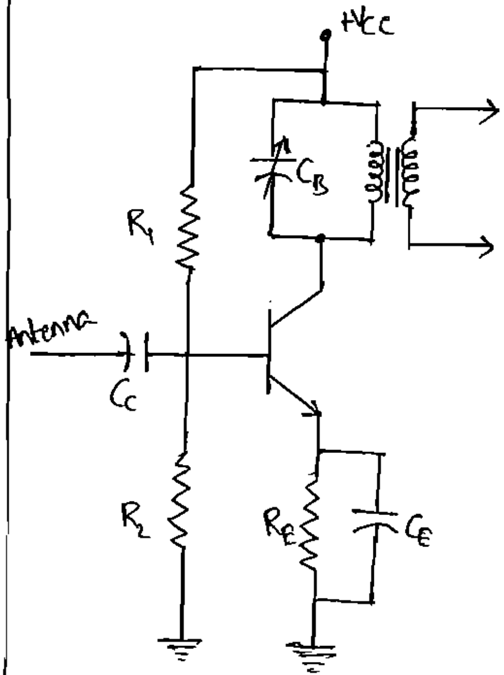


fig a.

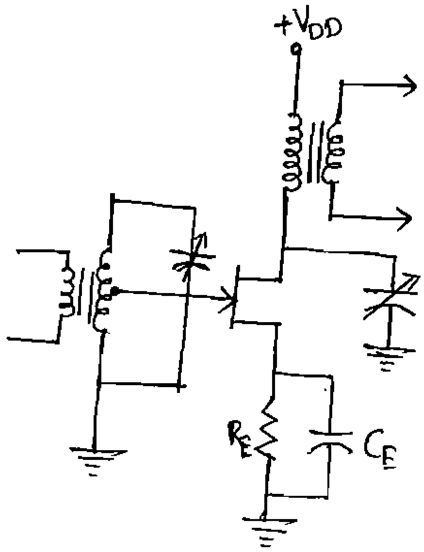


fig b.

The RF amplifier is a tunable circuit. It is there to select the wanted frequency and reject some of the unwanted frequencies and thus to improve signal to noise ratio. It provides initial gain and selectivity. It is a tuned circuit followed by an amplifier is usually a simple class A circuit.

The values of resistors R_1 and R_2 in the bipolar circuit are adjusted such that the amplifier works as class A amplifier. The antenna is connected through coupling capacitor to the base of the transistor. This makes the circuit very broad band as the transistor will amplify virtually any signal picked up by the antenna. However the collector is tuned with a parallel resonant circuit to provide the initial selectivity for the mixer input.

The FET circuit shown in figure is more effective than the transistor circuit. Their high input impedence minimizes the loading on tuned circuits, thereby permitting the Q of the circuit to be higher and selectivity to be sharper.

The receiver having an RF amplifier stage has following advantages.

1. It provides greater gain, i.e., better sensitivity.
2. It improves image-frequency rejection.
3. It improves signal to noise ratio.
4. It improves rejection of adjacent unwanted signals, providing better selectivity.
5. It provides better coupling of the ~~unwanted signals~~ receiver to the antenna.

Mixer or frequency changer/ converter

Types
- Seperately Excited Mixer
- Self Excited Mixer.

(i) Seperately excited mixer

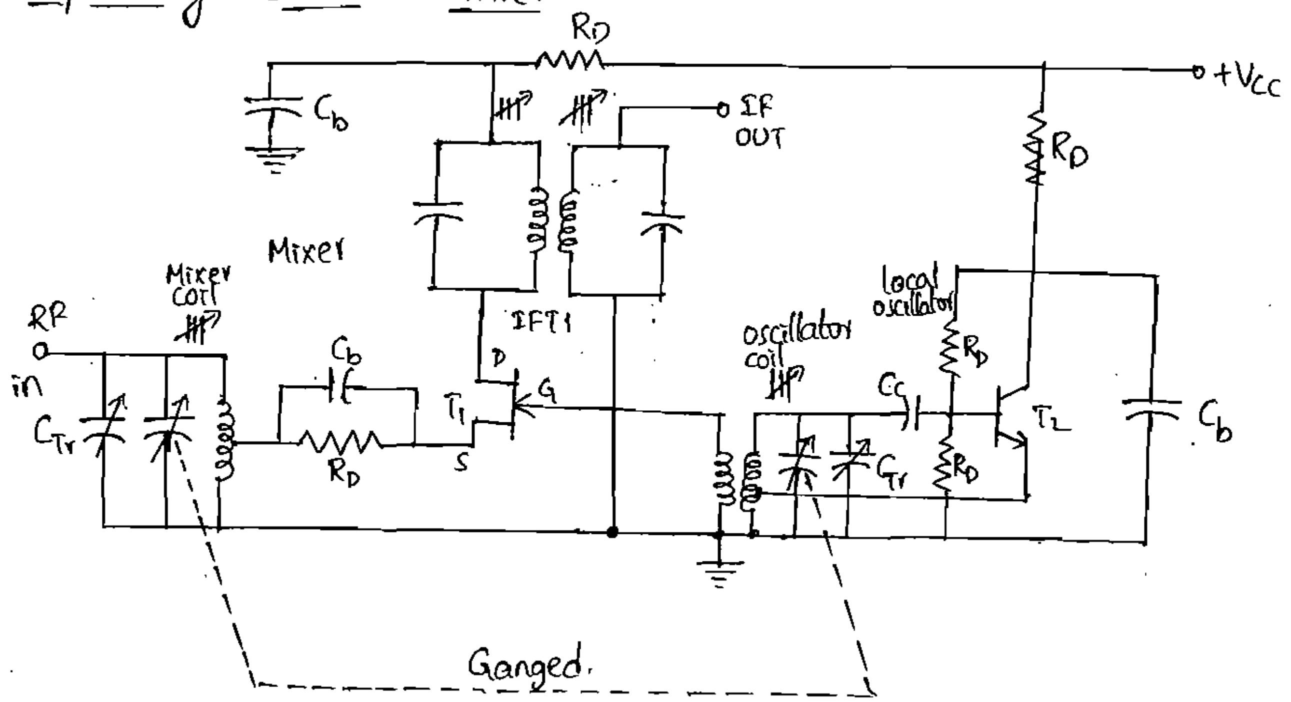


Fig: Seperately excited FET mixer.

In fig one device acts as a mixer while the other supplies the necessary oscillations. The bipolar transistor T_2, forms the Hartley oscillator circuit. It oscillates with local frequency (f_0) FET T_1, is a mixer, whose gate is fed with the output of local oscillator and its bias is adjusted such that it operates in a nonlinear portion of its characteristic. The local oscillator varies the gate bias of the FET to to vary its trans conductance in a nonlinear manner. resulting intermediate frequency at the output. The output is taken through double tuned transformer in the drain of the mixer and fed to the IF amplifier. The ganged tuning capacitor allows simultaneous tuning of mixer and local oscillator.

The C_{Tr}, a small trimmer capacitors across each of the tuning capacitors are used for fine adjustments.

(ii) Self excited Mixer

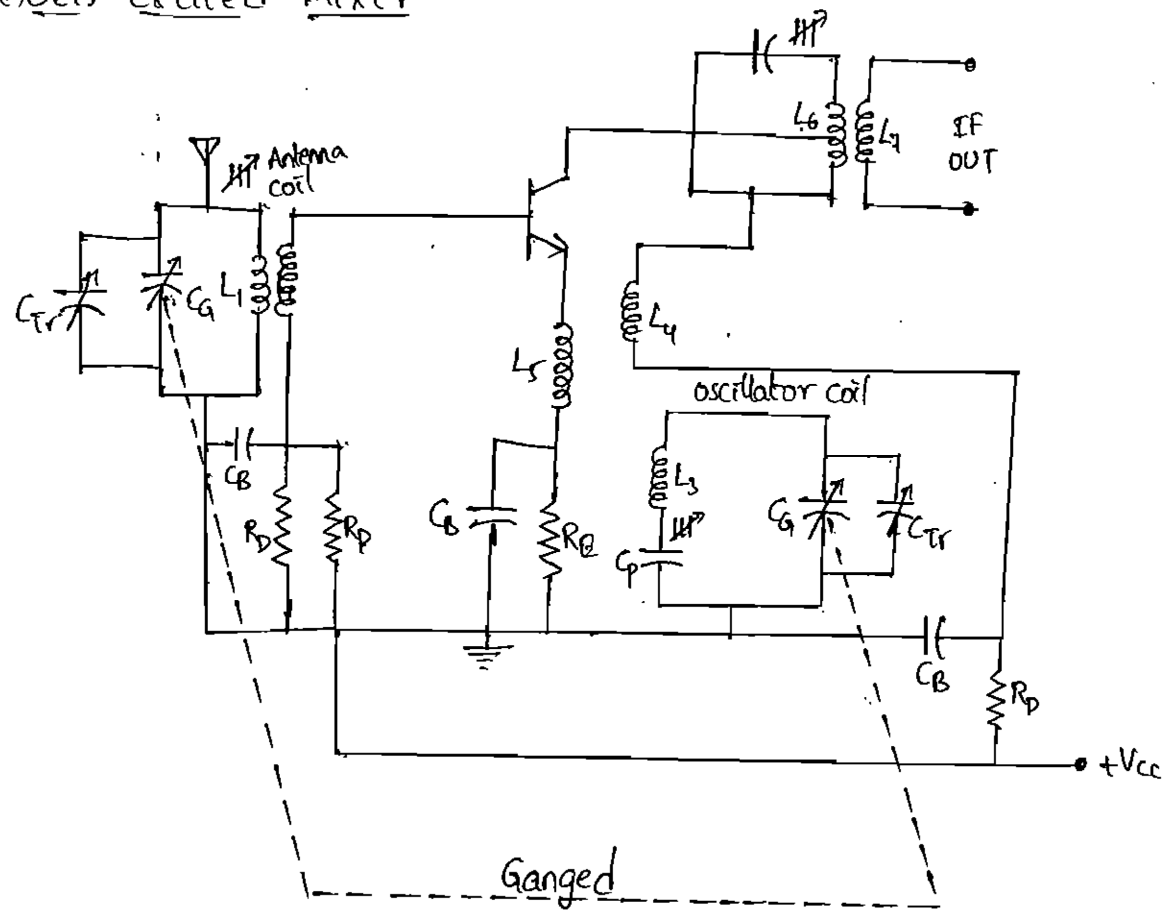


Fig: Self excited Mixer.

It is possible to combine the function of the mixer and local oscillator in one circuit. The circuit is commanly known as self excited mixer. The circuit oscillates and the transconductance of the transistor is varied in a nonlinear manner at the local oscillator rate. This variable transconductance (g_m) is used by the transistor to amplify the incoming RF signal.

Tracking

The process of tuning circuits to get the desired output is called Tracking. Any error that exists in the frequency being fed to the IF amplifier such errors are known as 'Tracking errors' and these must be avoided. To avoid tracking errors standard capacitors are not used, and ganged capacitors with identical sections are used. A different value of inductance and special extra capacitors called trimmers and padders are used to adjust the capacitance of the oscillator to the proper range. There are three common methods used for tracking. These are:

1) Padder tracking
2) Trimmer tracking
3) Three-point tracking.

1) Padder tracking

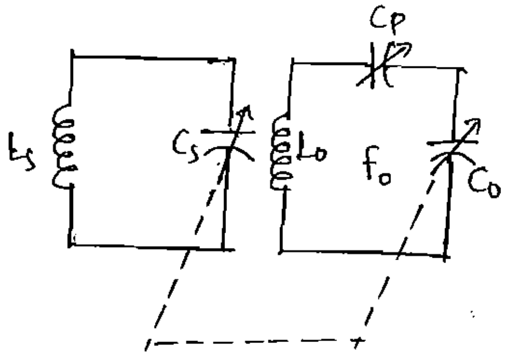


Fig: Padder tracking

In padder tracking the oscillator tunes below the frequency it should be in midband. So the IF created is higher than it is created.

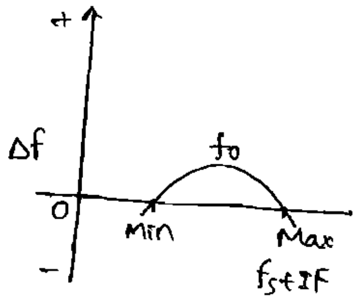


Fig: Tracking error in padder tracking.

2) Trimmer tracking

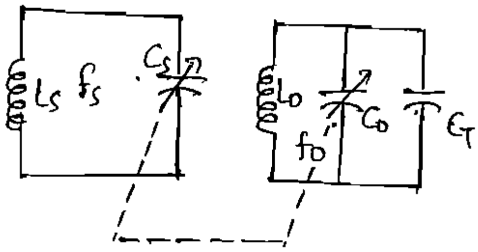


Fig: Trimmer tracking.

In trimmer tracking, the oscillator tunes higher the freq it should be in midband, So the IF created is less than it should be, and a -ve error is created.

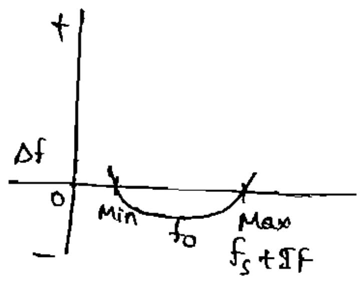


Fig: Tracking error in trimmer tracking.

3) Three-point tracking

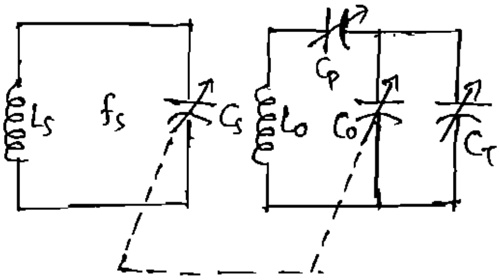


Fig: Three-point tracking.

The combination circuit called three point tracking can be adjusted to give zero error at three points across the band, at each end, and at the middle.

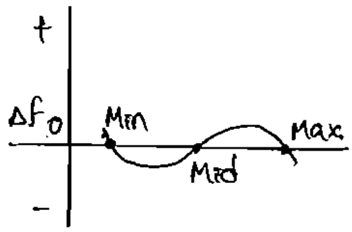


Fig. Tracking error in three point tracking.

Procedure to get values of capacitor required in above ckts

1) Find the min and max frequencies and the required oscillator capacitance ratio.

$$f_{omin} = f_{smin} + IF$$

$$f_{omax} = f_{smax} + IF$$

oscillator capacitance ratio can be given as

$$\frac{C_{omax}}{C_{omin}} = \left(\frac{f_{omax}}{f_{omin}}\right)^2$$

2. Calculate the capacitance ratio and max value of the signal circuit tuning capacitance.

$$\frac{C_{smax}}{C_{smin}} = \left(\frac{f_{smax}}{f_{smin}}\right)^2$$

$$C_{smax} = \left(\frac{f_{smax}}{f_{smin}}\right)^2 \times C_{smin}$$

3. Calculate the oscillator tuning capacitance. It is given by $C_o = C_s$ in series with C_p

$$C_o = \frac{C_s C_p}{C_s + C_p}$$

4. Calculate value of padder capacitor C_p using ratios.

$$\frac{C_{omax}}{C_{omin}} = \frac{C_{smax} \text{ in series with } C_p}{C_{smin} \text{ in series with } C_p} = \frac{(C_{smax}\,C_p)/(C_{smax}+C_p)}{(C_{smin}\cdot C_p)/(C_{smin}+C_p)}$$

$$\frac{C_{omax}}{C_{omin}} = \frac{C_{smax}\,(C_{smin}+C_p)}{C_{smin}\,(C_{smax}+C_p)}$$

5. Obtain the oscillator coil value. It is given as

$$L_o = \frac{1}{(2\pi f_{omin})^2 C_{omax}}$$

$$= \frac{1}{(2\pi f_{omax})^2 C_{omin}}$$

<u>Intermediate frequency amplifier</u>

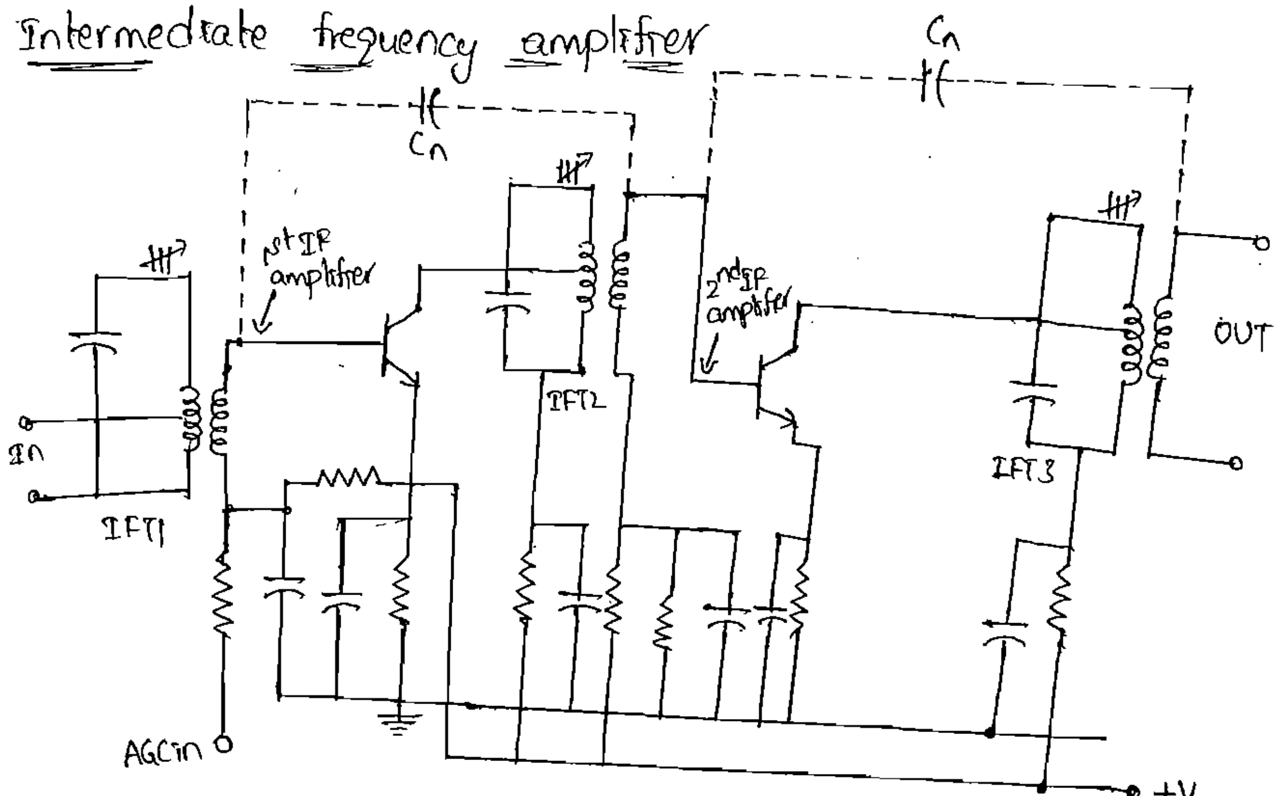


Fig: Two-stage IF amplifier.

IF amplifiers are tuned voltage amplifiers tuned for the fixed frequency. Its important function is to amplify only tuned frequency signal and reject all others. As we know most of the receiver gain, usually two or more stages of IF amplifiers are required.

The ~~above~~ figure shows the two stage IF amplifier. Two stages are transformer coupled and all IF transformers are single tuned, i.e., tuned for single frequency.

Choice of Intermediate frequency

Selection of the intermediate frequency depends on various factors while choosing the intermediate frequency it is necessary to consider following factors.

1. Very high intermediate frequency will result in poor selectivity and poor adjacent channel rejection
2. A high value of IF increases tracking difficulties,
3. At low values of intermediate frequency, image frequency rejection is poor.
4. At very low values of IF, selectivity is too sharp cutting off the sidebands.

With the above considerations the standard broadcast AM receivers [tuning to 540 to 1650 KHz] use an IF within the 438 KHz to 465 KHz range. The 465 KHz IF is most commanly used.

Automatic Gain Control (AGC)

Automatic gain control is a system by means of which the overall gain of a radio receiver is varied automatically with the variations in the strength of the receiver signal, to maintain the output constant.

There are two types of AGC circuits in use

→ Simple AGC

→ Delayed AGC

Simple AGC

In simple AGC receivers the AGC bias starts to increase as soon as the received signal level exceeds the background noiselevel. As a result receiver gain starts falling down, reducing the sensitivity of the receiver.

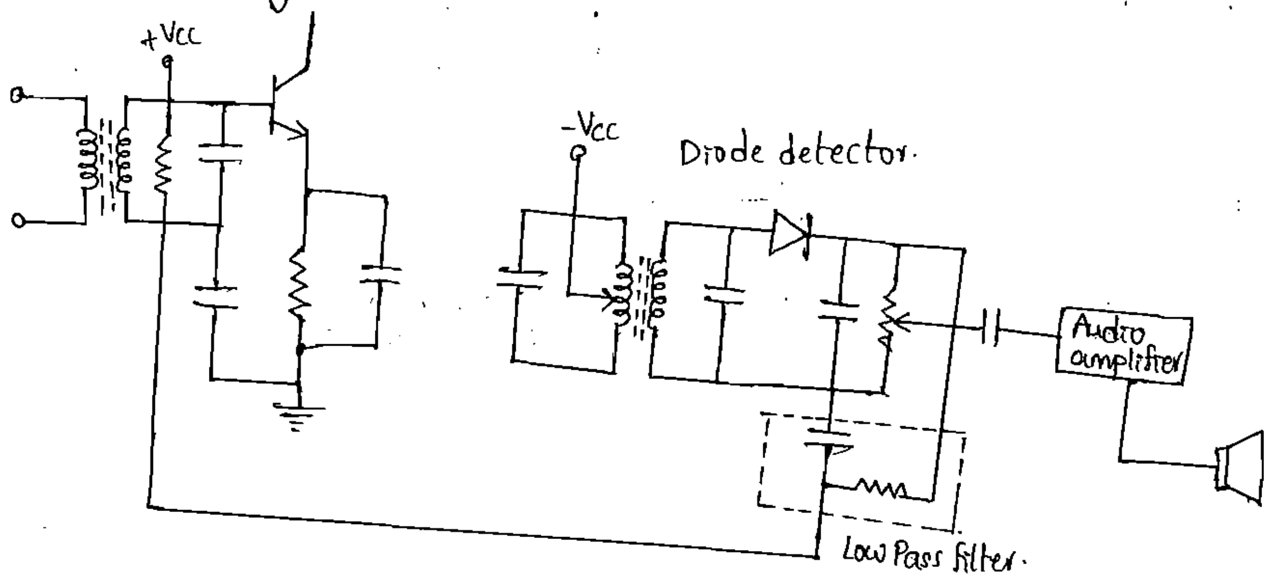


Fig: Simple automatic gain control circuit.

In this circuit, dc bias produced by halfwave rectifier as a AM detector is used to control the gain of RF or IF amplifier. Before application of this voltage to the base of the RF and local oscillator in IF stage amplifier the audio signal is removed by the low-pass filter. The time constant of the filter is kept at least 10 times longer than the period of the lowest modulation frequency received. If the time constant is kept longer; it will give better filtering, but it will cause on annoying delay in the application of the AGC control when tuning from one signal to another. The recovered signal is then passed through C_c to remove the dc. The resulting ac signal is further amplified and applied to the loudspeaker.

Delayed AGC

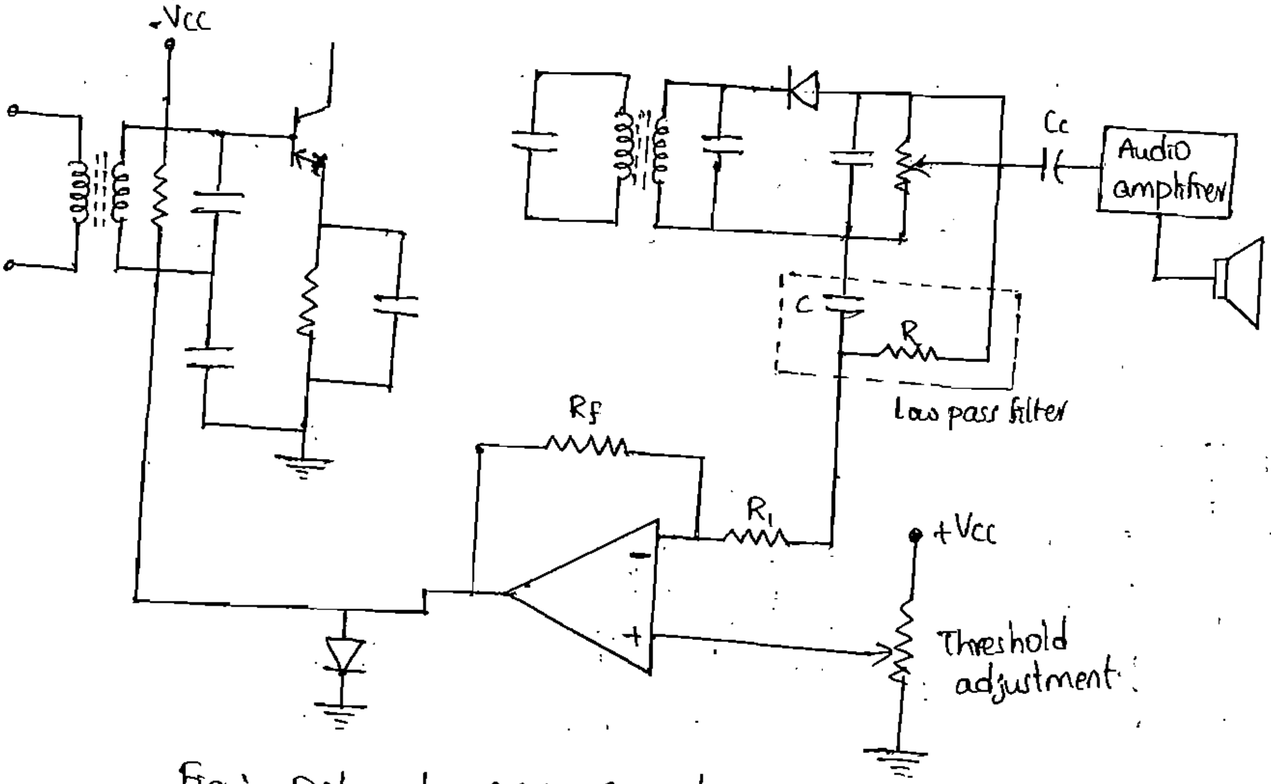
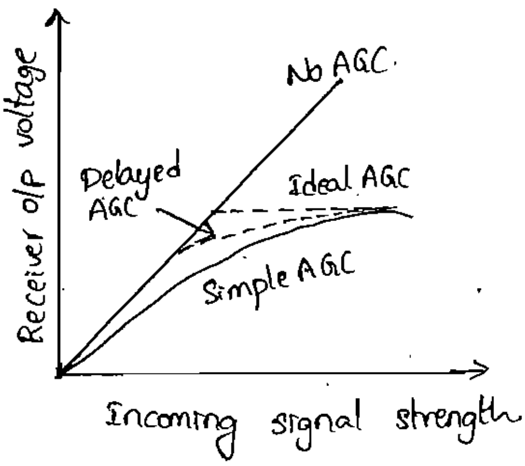


Fig: Delayed AGC Circuit.

Simple AGC is clearly an improvement over no AGC at all. Unfortunately, in simple AGC circuit, the unwanted weak signals are amplifier with high gain. To avoid this, in delayed AGC circuits AGC bias is not applied to amplifiers until signal strength has reached a predetermined level, after which AGC bias is applied as with simple AGC, but more strongly.

Here, AGC o/p is applied to the difference amplifier by diode detector is above certain dc threshold voltage. This threshold voltage can be adjusted by adjusting the voltage at the positive input of the operational amplifier.

The below figure shows the response of a receiver with either simple or delayed AGC compared to one without AGC.

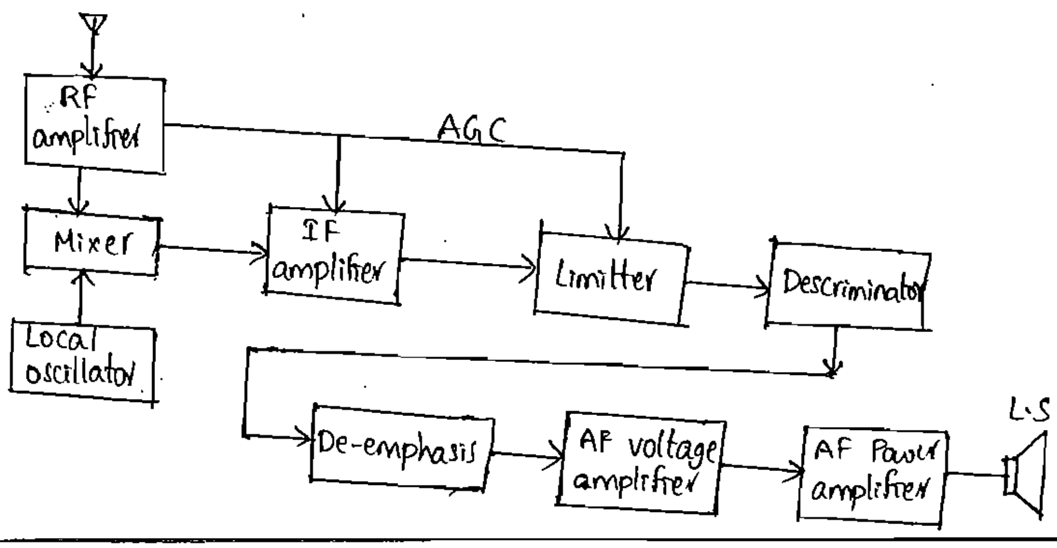


FM Receiver

The FM Receiver is basically a superheterodyne receiver, similar to AM receiver. However it differs from AM receiver with respect to following points.

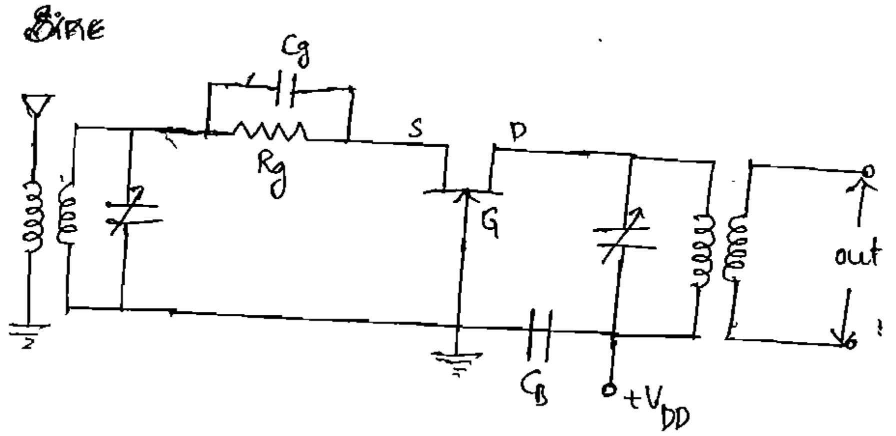
- → AM receiver operates in MW and SW bands, while FM receiver operates at much higher frequencies viz. 88MHz to 108MHz.
- → Limiter and deemphasis circuits are required only in FM receiver
- → The technique of demodulating FM signal is different from detection of AM signal.
- → FM receiver uses different methods of obtaining AGC.

Block diagram of FM Receiver



Different Stages in F.M Receivers.

R.F Amplifier Stage



Since F.M Signal has a larger bandwidth it is likely to encounter more noise. Hence to reduce the noise figure of the receiver. An RF amplifier stage is used. This circuits have low input impedance, Suitable for matching with antenna impedance.

A typical circuit is shown in the above figure. Since the gate terminal is grounded, the i/p and o/p sides are isolated for RF purposes. There is no possibility of feedback and there by instability in operation. Therefore the circuit doesnot require neutralization. The low input impedance of the FET amplifier can be easily matched to antenna through a single secondary tuned RF transformer. Bath the i/p and o/p tank circuits are tuned to carrier frequency.

Mixer Stage

With the help of local oscillator, this stage down converts the incoming carrier frequency to I.F, which is 10.7 MHZ for FM Receiver. The localoscillator is usually the clapp oscillator, suitable for VHF frequency and local oscillator frequency is not a

Problem in FM Receiver unlike in an AM Receiver. Compared to AM Receiver, tuning range of incoming carrier frequencies for FM Receiver is small, from 88MHZ to 108MHZ i.e; about 1:25:1. Thus the tracking is comparatively easy in FM Receiver.

Since FETs are less noisy than BJTs, RF amplifier stage and mixer stage uses FETs. With local oscillator constructed with BJT.

The mixer stage uses of tuned circuit as its load. The circuit is tuned to Intermediate frequency of 10.7 MHZ and hence selects the difference between incoming carrier frequency and locally generated oscillator frequency.

I.F. Amplifier Stage

In the I.F Amplifier stages, the most of the gain of receiver is developed. The intermediate frequency and bandwidth requirements are normally much larger than in AM Receiver. The typical values for an F.M Receiver operating in FM band from 88MHz to 108MHZ are 10.7MHZ for I.F and 200kHZ for bandwidth. Generally two I.F amplifier stages are employed.

The I.F Amplifier stage uses a tuned circuit as its load. The circuit is tuned to intermediate frequency.

Amplitude Limiting

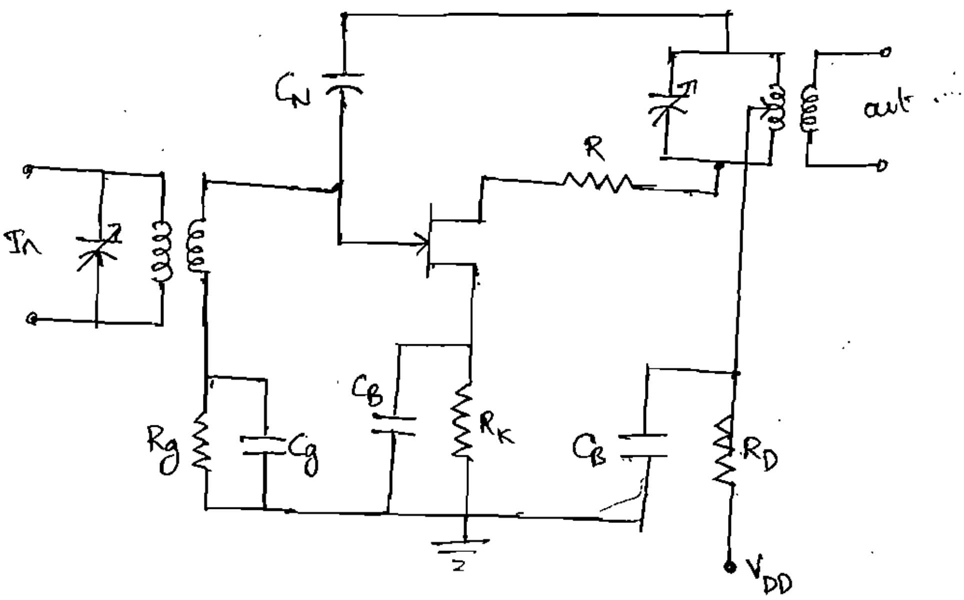


Fig: Amplitude Limitter.

To Remove the amplitude variations of the Signal is main function of the amplitude limiter.

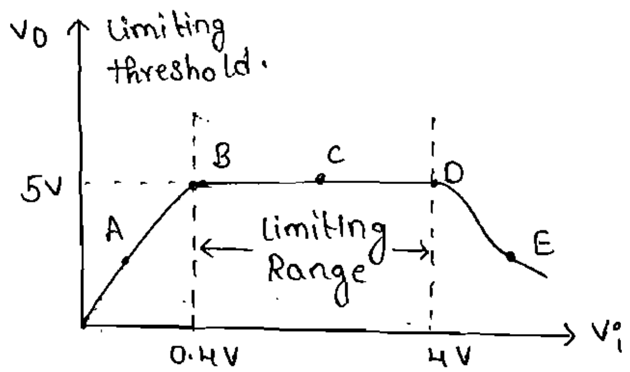


fig : Typical Limiter Response Characteristics.

The Response characteristic of the amplitude limiter. It indicates clearly that limiting takes place only for a Certain range of i/p voltage, outside which output varies with input.

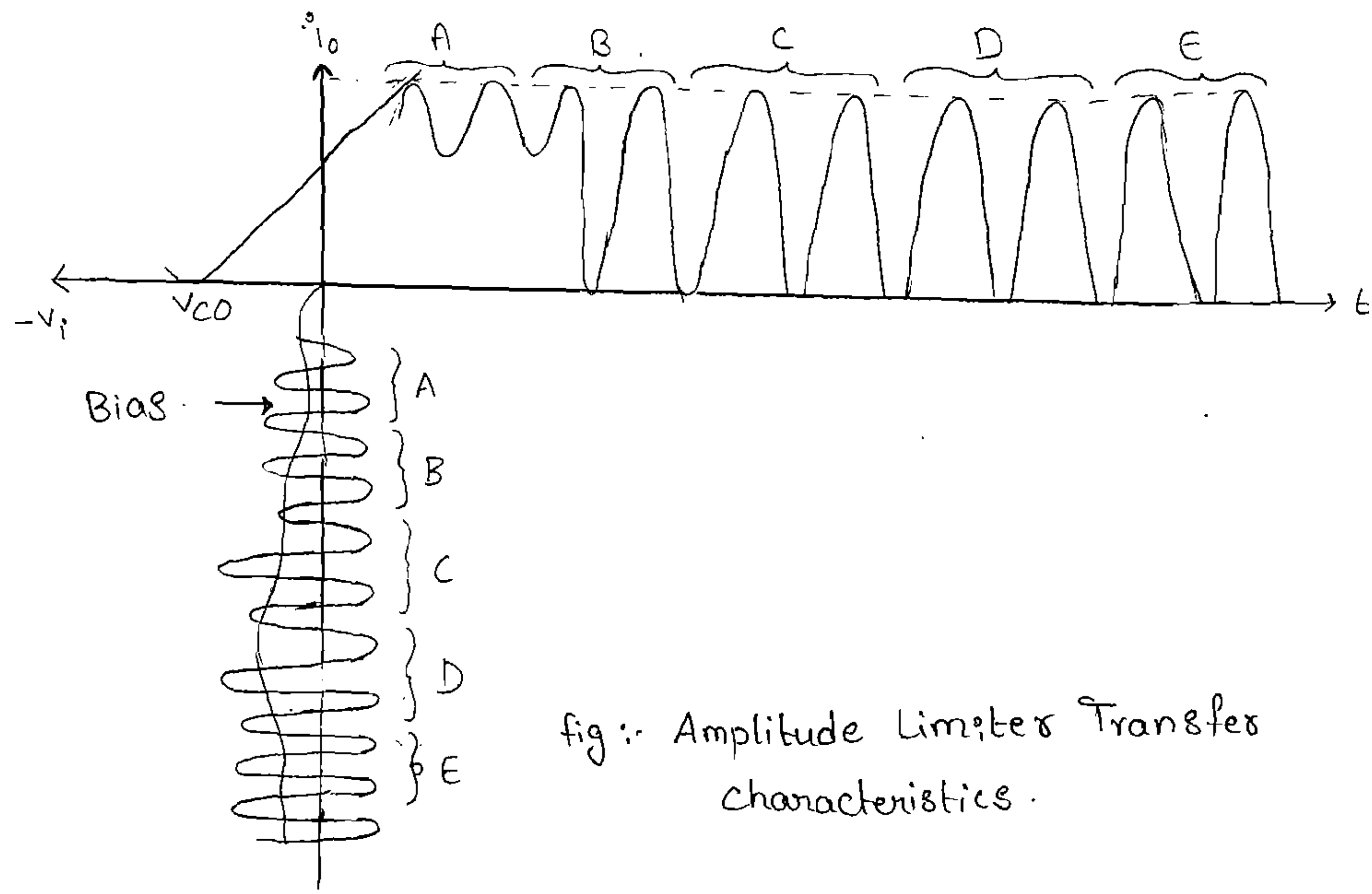


fig :- Amplitude Limiter Transfer Characteristics.

Referring to the above fig. we see that as i/p increases from value A to B ~~input~~ output current also increases. Thus no limiting has yet taken place. However, Comparision of B and c shows that they both yield the same o/p current and voltage. Thus limiting has now begun. value B is the point at which limiting starts and is called the threshold of limiting. As input increases from C to D, there is no rise in o/p, all that happens is that the O/P current flows for a some what shorter portion of the i/p cycle.